Computer Networking

Slide Set 3

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Topic 5 – Transport

Our goals:
• understand principles behind transport layer services:
  – multiplexing/demultiplexing
  – reliable data transfer
  – flow control
  – congestion control
  – buffers

• learn about transport layer protocols in the Internet:
  – UDP: connectionless transport
  – TCP: connection-oriented transport
  – TCP congestion control
  – TCP flow control
Transport Layer

• Commonly a 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/tasks at hosts
  – Need a way to decide which packets go to which applications (*more multiplexing*)
Why a transport layer?
Why a transport layer?

many application processes

Host A

Host B

Application
Transport
Network
Datalink
Physical

Drivers + NIC
Datalink
Physical

Operating System
IP

browser
browser
telnet
ftp
mmedia

Host A

Host B
Why a transport layer?

- Many application processes

Communication between processes at hosts

Communication between hosts (128.4.5.6 ↔ 162.99.7.56)

Host A

Host B
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 (mux/demux)

• 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
Role of the Transport Layer

• Communication between application processes
  – Multiplexing between application processes
  – Implemented using *ports*
Role of the Transport Layer

• Communication between application processes

• Provide common end-to-end services for app layer [optional]
  – Reliable, in-order data delivery
  – Paced data delivery: flow and congestion-control
    • too fast may overwhelm the network
    • too slow is not efficient

(Just Like Computer Networking Lectures....)
Role of the Transport Layer

• Communication between processes
• Provide common end-to-end services for app layer [optional]
• TCP and UDP are the common transport protocols
  – also SCTP, MTCP, SST, RDP, DCCP, ...
Role of the Transport Layer

• 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, no-frills transport protocol
  – only provides mux/demux capabilities
Role of the Transport Layer

• 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, no-frills transport protocol
• TCP is the *totus porcus* protocol
  – offers apps a reliable, in-order, byte-stream abstraction
  – with congestion control
  – but **no** performance (delay, bandwidth, ...) guarantees
Role of the Transport Layer

• Communication between processes
  – mux/demux from and to application processes
  – implemented using ports
Multiplexing/demultiplexing

**multiplexing as sender:**
handle data from multiple sockets, add transport header (later used for demultiplexing)

**demultiplexing as receiver:**
use header info to deliver received segments to correct socket
How demultiplexing Works

- host receives IP datagrams
  - each datagram has source IP address, destination IP address
  - each datagram carries one transport-layer segment
  - each segment has source, destination port number
- host uses **IP addresses & port numbers** to direct segment to appropriate socket
Connectionless demultiplexing

- when creating socket, must specify *host-local* port #:
  
  ```java
  DatagramSocket mySocket1 = new DatagramSocket(12534);
  ```

- when creating datagram to send into UDP socket, must specify
  - destination IP address
  - destination port #

when receiving host receives *UDP* segment:
- checks destination port # in segment
- directs UDP segment to socket with that port #

IP/UDP datagrams with *same dest. port #*, but different source IP addresses and/or source port numbers will be directed to *same socket* at receiving host.
Connectionless demultiplexing: an example

```
mySocket = socket(AF_INET,SOCK_STREAM)
mySocket.bind(myaddr,9157);
mySocket = socket(AF_INET,SOCK_DGRAM)
mySocket.bind(myaddr,6428);
mySocket = socket(AF_INET,SOCK_STREAM)
mySocket.bind(myaddr,5775);
```

A
source port: 9157
dest port: 6428
source port: 6428
dest port: 9157

B
source port: 6428
dest port: 9157

C
source port: ?
dest port: ?

D
source port: ?
dest port: ?
Connection-oriented demultiplexing

- TCP socket identified by 4-tuple:
  - source IP address
  - source port number
  - dest IP address
  - dest port number

- demux: receiver uses all four values (4-tuple) to direct segment to appropriate socket

- server may support many simultaneous TCP sockets:
  - each socket identified by its own 4-tuple
  - each socket associated with a different connecting client

Slight lie alert…. I should say that a common network tuple has FIVE values

- source IP address
- source port number
- dest IP address
- dest port number AND
- protocol e.g. TCP (6) or UDP (17)
Three segments, all destined to IP address: B, dest port: 80 are demultiplexed to different sockets
Multiplexing, demultiplexing: based on segment, datagram header field values

**UDP:** demultiplexing using destination port number (only)

**TCP:** demultiplexing using 4-tuple: source and destination IP addresses, and port numbers

Multiplexing/demultiplexing can happen at *any* layer
More on Ports

• Separate 16-bit port address space for UDP and TCP

• “Well known” ports (0-1023): everyone agrees which services run on these ports
  – e.g., ssh:22, http:80, https:443
  – helps client know server’s port

• Ephemeral ports (most 1024-65535): dynamically selected: as the source port for a client process
UDP: User Datagram Protocol

• Lightweight communication between processes
  – Avoid overhead and delays of ordered, reliable delivery

• UDP described in RFC 768 – (1980!)
  – Destination IP address and port to support demultiplexing
  – Optional error checking on the packet contents
    • (checksum field of 0 means “don’t verify checksum”) not in IPv6!
    • ((this idea of optional checksum is removed in IPv6))

<table>
<thead>
<tr>
<th>SRC port</th>
<th>DST port</th>
</tr>
</thead>
<tbody>
<tr>
<td>checksum</td>
<td>length</td>
</tr>
</tbody>
</table>

DATA
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 (mux/demux)

• IP provides a weak service model (*best-effort*)
  – Packets can be corrupted, delayed, dropped, reordered, duplicated
Principles of Reliable data transfer

• important in app., transport, link layers
• top-10 list of important networking topics!

- In a perfect world, reliable transport is easy

But the Internet default is best-effort

- All the bad things best-effort can do
  - a packet is corrupted (bit errors)
  - a packet is lost
  - a packet is delayed (why?)
  - packets are reordered (why?)
  - a packet is duplicated (why?)
Principles of Reliable data transfer

- important in app., transport, link layers
- top-10 list of important networking topics!
Principles of Reliable data transfer

- important in app., transport, link layers
- top-10 list of important networking topics!

Complexity of reliable data transfer protocol will depend (strongly) on characteristics of unreliable channel (lose, corrupt, reorder data?)
Reliable data transfer: getting started

**send side**
- **rdt_send()**: called from above, (e.g., by app.). Passed data to deliver to receiver upper layer
- **udt_send()**: called by rdt to transfer packet over unreliable channel to receiver

**receive side**
- **rdt_rcv()**: called by rdt to deliver data to upper
- **udt_rcv()**: called when packet arrives on rcv-side of channel
Reliable data transfer: getting started

We’ll:
• incrementally develop sender, receiver sides of reliable data transfer protocol (rdt)
• consider only unidirectional data transfer
  – but control info will flow on both directions!
• use finite state machines (FSM) to specify sender, receiver

state: when in this “state” next state uniquely determined by next event

```plaintext
state 1

event causing state transition
actions taken on state transition

state 2

event
actions```
KR state machines – a note.

Beware
Kurose and Ross has a confusing/confused attitude to state-machines.
I’ve attempted to normalise the representation.
UPSHOT: these slides have differing information to the KR book (from which the RDT example is taken.)
in KR “actions taken” appear wide-ranging, my interpretation is more specific/relevant.

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**Table:**

<table>
<thead>
<tr>
<th>State name</th>
<th>Relevant event causing state transition</th>
<th>Relevant action taken on state transition</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>state: when in this “state” next state uniquely determined by next event</td>
<td>event actions</td>
</tr>
</tbody>
</table>
Rdt1.0: reliable transfer over a reliable channel

• underlying channel perfectly reliable
  – no bit errors
  – no loss of packets

• separate FSMs for sender, receiver:
  – sender sends data into underlying channel
  – receiver read data from underlying channel
Rdt2.0: channel with bit errors

• underlying channel may flip bits in packet
  – checksum to detect bit errors

• the question: how to recover from errors:
  – acknowledgements (ACKs): receiver explicitly tells sender that packet received is OK
  – negative acknowledgements (NAKs): receiver explicitly tells sender that packet had errors
  – sender retransmits packet on receipt of NAK

• new mechanisms in rdt2.0 (beyond rdt1.0):
  – error detection
  – receiver feedback: control msgs (ACK,NAK) receiver->sender
rdt2.0: FSM specification

**sender**

- \[ \text{rdt\_send(data)} \]
- \[ \text{udt\_send(packet)} \]

**Waiting for reply**

- \[ \text{udt\_rcv(reply)} \land \text{isNAK(reply)} \]
- \[ \text{udt\_send(packet)} \]

- \[ \text{udt\_rcv(reply)} \land \text{isACK(reply)} \]

**receiver**

- \[ \text{udt\_rcv(packet)} \land \text{corrupt(packet)} \]
- \[ \text{udt\_send(NAK)} \]

- \[ \text{udt\_rcv(packet)} \land \text{notcorrupt(packet)} \]
- \[ \text{rdt\_rcv(data)} \]
- \[ \text{udt\_send(ACK)} \]

**Note:** the sender holds a copy of the packet being sent until the delivery is acknowledged.
rdt2.0: operation with no errors

- **IDLE**
  - `rdt_send(data)`
  - `udt_send(packet)`

- **Waiting for reply**
  - `udt_rcv(reply) && isNAK(reply)`
  - `udt_send(packet)`

- **IDLE**
  - `udt_rcv(reply) && isACK(reply)`
  - `udt_send(ACK)`

- **IDLE**
  - `udt_rcv(NAK)`

- **IDLE**
  - `udt_rcv(packet) && notcorrupt(packet)`
  - `rdt_rcv(data)`
  - `udt_send(ACK)}`
rdt2.0: error scenario

1. `rdt_send(data)`
2. `udt_send(packet)`
3. `udt_rcv(reply) && isNAK(reply)`
4. `udt_send(packet)`
5. `udt_rcv(packet) && corrupt(packet)`
6. `udt_send(NAK)`
7. `udt_rcv(packet) && notcorrupt(packet)`
8. `rdt_rcv(data)`
9. `udt_send(ACK)`
10. `udt_rcv(reply) && isACK(reply)`
11. `Lambda`
rdt2.0 has a fatal flaw!

What happens if ACK/NAK corrupted?
• sender doesn’t know what happened at receiver!
• can’t just retransmit: possible duplicate

Handling duplicates:
• sender retransmits current packet if ACK/NAK garbled
• sender adds sequence number to each packet
• receiver discards (doesn’t deliver) duplicate packet

stop and wait
Sender sends one packet, then waits for receiver response
rdt2.1: sender, handles garbled ACK/NAKs

- **IDLE**
  - `udt_send(packet)`
  - `udt_rcv(reply) && notcorrupt(reply) && isACK(reply)`

- **Waiting For reply**
  - `udt_rcv(reply) && (corrupt(reply) || isNAK(reply))`
  - `udt_send(packet)`

- **Waiting for reply**
  - `udt_rcv(reply) && (corrupt(reply) || isNAK(reply))`

- **rdt_send(data)**
  - `sequence=0`
rdt2.1: receiver, handles garbled ACK/NAKs

- receive(packet) && corrupt(packet) 
  udt_send(NAK)

- receive(packet) && not corrupt(packet) && has_seq1(packet) 
  udt_send(ACK)

- udt_rcv(packet) && not corrupt(packet) && has_seq0(packet) 
  udt_send(ACK)
  rdt_rcv(data)

- receive(packet) && not corrupt(packet) && has_seq0(packet) 
  udt_send(ACK)
  rdt_rcv(data)

- receive(packet) && not corrupt(packet) && has_seq1(packet) 
  udt_send(ACK)

- udt_send(ACK)
  rdt_rcv(data)

Wait for 0 from below
Wait for 1 from below
rdt2.1: discussion

**Sender:**
- seq # added to pkt
- two seq. #’s (0,1) will suffice. Why?
- must check if received ACK/NAK corrupted
- twice as many states
  - state must “remember” whether “current” pkt has a 0 or 1 sequence number

**Receiver:**
- must check if received packet is duplicate
  - state indicates whether 0 or 1 is expected pkt seq #
- note: receiver can *not* know if its last ACK/NAK received OK at sender
rdt2.2: a NAK-free protocol

• same functionality as rdt2.1, using ACKs only
• instead of NAK, receiver sends ACK for last pkt received OK
  – receiver must explicitly include seq # of pkt being ACKed
• duplicate ACK at sender results in same action as NAK: *retransmit current pkt*

As we will see, TCP uses this approach to be NAK-free
**rdt2.2: sender, receiver fragments**

**sender FSM fragment**
- `rdt_send(data)`
  - `sequence=0`
  - `udt_send(packet)`
- `Wait for call 0 from above`
- `rdt_rcv(reply) && (corrupt(reply) || isACK1(reply))`
  - `udt_send(packet)`

**receiver FSM fragment**
- `udt_rcv(packet) && (corrupt(packet) || has_seq1(packet))`
- `udt_send(ACK1)`
- `receive(packet) && not corrupt(packet)`
  - `&& has_seq1(packet)`
  - `send(ACK1)`
- `rdt_rcv(data)`
- `Lambda`
rdt3.0: channels with errors \textit{and} loss

\textit{New channel assumption}: underlying channel can also lose packets (data, ACKs)
  \begin{itemize}
    \item checksum, sequence #s, ACKs, retransmissions will be of help …
    \item but not quite enough
  \end{itemize}

\textbf{Q}: How do \textit{humans} handle lost sender-to-receiver words in conversation?
rdt3.0: channels with errors and loss

Approach: sender waits “reasonable” amount of time for ACK

- retransmits if no ACK received in this time
- if pkt (or ACK) just delayed (not lost):
  - retransmission will be duplicate, but seq #s already handles this!
  - receiver must specify seq # of packet being ACKed
- use countdown timer to interrupt after “reasonable” amount of time
rdt3.0 sender

- **IDLE state 0**
  - `udt_rcv(reply)`
  - `udt_send(packet)`
  - `start_timer`

- **Wait for ACK0**
  - `udt_rcv(reply) && notcorrupt(reply) && isACK(reply,1)`
  - `stop_timer`
  - `udt_send(packet)`

- **Wait for ACK1**
  - `timeout`
  - `udt_send(packet)`
  - `udt_rcv(packet) && (corrupt(packet) || isACK(reply,0))`
  - `start_timer`

- **IDLE state 1**
  - `udt_rcv(reply)`
  - `udt_send(packet)`
  - `timeout`

- **Wait for ACK0**
  - `udt_send(data)`
  - `sequence=0`

- **Wait for ACK1**
  - `udt_send(data)`
  - `sequence=1`
rdt3.0 in action
rdt3.0 in action

**sender**
- send pkt0
- rcv ptk0 send ack0
- rcv ptk1 send ack1
- timeout resend ptk1
- rcv ack1 send pkt0
- ACK loss

**receiver**
- send pkt0
- rcv pkt0 send ack0
- rcv pkt1 send ack1
- (detect duplicate) send ack1
- rcv ptk0 send ack0

**sender**
- send ptk0
- rcv ptk0 send ack0
- rcv ptk1 send ack1
- timeout resend ptk1
- rcv ack1 send ptk0
- (ignore)

**receiver**
- rcv pkt0 send ack0
- rcv pkt1 send ack1
- rcv pkt1 (detect duplicate) send ack1
- rcv ptk0 send ack0
- premature timeout/ delayed ACK
rdt3.0: stop-and-wait operation

\[
U_{\text{sender}} = \frac{L / R}{RTT + L / R} = \frac{0.008}{30.008} = 0.00027
\]
Performance of rdt3.0 (stop-and-wait)

• rdt3.0 works, but performance stinks
• ex: 1 Gbps link, 15 ms prop. delay, 8000 bit packet:

\[ d_{trans} = \frac{L}{R} = \frac{8000\text{bits}}{10^9\text{bps}} = 8\text{microseconds} \]

• \( U_{\text{sender}} \): utilization – fraction of time sender busy sending

\[ U_{\text{sender}} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027 \]

• 1KB pkt every 30 msec -> 33kB/sec throughput over 1 Gbps link

• The network protocol limits use of physical resources!
Pipelined (Packet-Window) protocols

Pipelining: sender allows multiple, “in-flight”, yet-to-be-acknowledged pkts
  – range of sequence numbers must be increased
  – buffering at sender and/or receiver
Pipelining: increased utilization

\[ U_{\text{sender}} = \frac{3L / R}{RTT + L / R} = \frac{.0024}{30.008} = 0.00081 \]
A Sliding Packet 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
Acknowledgements w/ Sliding Window

• Two common options
  – cumulative ACKs: ACK carries next in-order sequence number that the receiver expects
Cumulative Acknowledgements (1)

- At receiver

- After receiving B+1, B+2

- Receiver sends ACK(B\textsubscript{new}+1)
Cumulative Acknowledgements (2)

- At receiver

- After receiving B+4, B+5

- Receiver sends \text{ACK}(B+???)

Oh.... how do we recover?
Go-Back-N: sender

- **sender**: “window” of up to N, consecutive transmitted but unACKed pkts
  - k-bit seq # in pkt header

- **cumulative ACK**: ACK($n$): ACKs all packets up to, including seq # $n$
  - on receiving ACK($n$): move window forward to begin at $n+1$

- timer for oldest in-flight packet

- **timeout($n$)**: retransmit packet $n$ and all higher seq # packets in window
Go-Back-N: receiver

- **ACK-only**: always send ACK for correctly-received packet so far, with highest *in-order* seq #
  - may generate duplicate ACKs
  - need only remember `rcv_base`
- **on receipt of out-of-order packet:**
  - can discard (don’t buffer) or buffer: an implementation decision
  - re-ACK pkt with highest in-order seq #

Receiver view of sequence number space:

- **received and ACKed**
- **Out-of-order: received but not ACKed**
- **Not received**
Go-Back-N in action

sender window (N=4)

sender

receiver

send pkt0, send ack0
receive pkt0, send ack0

send pkt1, send ack1
receive pkt1, send ack1

send pkt2, (wait)
rcv ack0, send pkt4
rcv pkt4, discard, (re)send ack1

send pkt3
rcv ack1, send pkt5
rcv pkt5, discard, (re)send ack1

receiver pkt0, send ack0
Receive pkt1, send ack1

receiver pkt3, discard, (re)send ack1
Receive pkt4, discard, (re)send ack1

receiver pkt2 timeout
Ignore duplicate ACK

receiver pkt2, deliver, send ack2
rcv pkt3, deliver, send ack3
rcv pkt4, deliver, send ack4
rcv pkt5, deliver, send ack5
Selective repeat

- receiver *individually* acknowledges all correctly received packets
  - buffers packets, as needed, for eventual in-order delivery to upper layer
- sender times-out/retransmits individually for unACKed packets
  - sender maintains timer for each unACKed pkt
- sender window
  - $N$ consecutive seq #s
  - limits seq #s of sent, unACKed packets
Selective repeat: sender, receiver windows

(a) sender view of sequence numbers
Selective repeat: sender and receiver

**Sender**

- Data from above:
  - if next available seq # in window, send packet

- Timeout($n$):
  - resend packet $n$, restart timer

- ACK($n$) in [sendbase, sendbase+N-1]:
  - mark packet $n$ as received
  - if $n$ smallest unACKed packet, advance window base to next unACKed seq #

**Receiver**

- Packet $n$ in [rcvbase, rcvbase+N-1]
  - send ACK($n$)
  - out-of-order: buffer
  - in-order: deliver (also deliver buffered, in-order packets), advance window to next not-yet-received packet

- Packet $n$ in [rcvbase-N, rcvbase-1]
  - ACK($n$)

- Otherwise:
  - ignore
Selective Repeat in action

**sender window (N=4)**

0 1 2 3 4 5 6 7 8
0 1 2 3 4 5 6 7 8
0 1 2 3 4 5 6 7 8
0 1 2 3 4 5 6 7 8

**sender**

- send pkt0
- send pkt1
- send pkt2
- send pkt3 (wait)

**receiver**

- receive pkt0, send ack0
- receive pkt1, send ack1
- receive pkt3, buffer, send ack3
- receive pkt4, buffer, send ack4
- receive pkt5, buffer, send ack5

**pkt 2 timeout**

send pkt2 (but not 3,4,5)

**Q: what happens when ack2 arrives?**

- rcv pkt2; deliver pkt2, pkt3, pkt4, pkt5; send ack2

- record ack3 arrived
- rcv ack0, send pkt4
- rcv ack1, send pkt5
Selective repeat: a dilemma!

example:
- seq #s: 0, 1, 2, 3 (base 4 counting)
- window size=3

(a) no problem

(b) oops!
Selective repeat: a dilemma!

example:
- seq #s: 0, 1, 2, 3 (base 4 counting)
- window size=3

Q: what relationship is needed between sequence # size and window size to avoid problem in scenario (b)?

Solution:

maximum allowable window size = half the sequence number space.
Observations

• With sliding windows, it is possible to fully utilize a link, provided the window size \( (n) \) is large enough. Throughput is \( \sim (n/RTT) \)
  – Stop & Wait is like \( n = 1 \).

• 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)
Recap: 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
What does TCP do?

Most of our previous tricks + a few more beside

- Sequence numbers are byte offsets
- Sender and receiver maintain a sliding window
- Receiver sends cumulative acknowledgements (like GBN)
- Sender maintains a single retransmit timer
- Receivers do not drop out-of-sequence packets (like SR)
- Introduces fast retransmit: optimization that uses duplicate ACKs to trigger early retransmit
- Introduces timeout estimation algorithms
TCP: overview  

RFCs: 793, 1122, 2018, 5681, 7323

- **point-to-point:**
  - one sender, one receiver

- **reliable, in-order *byte steam:***
  - no “message boundaries"

- **full duplex data:**
  - bi-directional data flow in same connection
  - MSS: maximum segment size

- **cumulative ACKs

- **pipelining:**
  - TCP congestion and flow control
  - set window size

- **connection-oriented:**
  - handshaking (exchange of control messages) initializes sender, receiver state before data exchange

- **flow controlled:**
  - sender will not overwhelm receiver
### TCP Header

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Source port</td>
<td>Source port</td>
</tr>
<tr>
<td>Destination port</td>
<td>Destination port</td>
</tr>
<tr>
<td>Sequence number</td>
<td>Sequence number</td>
</tr>
<tr>
<td>Acknowledgment</td>
<td>Acknowledgment</td>
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<tr>
<td>HdrLen</td>
<td>HdrLen</td>
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<tr>
<td>Flags</td>
<td>Flags</td>
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<tr>
<td>Advertised window</td>
<td>Advertised window</td>
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<tr>
<td>Checksum</td>
<td>Checksum</td>
</tr>
<tr>
<td>Urgent pointer</td>
<td>Urgent pointer</td>
</tr>
<tr>
<td>Options (variable)</td>
<td>Options (variable)</td>
</tr>
<tr>
<td>Data</td>
<td>Data</td>
</tr>
</tbody>
</table>

Used to multiplex and demultiplex.
What does TCP do?

Many of our previous ideas, but some key differences

• Checksum
TCP Header

<table>
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<th>Destination port</th>
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Computed over header and data
What does TCP do?

Many of our previous ideas, but some key differences

• Checksum
• Sequence numbers are byte offsets
TCP: Segments and Sequence Numbers
TCP “Stream of Bytes” Service...

Application @ Host A

Application @ Host B
… Provided Using TCP “Segments”

Segment sent when:
1. Segment full (Max Segment Size),
2. Not full, but times out
TCP Segment

- IP packet
  - No bigger than Maximum Transmission Unit (MTU)
  - E.g., up to 1500 bytes with Ethernet
- TCP packet
  - IP packet with a TCP header and data inside
  - TCP header ≥ 20 bytes long
- TCP segment
  - No more than Maximum Segment Size (MSS) bytes
  - E.g., up to 1460 consecutive bytes from the stream
  - MSS = MTU – (IP header) – (TCP header)
Sequence Numbers

ISN (initial sequence number)

Host A

Sequence number = 1\textsuperscript{st} byte in segment = ISN + k
**Sequence Numbers**

**ISN (initial sequence number)**

- Host A
- Host B

Sequence number = 1st byte in segment = ISN + k

ACK sequence number = next expected byte = seqno + length(data)
TCP Header

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Sequence number

Acknowledgment

HdrLen | Flags | Advertised window

Checksum | Urgent pointer

Options (variable)

Data

Starting byte offset of data carried in this segment
Sequence Numbers

Sequence number = 1st byte in segment = ISN + k

Host A

Host B

Host A - > B
DATA

Host B - > A
ACK

ACK sequence number = next expected byte = seqno + length(data)
TCP Sequences and ACKS

TCP is full duplex by default
  • two independently flows of sequence numbers

Sequence acknowledgement is given in terms of BYTES (not packets); the window is in terms of bytes.

number of packets = window size (bytes) / Segment Size

Servers and Clients are not Source and Destination

Piggybacking increases efficiency but many flows may only have data moving in one direction
What does TCP do?

Most of our previous tricks, but a few differences

- Checksum
- Sequence numbers are byte offsets
- Receiver sends cumulative acknowledgements (like GBN)
ACKing and Sequence Numbers

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

- **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)
  - If highest in-order byte received is $Y$ s.t. $(Y+1) < X$
    - ACK acknowledges $Y+1$
    - Even if this has been ACKed before
Normal Pattern

- Sender: seqno=X, length=B
- Receiver: ACK=X+B
- Sender: seqno=X+B, length=B
- Receiver: ACK=X+2B
- Sender: seqno=X+2B, length=B

- Seqno of next packet is same as last ACK field
### TCP Header

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Acknowledgment gives seqno just beyond highest seqno received in order ("What Byte is Next")
What does TCP do?

Most of our previous tricks, but a few differences

• Checksum
• Sequence numbers are byte offsets
• Receiver sends cumulative acknowledgements (like GBN)
• Receivers can buffer out-of-sequence packets (like SR)
Loss with cumulative ACKs

• Sender sends packets with 100B and seqnos.:  
  – 100, 200, 300, 400, 500, 600, 700, 800, 900, ...

• Assume the fifth packet (seqno 500) is lost, but no others

• Stream of ACKs will be:  
  – 200, 300, 400, 500, 500, 500, 500,...
What does TCP do?

Most of our previous tricks, but a few differences

- Checksum
- Sequence numbers are byte offsets
- Receiver sends cumulative acknowledgements (like GBN)
- Receivers may not drop out-of-sequence packets (like SR)
- Introduces fast retransmit: optimization that uses duplicate ACKs to trigger early retransmission
Loss with cumulative ACKs

• “Duplicate ACKs” are a sign of an isolated loss
  – The lack of ACK progress means 500 hasn’t been delivered
  – Stream of ACKs means some packets are being delivered

• Therefore, could trigger resend upon receiving k duplicate ACKs
  • TCP uses k=3

• But response to loss is trickier....
Loss with cumulative ACKs

• Two choices:
  – Send missing packet and increase $W$ by the number of dup ACKs
  – Send missing packet, and wait for ACK to increase $W$

• Which should TCP do?
What does TCP do?

Most of our previous tricks, but a few differences

- Checksum
- Sequence numbers are byte offsets
- Receiver sends cumulative acknowledgements (like GBN)
- Receivers do not drop out-of-sequence packets (like SR)
- Introduces fast retransmit: optimization that uses duplicate ACKs to trigger early retransmission
- Sender maintains a single retransmission timer (like GBN) and retransmits on timeout
Retransmission Timeout

• If the sender hasn’t received an ACK by timeout, retransmit the first packet in the window

• How do we pick a timeout value?
Timing Illustration

Timeout too long $\rightarrow$ inefficient

Timeout too short $\rightarrow$ duplicate packets
Retransmission Timeout

- If haven’t received ack by timeout, retransmit the first packet in the window

- How to set timeout?
  - Too long: connection has low throughput
  - Too short: retransmit packet that was just delayed

- Solution: make timeout proportional to RTT

- But how do we measure RTT?
RTT Estimation

• Use exponential averaging of RTT samples

\[ SampleRTT = AckRcvdTime - SendPacketTime \]
\[ EstimatedRTT = \alpha \times EstimatedRTT + (1 - \alpha) \times SampleRTT \]
\[ 0 < \alpha \leq 1 \]
Exponential Averaging Example

\[ \text{EstimatedRTT} = \alpha \times \text{EstimatedRTT} + (1 - \alpha) \times \text{SampleRTT} \]

Assume RTT is constant \( \rightarrow \text{SampleRTT} = \text{RTT} \)

![Graph showing exponential averaging example with \( \alpha = 0.5 \) and \( \alpha = 0.8 \)]
Problem: Ambiguous Measurements

- How do we differentiate between the real ACK, and ACK of the retransmitted packet?
Karn/Partridge Algorithm
Discard junk measures

• Measure SampleRTT only for original transmissions
  – Once a segment has been retransmitted, do not use it for any further measurements
• Computes EstimatedRTT using $\alpha = 0.875$

• Timeout value (RTO) = $2 \times$ EstimatedRTT
• Employs exponential backoff
  – Every time RTO timer expires, set RTO $\leftarrow 2 \cdot$RTO
  – (Up to maximum $\geq 60$ sec)
  – Every time new measurement comes in (= successful original transmission), collapse RTO back to $2 \times$ EstimatedRTT
Jacobson/Karels Algorithm
Add a safety margin

• Problem: need to better capture variability in RTT
  – Directly measure deviation

  • Deviation = | SampleRTT – EstimatedRTT |
  • EstimatedDeviation: exponential average of Deviation

  • RTO = EstimatedRTT + 4 x EstimatedDeviation
What does TCP do?

Most of our previous ideas, but some key differences

- Checksum
- Sequence numbers are byte offsets
- Receiver sends cumulative acknowledgements (like GBN)
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- **“Must Be Zero”**
  - 6 bits reserved
- Number of 4-byte words in TCP header; 5 = no options

- 
  - HdrLen: 0
  - Flags:
  - Checksum
  - Urgent pointer:
TCP Header: What’s left?

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Data

Used with **URG** flag to indicate urgent data (not discussed further)
TCP Header: What’s left?

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TCP Connection Establishment and Initial Sequence Numbers
Initial Sequence Number (ISN)

• Sequence number for the very first byte
• Why not just use ISN = 0?
• Practical issue
  – IP addresses and port #s uniquely identify a connection
  – Eventually, though, these port #s do get used again
  – … small chance an old packet is still in flight
• TCP therefore requires changing ISN
• Hosts exchange ISNs when they establish a connection
Establishing a TCP Connection

- Three-way handshake to establish connection
  - Host A sends a **SYN** (open; “synchronize sequence numbers”) to host B
  - Host B returns a SYN acknowledgment (**SYN ACK**)
  - Host A sends an **ACK** to acknowledge the SYN ACK

Each host tells its ISN to the other host.
### TCP Header

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**Flags:**
- **SYN**
- **ACK**
- **FIN**
- **RST**
- **PSH**
- **URG**
### Step 1: A’s Initial SYN Packet

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<td>FIN</td>
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<td>RST</td>
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</tr>
<tr>
<td>URG</td>
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</tr>
<tr>
<td>A’s Initial Sequence Number</td>
<td></td>
<td>(Irrelevant since ACK not set)</td>
</tr>
<tr>
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<td>5</td>
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A tells B it wants to open a connection…
Step 2: B’s SYN-ACK Packet

B tells A it accepts, and is ready to hear the next byte…

… upon receiving this packet, A can start sending data
Step 3: A’s ACK of the SYN-ACK

A’s port | B’s port
---|---
A’s Initial Sequence Number | B’s ISN plus 1
20B | 0
Flags | Advertised window
Checksum | Urgent pointer
Options (variable)

Flags: SYN ACK FIN RST PSH URG

A tells B it’s likewise okay to start sending

... upon receiving this packet, B can start sending data
Timing Diagram: 3-Way Handshaking

Client (initiator)

Active
Open

connect()

Server

Passive
Open

listen()

SYN, SeqNum = x

SYN + ACK, SeqNum = y, Ack = x + 1

ACK, Ack = y + 1
What if the SYN Packet Gets Lost?

- Suppose the SYN packet gets lost
  - Packet is lost inside the network, or:
  - Server discards the packet (e.g., it’s too busy)

- Eventually, no SYN-ACK arrives
  - Sender sets a timer and waits for the SYN-ACK
  - … and retransmits the SYN if needed

- How should the TCP sender set the timer?
  - Sender has no idea how far away the receiver is
  - Hard to guess a reasonable length of time to wait
  - SHOULD (RFCs 1122 & 2988) use default of 3 seconds
    - Some implementations instead use 6 seconds
Tearing Down the Connection
Normal Termination, One Side At A Time

- Finish (**FIN**) to close and receive remaining bytes
  - **FIN** occupies one byte in the sequence space
- Other host acks the byte to confirm
- Closes A’s side of the connection, but **not** B’s
  - Until B likewise sends a **FIN**
  - Which A then acks

**TIME_WAIT:**
- Avoid reincarnation
- B will retransmit FIN if ACK is lost
Normal Termination, Both Together

- Same as before, but B sets **FIN** with their ack of A’s **FIN**

**TIME_WAIT:**
Avoid reincarnation
Can retransmit FIN ACK if ACK lost

Connection now closed
Abrupt Termination

- A sends a RESET (RST) to B
  - E.g., because application process on A crashed
- That’s it
  - B does not ack the RST
  - Thus, RST is not delivered reliably
  - And: any data in flight is lost
  - But: if B sends anything more, will elicit another RST
TCP State Transitions

Data, ACK exchanges are in here
An Simpler View of the Client Side

- **CLOSED**
  - SYN (Send)

- **SYN_SENT**
  - Rcv. SYN+ACK, Send ACK

- **ESTABLISHED**
  - Send FIN

- **FIN_WAIT1**
  - Rcv. ACK, Send Nothing

- **FIN_WAIT2**
  - Rcv. FIN, Send ACK

- **TIME_WAIT**
TCP Header

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• What does TCP do?
  – ARQ windowing, set-up, tear-down
• Flow Control in TCP
**Q:** What happens if network layer delivers data faster than application layer removes data from socket buffers?
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**Q:** What happens if network layer delivers data faster than application layer removes data from socket buffers?

**TCP flow control**

- **Application process**
- **TCP socket receiver buffers**
- **TCP code**
- **IP code**

- Flow control: # bytes receiver willing to accept

- Application removing data from TCP socket buffers

- Receive window

- From sender

**Receiver protocol stack**
**Q:** What happens if network layer delivers data faster than application layer removes data from socket buffers?

**Flow control:**
receiver controls sender, so sender won’t overflow receiver’s buffer by transmitting too much, too fast.
TCP flow control

- TCP receiver “advertises” free buffer space in **rwnd** field in TCP header
  - **RcvBuffer** size set via socket options (typical default is 4096 bytes)
  - many operating systems autoadjust **RcvBuffer**
- sender limits amount of unACKed ("in-flight") data to received **rwnd**
- guarantees receive buffer will not overflow
TCP flow control

- TCP receiver “advertises” free buffer space in `rwnd` field in TCP header
  - `RcvBuffer` size set via socket options (typical default is 4096 bytes)
  - many operating systems autoadjust `RcvBuffer`
- sender limits amount of unACKed (“in-flight”) data to received `rwnd`
- guarantees receive buffer will not overflow
• What does TCP do?
  – ARQ windowing, set-up, tear-down
• Flow Control in TCP
• Congestion Control in TCP
We have seen:

- **Flow control**: adjusting the sending rate to keep from overwhelming a slow *receiver*

Now let's attend...

- **Congestion control**: adjusting the sending rate to keep from overloading the *network*
Principles of congestion control

Congestion:

- informally: “too many sources sending too much data too fast for network to handle”
- manifestations:
  - long delays (queueing in router buffers)
  - packet loss (buffer overflow at routers)
- different from flow control!
- a top-10 problem!

congestion control: too many senders, sending too fast

flow control: one sender too fast for one receiver
Causes/costs of congestion: scenario 1

Simplest scenario:
- one router, infinite buffers
- input, output link capacity: \( R \)
- two flows
- no retransmissions needed

\[ Q: \text{What happens as arrival rate } \lambda_{\text{in}} \text{ approaches } R/2? \]

maximum per-connection throughput: \( R/2 \)

large delays as arrival rate \( \lambda_{\text{in}} \) approaches capacity
Causes/costs of congestion: scenario 2

- one router, *finite* buffers
- sender retransmits lost, timed-out packet
  - application-layer input = application-layer output: \( \lambda_{in} = \lambda_{out} \)
  - transport-layer input includes *retransmissions*: \( \lambda'_{in} \geq \lambda_{in} \)
Causes/costs of congestion: scenario 2

Idealization: perfect knowledge

- sender sends only when router buffers available

\[ \lambda_{in} : \text{original data} \]
\[ \lambda_{in}' : \text{original data, plus retransmitted data} \]

finite shared output link buffers

free buffer space!
Causes/costs of congestion: scenario 2

**Idealization:** *some* perfect knowledge

- packets can be lost (dropped at router) due to full buffers
- sender knows when packet has been dropped: only resends if packet *known* to be lost

![Diagram of network with Host A and Host B, showing original data and retransmitted data]

- \( \lambda_{in} \): original data
- \( \lambda_{in}' \): original data, plus retransmitted data

**Finite shared output link buffers**

**No buffer space!**
Causes/costs of congestion: scenario 2

Idealization: *some* perfect knowledge

- packets can be lost (dropped at router) due to full buffers
- sender knows when packet has been dropped: only resends if packet known to be lost

![Diagram showing a network with Host A and Host B communicating through a shared buffer. The diagram includes symbols for the rate of incoming data ($\lambda_{in}$), the rate of transmitted data ($\lambda_{r,in}$), free buffer space, and finite shared output link buffers. The graph shows throughput as a function of input rate ($\lambda_{in}$) with a point indicating "wasted" capacity due to retransmissions when sending at $R/2$, some packets are needed for retransmissions.](image-url)
Causes/costs of congestion: scenario 2

Realistic scenario: *un-needed duplicates*

- packets can be lost, dropped at router due to full buffers – requiring retransmissions
- but sender times can time out prematurely, sending *two* copies, *both* of which are delivered

```
Host A
```

```
Host B
```

```
finite shared output link buffers
```

```
\lambda_{in}: original data
\lambda'_{in}: original data, plus retransmitted data
```

```
\text{timeout}
```

```
\text{free buffer space!}
```

```
\text{R/2}
```

```
\text{throughput:}
```

```
\lambda_{in} \rightarrow R/2
```

```
“wasted” capacity due to un-needed retransmissions
```

```
when sending at R/2, some packets are retransmissions, including needed and un-needed duplicates, that are delivered!
```

R/2
Causes/costs of congestion: scenario 2

Realistic scenario: *un-needed duplicates*
- packets can be lost, dropped at router due to full buffers – requiring retransmissions
- but sender times can time out prematurely, sending *two* copies, *both* of which are delivered

“costs” of congestion:
- more work (retransmission) for given receiver throughput
- unneeded retransmissions: link carries multiple copies of a packet
  - decreasing maximum achievable throughput

\[ \text{throughput: } \lambda_{\text{in}} \rightarrow \text{R/2} \]

“wasted” capacity due to un-needed retransmissions when sending at R/2, some packets are retransmissions, including needed and *un-needed duplicates*, that are delivered!
Causes/costs of congestion: scenario 3

- *four senders*
- *multi-hop paths*
- *timeout/retransmit*

**Q:** what happens as \( \lambda_{in} \) and \( \lambda_{in}' \) increase?

**A:** as red \( \lambda_{in}' \) increases, all arriving blue pkts at upper queue are dropped, blue throughput \( \rightarrow 0 \)
another “cost” of congestion:

- when packet dropped, any upstream transmission capacity and buffering used for that packet was wasted!
Causes/costs of congestion: insights

- throughput can never exceed capacity
- delay increases as capacity approached
- loss/retransmission decreases effective throughput
- un-needed duplicates further decreases effective throughput
- upstream transmission capacity / buffering wasted for packets lost downstream
Approaches towards congestion control

End-end congestion control:
- no explicit feedback from network
- congestion *inferred* from observed loss, delay
  - approach taken by TCP
Approaches towards congestion control

Network-assisted congestion control:

- routers provide *direct* feedback to sending/receiving hosts with flows passing through congested router
- may indicate congestion level or explicitly set sending rate

- TCP ECN, ATM, DECBit protocols
Three Issues to Consider

• Discovering the available (bottleneck) bandwidth

• Adjusting to variations in bandwidth

• Sharing bandwidth between flows
• Ignore internal structure of router and model it as having a single queue for a particular input-output pair
Discovering available bandwidth

- Pick sending rate to match bottleneck bandwidth
  - Without any *a priori* knowledge
  - Could be gigabit link, could be a modem
Adjusting to variations in bandwidth

- Adjust rate to match **instantaneous** bandwidth
  - Assuming you have rough idea of bandwidth
Multiple flows and sharing bandwidth

Two Issues:
• Adjust total sending rate to match bandwidth
• Allocation of bandwidth between flows
Reality

Congestion control is a resource allocation problem involving many flows, many links, and complicated global dynamics.
View from a single flow

- **Knee** – point after which
  - Throughput increases slowly
  - Delay increases fast

- **Cliff** – point after which
  - Throughput starts to drop to zero (congestion collapse)
  - Delay approaches infinity
General Approaches

(0) Send without care
   – Many packet drops
General Approaches

(0) Send without care

(1) Reservations
   – Pre-arrange bandwidth allocations
   – Requires negotiation before sending packets
   – Low utilization
General Approaches

(0) Send without care
(1) Reservations
(2) Pricing
  – Don’t drop packets for the high-bidders
  – Requires payment model
General Approaches

(0) Send without care
(1) Reservations
(2) Pricing
(3) Dynamic Adjustment
   – Hosts probe network; infer level of congestion; adjust
   – Network reports congestion level to hosts; hosts adjust
   – Combinations of the above
   – Simple to implement but suboptimal, messy dynamics
General Approaches

(0) Send without care
(1) Reservations
(2) Pricing
(3) Dynamic Adjustment

All three techniques have their place

• Generality of dynamic adjustment has proven powerful
• Doesn’t presume business model, traffic characteristics, application requirements; does assume good citizenship
Who Takes Care of Congestion?

• Network? End hosts? Both?

• TCP’s approach:
  – End hosts adjust sending rate
  – Based on implicit feedback from network

• Not the only approach
  – A consequence of history rather than planning
Some History: TCP in the 1980s

• Sending rate only limited by flow control
  – Packet drops $\rightarrow$ senders (repeatedly!) retransmit a full window’s worth of packets

• Led to “congestion collapse” starting Oct. 1986
  – Throughput on the NSF network dropped from 32Kbits/s to 40bits/sec

• “Fixed” by Van Jacobson’s development of TCP’s congestion control (CC) algorithms
Jacobson’s Approach

• Extend TCP’s existing window-based protocol but adapt the window size in response to congestion
  – required no upgrades to routers or applications!
  – patch of a few lines of code to TCP implementations

• A pragmatic and effective solution
  – but many other approaches exist

• Extensively improved on since
  – topic now sees less activity in ISP contexts
  – but is making a comeback in datacenter environments
TCP’s Approach in a Nutshell

• TCP connection has window
  – Controls number of packets in flight

• Sending rate: ~Window/RTT

• Vary window size to control sending rate
Windows, Buffers, and TCP
Windows, Buffers, and TCP

• TCP connection has a window
  – Controls number of packets in flight;
    filling a channel to improve throughput, and
    vary window size to control sending rate

• Buffers adapt mis-matched channels
  – Buffers smooth bursts
  – Adapt (re-time) arrivals for multiplexing
Windows, Buffers, and TCP

Buffers & TCP can make link utilization 100%

but

Buffers add delay, **variable** delay
Sizing Buffers in Routers

– Packet loss
  • Queue overload, and subsequent packet loss

– End-to-end delay
  • Transmission, propagation, and queueing delay
  • The only variable part is queueing delay

– Router architecture
  • Board space, power consumption, and cost
  • On chip buffers: higher density, higher capacity
Buffer Sizing Story

<table>
<thead>
<tr>
<th># of packets</th>
<th>Rule-of-thumb</th>
<th>Intuition</th>
<th>Assume</th>
<th>Evidence</th>
</tr>
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<tbody>
<tr>
<td>1,000,000</td>
<td>2T × C</td>
<td>TCP Sawtooth</td>
<td>Single TCP Flow, 100% Utilization</td>
<td>Simulation, Emulation</td>
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Continuous ARQ (TCP) adapting to congestion

Only $W$ packets may be outstanding

Rule for adjusting $W$
- If an ACK is received: $W \leftarrow W + 1/W$
- If a packet is lost: $W \leftarrow W/2$
Continuous ARQ (TCP) adapting to congestion

Only $W$ packets may be outstanding

**Rule for adjusting $W$**
- If an ACK is received: $W \leftarrow W + 1/W$
- If a packet is lost: $W \leftarrow W/2$

$W = 1$

util = 0%

$W$

time
Rule-of-thumb – Intuition

Rule for adjusting $W$
- If an ACK is received: $W \leftarrow W + 1/W$
- If a packet is lost: $W \leftarrow W/2$

Only $W$ packets may be outstanding

Window size

$W_{\text{max}}$

$W_{\text{max}}/2$

$2T \times C$

$t$

Source

Dest

$2T \times C$
Buffers in Routers
So how large should the buffers be?

Buffer size matters

– Packet loss
  • Queue overload, and subsequent packet loss

– End-to-end delay
  • Transmission, propagation, and queueing delay
  • The only variable part is queueing delay
### Buffer Sizing Story

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<td>Simulations, Test-bed and Real Network Experiments</td>
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Buffers in Routers

So how large should the buffers be?

Buffer size matters

– Packet loss
  • Queue overload, and subsequent packet loss

– End-to-end delay
  • Transmission, propagation, and queueing delay
  • The only variable part is queueing delay

– Router architecture
  • Board space, power consumption, and cost
  • On chip buffers: higher density, higher capacity
Synchronized Flows

- Aggregate window has same dynamics
- Therefore buffer occupancy has same dynamics
- Rule-of-thumb still holds.

Many TCP Flows

- Independent, desynchronized
- Central limit theorem says the aggregate becomes Gaussian
- Variance (buffer size) decreases as N increases
Buffer Sizing Story

What size do we make the buffer? Well it depends…

One TCP connection?

Many Synchronized TCP connections?

Just TCP – what about other applications?

Small BDP link?

Large BDP link?

How many devices?

How many flows?

How much do you know about your traffic?

What is best for your traffic?
TCP’s Approach in a Nutshell

• TCP connection has window
  – Controls number of packets in flight

• Sending rate: ~Window/RTT

• Vary window size to control sending rate
All These Windows...

• Congestion Window: \textit{CWND}
  – How many bytes can be sent without overflowing routers
  – Computed by the sender using congestion control algorithm

• Flow control window: \textit{AdvertisedWindow (RWND)}
  – How many bytes can be sent without overflowing receiver’s buffers
  – Determined by the receiver and reported to the sender

• Sender-side window = \textit{minimum}\{\textit{CWND, RWND}\}
  • Assume for this material that RWND >> CWND
Note

• This lecture will talk about CWND in units of MSS
  – (Recall MSS: Maximum Segment Size, the amount of payload data in a TCP packet)
  – This is only for pedagogical purposes

• In reality this is a LIE: Real implementations maintain CWND in bytes
Two Basic Questions

• How does the sender detect congestion?

• How does the sender adjust its sending rate?
  – To address three issues
    • Finding available bottleneck bandwidth
    • Adjusting to bandwidth variations
    • Sharing bandwidth
(Recall) Detecting Congestion

• Packet delays
  – Tricky: noisy signal (delay often varies considerably)

• Router tell end-hosts they’re congested

• Packet loss
  – Fail-safe signal that TCP already has to detect
  – Complication: non-congestive loss (checksum errors)

• Two indicators of packet loss
  – No ACK after certain time interval: timeout
  – Multiple duplicate ACKs
Not All Losses the Same

• Duplicate ACKs: isolated loss
  – Still getting ACKs

• Timeout: much more serious
  – Not enough packets in progress to trigger duplicate acks, OR
  – Suffered several losses

• We will adjust rate differently for each case
Rate Adjustment

• Basic structure:
  – Upon receipt of ACK (of new data): increase rate
  – Upon detection of loss: decrease rate

• How we increase/decrease the rate depends on the phase of congestion control we’re in:
  – Discovering available bottleneck bandwidth vs.
  – Adjusting to bandwidth variations
Bandwidth Discovery with Slow Start

• Goal: estimate available bandwidth
  – start slow (for safety)
  – but ramp up quickly (for efficiency)

• Consider
  – RTT = 100ms, MSS=1000bytes
  – Window size to fill 1Mbps of BW = 12.5 packets
  – Window size to fill 1Gbps = 12,500 packets
  – Either is possible!
“Slow Start” Phase

- Sender starts at a slow rate but increases **exponentially** until first loss

- Start with a small congestion window
  - Initially, CWND = 1
  - So, initial sending rate is MSS/RTT

- Double the CWND for each RTT with no loss
Slow Start in Action

- For each RTT: double CWND
- Simpler implementation: for each ACK, CWND += 1
Adjusting to Varying Bandwidth

• Slow start gave an estimate of available bandwidth

• Now, want to track variations in this available bandwidth, oscillating around its current value
  – Repeated probing (rate increase) and backoff (rate decrease)

• TCP uses: “Additive Increase Multiplicative Decrease” (AIMD)
  – We’ll see why shortly…
AIMD

• Additive increase
  – Window grows by one MSS for every RTT with no loss
  – For each successful RTT, CWND = CWND + 1
  – Simple implementation:
    • for each ACK, CWND = CWND + 1/CWND

• Multiplicative decrease
  – On loss of packet, divide congestion window in half
  – On loss, CWND = CWND/2
Leads to the TCP “Sawtooth”
Slow-Start vs. AIMD

• When does a sender stop Slow-Start and start Additive Increase?

• Introduce a “slow start threshold” (ssthresh)
  – Initialized to a large value
  – On timeout, ssthresh = CWND/2

• When CWND = ssthresh, sender switches from slow-start to AIMD-style increase
• What does TCP do?
  – ARQ windowing, set-up, tear-down
• Flow Control in TCP
• Congestion Control in TCP
  – AIMD (slow-start, congestion avoidance)
• What does TCP do?
  – ARQ windowing, set-up, tear-down
• Flow Control in TCP
• Congestion Control in TCP
  – AIMD (slow-start, congestion avoidance) and Fast-Recovery
One Final Phase: Fast Recovery

• The problem: congestion avoidance too slow in recovering from an isolated loss
Example (in units of MSS, not bytes)

• Consider a TCP connection with:
  – CWND=10 packets
  – Last ACK was for packet # 101
    • i.e., receiver expecting next packet to have seq. no. 101

• 10 packets [101, 102, 103,..., 110] are in flight
  – Packet 101 is dropped
  – What ACKs do they generate?
  – And how does the sender respond?
The problem – A timeline

- ACK 101 (due to 102) cwnd=10 dupACK#1 (no xmit)
- ACK 101 (due to 103) cwnd=10 dupACK#2 (no xmit)
- ACK 101 (due to 104) cwnd=10 dupACK#3 (no xmit)
- RETRANSMIT 101 ssthresh=5 cwnd= 5
- ACK 101 (due to 105) cwnd=5 + 1/5 (no xmit)
- ACK 101 (due to 106) cwnd=5 + 2/5 (no xmit)
- ACK 101 (due to 107) cwnd=5 + 3/5 (no xmit)
- ACK 101 (due to 108) cwnd=5 + 4/5 (no xmit)
- ACK 101 (due to 109) cwnd=5 + 5/5 (no xmit)
- ACK 101 (due to 110) cwnd=6 + 1/5 (no xmit)
- ACK 111 (due to 101) only now can we transmit new packets
- Plus no packets in flight so ACK “clocking” (to increase CWND) stalls for another RTT
Solution: Fast Recovery

Idea: Grant the sender temporary “credit” for each dupACK so as to keep packets in flight

• If dupACKcount = 3
  – ssthresh = cwnd/2
  – cwnd = ssthresh + 3

• While in fast recovery
  – cwnd = cwnd + 1 for each additional duplicate ACK

• Exit fast recovery after receiving new ACK
  – set cwnd = ssthresh
Example

• Consider a TCP connection with:
  – CWND=10 packets
  – Last ACK was for packet # 101
    • i.e., receiver expecting next packet to have seq. no. 101

• 10 packets [101, 102, 103,..., 110] are in flight
  – Packet 101 is dropped
Timeline

- ACK 101 (due to 102) cwnd=10 dup#1
- ACK 101 (due to 103) cwnd=10 dup#2
- ACK 101 (due to 104) cwnd=10 dup#3
- REXMIT 101 ssthresh=5 cwnd= 8 (5+3)
- ACK 101 (due to 105) cwnd= 9 (no xmit)
- ACK 101 (due to 106) cwnd=10 (no xmit)
- ACK 101 (due to 107) cwnd=11 (xmit 111)
- ACK 101 (due to 108) cwnd=12 (xmit 112)
- ACK 101 (due to 109) cwnd=13 (xmit 113)
- ACK 101 (due to 110) cwnd=14 (xmit 114)
- ACK 111 (due to 101) cwnd = 5 (xmit 115) ➞ exiting fast recovery
- Packets 111-114 already in flight
- ACK 112 (due to 111) cwnd = 5 + 1/5 ➞ back in congestion avoidance
Summary: TCP congestion control

- **Slow Start**
  - $cwnd = 1 \times MSS$
  - $ssthresh = 64$ KB
  - $dupACKcount = 0$
  - Retransmit missing segment
  - $cwnd > ssthresh$

- **Fast Recovery**
  - $dupACKcount == 3$
  - $ssthresh = cwnd/2$
  - $cwnd = ssthresh + 3$
  - Retransmit missing segment

- **Congestion Avoidance**
  - $new ACK$
  - $cwnd = cwnd + MSS$
  - $dupACKcount = 0$
  - Transmit new segment(s), as allowed
  - $cwnd > ssthresh$

- **Fast Recovery**
  - $dupACKcount == 3$
  - $ssthresh = cwnd/2$
  - $cwnd = ssthresh + 3$
  - Retransmit missing segment
<table>
<thead>
<tr>
<th>Variant</th>
<th>Feedback</th>
<th>Required changes</th>
<th>Benefits</th>
<th>Fairness</th>
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</thead>
<tbody>
<tr>
<td>(New) Reno</td>
<td>Loss</td>
<td>—</td>
<td>—</td>
<td>Delay</td>
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<tr>
<td>Vegas</td>
<td>Delay</td>
<td>Sender</td>
<td>Less loss</td>
<td>Proportional</td>
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<td>High Speed</td>
<td>Loss</td>
<td>Sender</td>
<td>High bandwidth</td>
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<td>BIC</td>
<td>Loss</td>
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<td>CUBIC</td>
<td>Loss</td>
<td>Sender</td>
<td>High bandwidth</td>
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<tr>
<td>C2TCP</td>
<td>Loss/Delay</td>
<td>Sender</td>
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<tr>
<td>NATCP</td>
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<td>Elastic-TCP</td>
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<td>Compound TCP</td>
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<td>Westwood</td>
<td>Loss/Delay</td>
<td>Sender</td>
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<td>Jersey</td>
<td>Loss/Delay</td>
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<tr>
<td>BBR</td>
<td>Delay</td>
<td>Sender</td>
<td>BLVC, Bufferbloat</td>
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<tr>
<td>CLAMP</td>
<td>Multi-bit signal</td>
<td>Receiver, Router</td>
<td>V</td>
<td>Max-min</td>
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<tr>
<td>TFRC</td>
<td>Loss</td>
<td>Sender, Receiver</td>
<td>No Retransmission</td>
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<td>RED</td>
<td>Loss</td>
<td>Router</td>
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TCP Throughput Equation
A Simple Model for TCP Throughput

\[
cwnd
\]

\[
\frac{W_{\max}}{2}
\]

\[
W_{\max}
\]

Loss

\[
\frac{1}{2} W_{\max} \text{ RTTs between drops}
\]

Avg. \( \frac{3}{4} W_{\max} \) packets per RTTs

\[
RTT
\]
A Simple Model for TCP Throughput

Packet drop rate, \( p = \frac{1}{A} \), where \( A = \frac{3}{8} W_{\text{max}}^2 \)

Throughput, \( B = \frac{A}{\left(\frac{W_{\text{max}}}{2}\right)^{RTT}} = \sqrt{\frac{3}{2} \frac{1}{RTT \sqrt{p}}} \)
Implications (1): Different RTTs

Throughput = $\sqrt{\frac{3}{2}} \frac{1}{RTT \sqrt{p}}$

- Flows get throughput inversely proportional to RTT
- TCP unfair in the face of heterogeneous RTTs!
Implications (2): High Speed TCP

\[
\text{Throughput} = \sqrt{\frac{3}{2}} \frac{1}{RTT \sqrt{p}}
\]

- Assume RTT = 100ms, MSS=1500bytes
- What value of \( p \) is required to reach 100Gbps throughput
  - \( \sim 2 \times 10^{-12} \)
- How long between drops?
  - \( \sim 16.6 \) hours
- How much data has been sent in this time?
  - \( \sim 6 \) petabits
- These are not practical numbers!
Adapting TCP to High Speed

- Once past a threshold speed, increase CWND faster
  - A proposed standard [Floyd’03]: once speed is past some threshold, change equation to \( p^{-0.8} \) rather than \( p^{-0.5} \)
  - Let the additive constant in AIMD depend on CWND

- Other approaches?
  - Multiple simultaneous connections (\textit{hacky} but works today)
  - Router-assisted approaches (will see shortly)
Implications (3): *Rate*-based CC

- TCP throughput is “choppy”
  - repeated swings between $W/2$ to $W$

- Some apps would prefer sending at a steady rate
  - e.g., streaming apps

- A solution: “Equation-Based Congestion Control”
  - ditch TCP’s increase/decrease rules and just follow the equation
  - measure drop percentage $p$, and set rate accordingly

- Following the TCP equation ensures we’re “TCP friendly”
  - i.e., use no more than TCP does in similar setting
TCP CUBIC

- Is there a better way than AIMD to “probe” for usable bandwidth?

- Insight/intuition:
  - $W_{\text{max}}$: sending rate at which congestion loss was detected
  - congestion state of bottleneck link probably (?) hasn’t changed much
  - after cutting rate/window in half on loss, initially ramp to to $W_{\text{max}}$ faster, but then approach $W_{\text{max}}$ more slowly
TCP CUBIC

- K: point in time when TCP window size will reach $W_{\text{max}}$
  - K itself is tuneable
- increase W as a function of the cube of the distance between current time and K
  - larger increases when further away from K
  - smaller increases (cautious) when nearer K
- TCP CUBIC default in Linux, most popular TCP for popular Web servers
TCP and the congested “bottleneck link”

- TCP (classic, CUBIC) increase TCP’s sending rate until packet loss occurs at some router’s output: the *bottleneck link*
TCP and the congested “bottleneck link”

- TCP (classic, CUBIC) increase TCP’s sending rate until packet loss occurs at some router’s output: the *bottleneck link*
- understanding congestion: useful to focus on congested bottleneck link

**insight:** increasing TCP sending rate will *not* increase end-end throughout with congested bottleneck

**Goal:** “keep the end-end pipe just full, but not fuller”
Delay-based TCP Congestion Control

Keeping sender-to-receiver pipe “just full enough, but no fuller”: keep bottleneck link busy transmitting, but avoid high delays/buffering

Delay-based approach:

- $\text{RTT}_{\text{min}}$ - minimum observed RTT (uncongested path)
- uncongested throughput with congestion window $cwnd$ is $cwnd/\text{RTT}_{\text{min}}$

If measured throughput “very close” to uncongested throughput:
- increase $cwnd$ linearly /* since path not congested */

Else if measured throughput “far below” uncongested throughput:
- decrease $cwnd$ linearly /* since path is congested */
Delay-based TCP Congestion Control

- congestion control without inducing/forcing loss
- maximizing throughout (“keeping the just pipe full...”) while keeping delay low (“...but not fuller”)
- a number of deployed TCPs take a delay-based approach
  - BBR deployed on Google’s (internal) backbone network
Recap: TCP problems

- Misled by non-congestion losses
- Fills up queues leading to high delays
- Short flows complete before discovering available capacity
- AIMD impractical for high speed links
- Sawtooth discovery too choppy for some apps
- Unfair under heterogeneous RTTs
- Tight coupling with reliability mechanisms
- Endhosts can cheat

Could fix many of these with some help from routers!
Router-Assisted Congestion Control

• Three tasks for CC:
  – Isolation/fairness
  – Adjustment*
  – Detecting congestion

* This may be *automatic* eg loss-response of TCP
How can routers ensure each flow gets its “fair share”?
Fairness: General Approach

• Routers classify packets into “flows”
  – (For now) flows are packets between same source/destination

• Each flow has its own FIFO queue in router

• Router services flows in a fair fashion
  – When line becomes free, take packet from next flow in a fair order

• What does “fair” mean exactly?
Max-Min Fairness

• Given set of bandwidth demands $r_i$ and total bandwidth $C$, max-min bandwidth allocations are:

$$a_i = \min(f, r_i)$$

where $f$ is the unique value such that $\text{Sum}(a_i) = C$
Example

- $C = 10; \quad r_1 = 8, r_2 = 6, r_3 = 2; \quad N = 3$
- $C/3 = 3.33 \rightarrow$
  - Can service all of $r_3$
  - Remove $r_3$ from the accounting: $C = C - r_3 = 8; N = 2$
- $C/2 = 4 \rightarrow$
  - Can’t service all of $r_1$ or $r_2$
  - So hold them to the remaining fair share: $f = 4$

$\min(8, 4) = 4$
$\min(6, 4) = 4$
$\min(2, 4) = 2$
Max-Min Fairness

• Given set of bandwidth demands $r_i$ and total bandwidth $C$, max-min bandwidth allocations are:
  
  \[ a_i = \min(f, r_i) \]

• where $f$ is the unique value such that \( \sum(a_i) = C \)

• Property:
  – If you don’t get full demand, no one gets more than you

• This is what round-robin service gives if all packets are the same size
How do we deal with packets of different sizes?

- Mental model: Bit-by-bit round robin ("fluid flow")

- Can you do this in practice?

- No, packets cannot be preempted

- But we can approximate it
  - This is what "fair queuing" routers do
Fair Queuing (FQ)

• For each packet, compute the time at which the last bit of a packet would have left the router if flows are served bit-by-bit
• Then serve packets in the increasing order of their deadlines
Flow 1 (arrival traffic)

Flow 2 (arrival traffic)

Service in fluid flow system

FQ Packet system

Example

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Fair Queuing (FQ)

• Think of it as an implementation of round-robin generalized to the case where not all packets are equal sized

• **Weighted** fair queuing (WFQ): assign different flows different shares

• Today, some form of WFQ implemented in almost all routers
  – Not the case in the 1980-90s, when CC was being developed
  – Mostly used to isolate traffic at larger granularities (e.g., per-prefix)
FQ vs. FIFO

- **FQ advantages:**
  - Isolation: cheating flows don’t benefit
  - Bandwidth share does not depend on RTT
  - Flows can pick any rate adjustment scheme they want

- **Disadvantages:**
  - More complex than FIFO: per flow queue/state, additional per-packet book-keeping
FQ in the big picture

- FQ does not eliminate congestion ➔ it just manages the congestion

Blue and Green get 0.5Gbps; any excess will be dropped

Will drop an additional 400Mbps from the green flow

If the green flow doesn’t drop its sending rate to 100Mbps, we’re wasting 400Mbps that could be usefully given to the blue flow
FQ in the big picture

• FQ does not eliminate congestion → it just manages the congestion
  – robust to cheating, variations in RTT, details of delay, reordering, retransmission, etc.

• But congestion (and packet drops) still occurs

• And we still want end-hosts to discover/adapt to their fair share!

• What would the end-to-end argument say w.r.t. congestion control?
Fairness is a controversial goal

• What if you have 8 flows, and I have 4?
  – Why should you get twice the bandwidth

• What if your flow goes over 4 congested hops, and mine only goes over 1?
  – Why shouldn’t you be penalized for using more scarce bandwidth?

• And what is a flow anyway?
  – TCP connection
  – Source-Destination pair?
  – Source?
Explicit Congestion Notification (ECN)

• Single bit in packet header; set by congested routers
  – If data packet has bit set, then ACK has ECN bit set
• Many options for when routers set the bit
  – tradeoff between (link) utilization and (packet) delay
• Congestion semantics can be exactly like that of drop
  – I.e., endhost reacts as though it saw a drop

• Advantages:
  – Don’t confuse corruption with congestion; recovery w/ rate adjustment
  – Can serve as an early indicator of congestion to avoid delays
  – Easy (easier) to incrementally deploy
    • defined as extension to TCP/IP in RFC 3168 (uses diffserv bits in the IP header)
Explicit congestion notification (ECN)

TCP deployments often implement *network-assisted* congestion control:
- two bits in IP header (ToS field) marked *by network router* to indicate congestion
  - *policy* to determine marking chosen by network operator
- congestion indication carried to destination
- destination sets ECE bit on ACK segment to notify sender of congestion
- involves both IP (IP header ECN bit marking) and TCP (TCP header C,E bit marking)
Transport Recap

A “big bag”:
  Multiplexing, reliability, error-detection, error-recovery, flow and congestion control, ....

• **UDP:**
  – Minimalist - multiplexing and error detection

• **TCP:**
  – somewhat hacky
  – but practical/deployable
  – good enough to have raised the bar for the deployment of new approaches
  – though the needs of datacenters change the status quos

• **Beyond TCP (discussed in Topic 6):**
  – QUIC / application-aware transport layers