CMSC 417

Computer Networks

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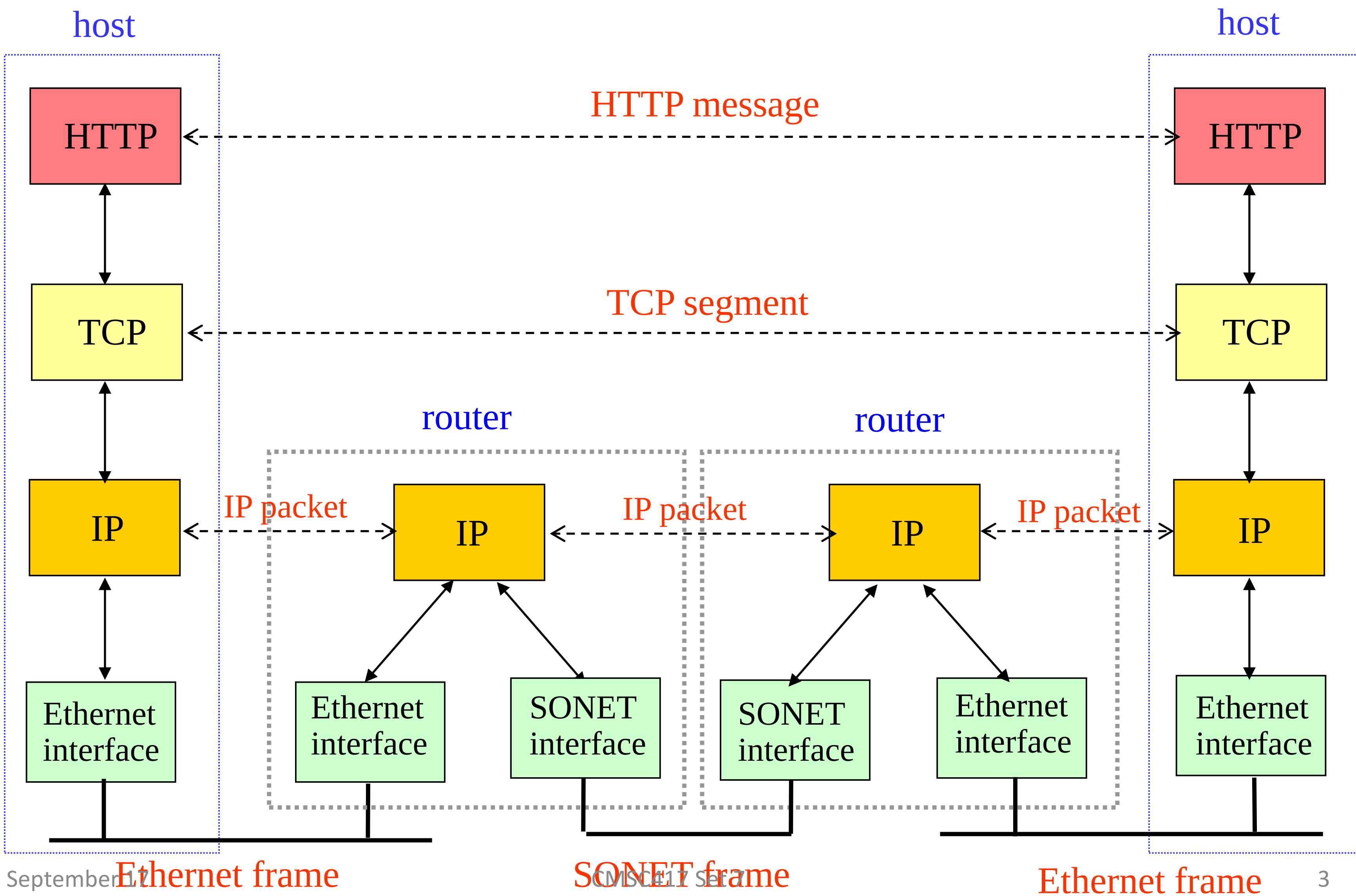
Set 7

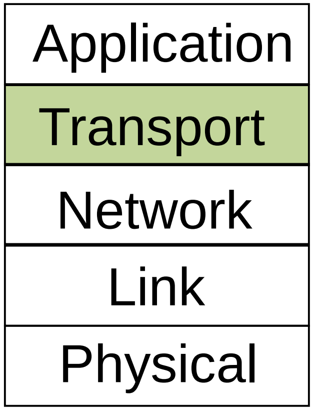
The Transport Layer

Message, Segment, Packet, and Frame



The Transport Layer

Responsible for delivering data across networks with the desired reliability or quality



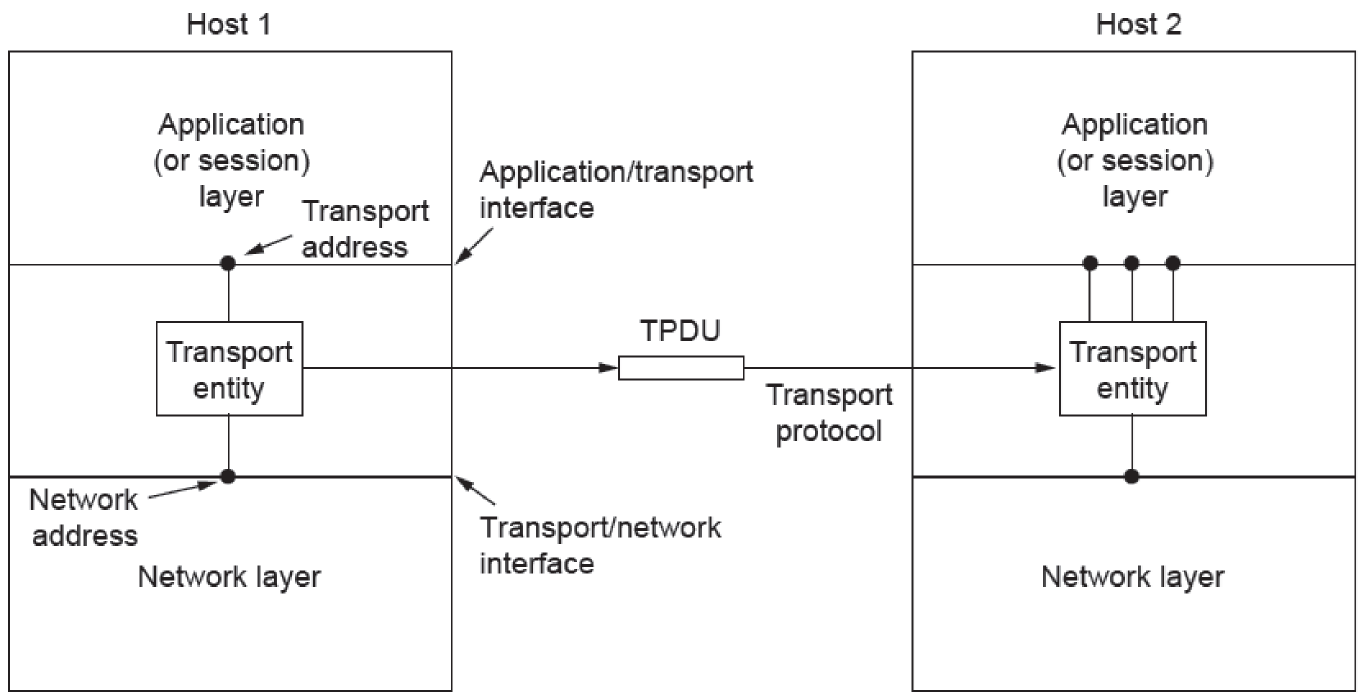
The Transport Service

- Services Provided to the Upper Layers
- Transport Service Primitives
- Berkeley Sockets
- An Example of Socket Programming:
  - An Internet File Server

Services Provided to the Upper Layers (1)

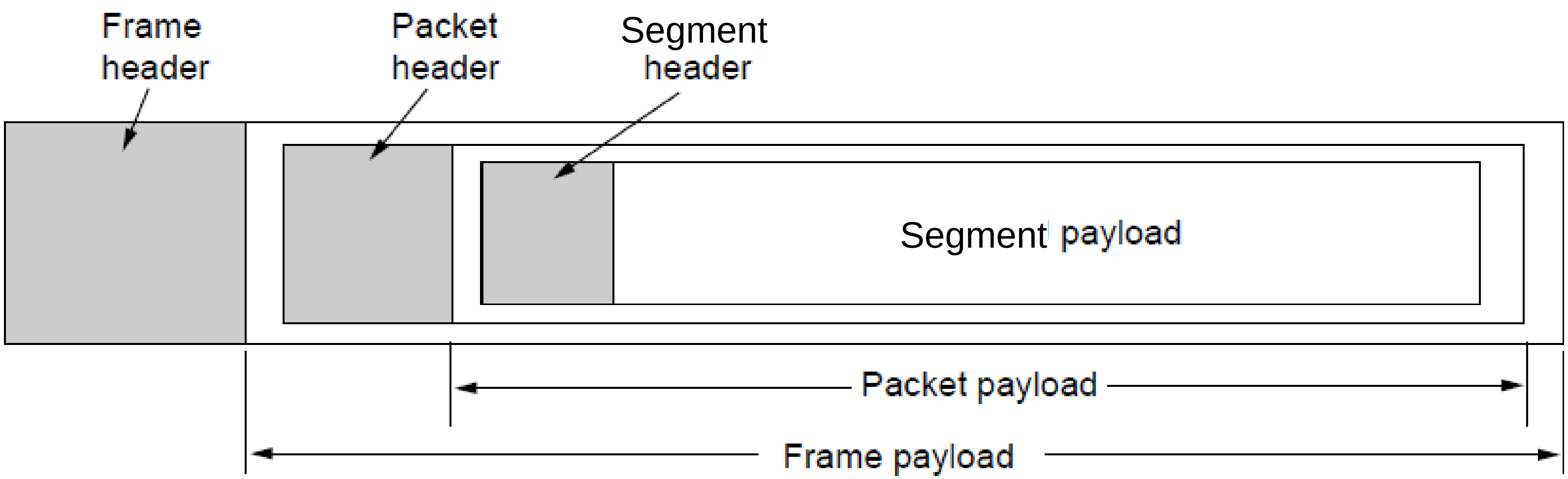
Transport layer adds reliability to the network layer

– Offers connectionless (e.g., UDP) and connection-oriented (e.g, TCP) service to applications



Services Provided to the Upper Layers (2)

Transport layer sends <u>segments</u> in packets (in frames)



Berkeley Sockets

Very widely used primitives started with TCP on UNIX

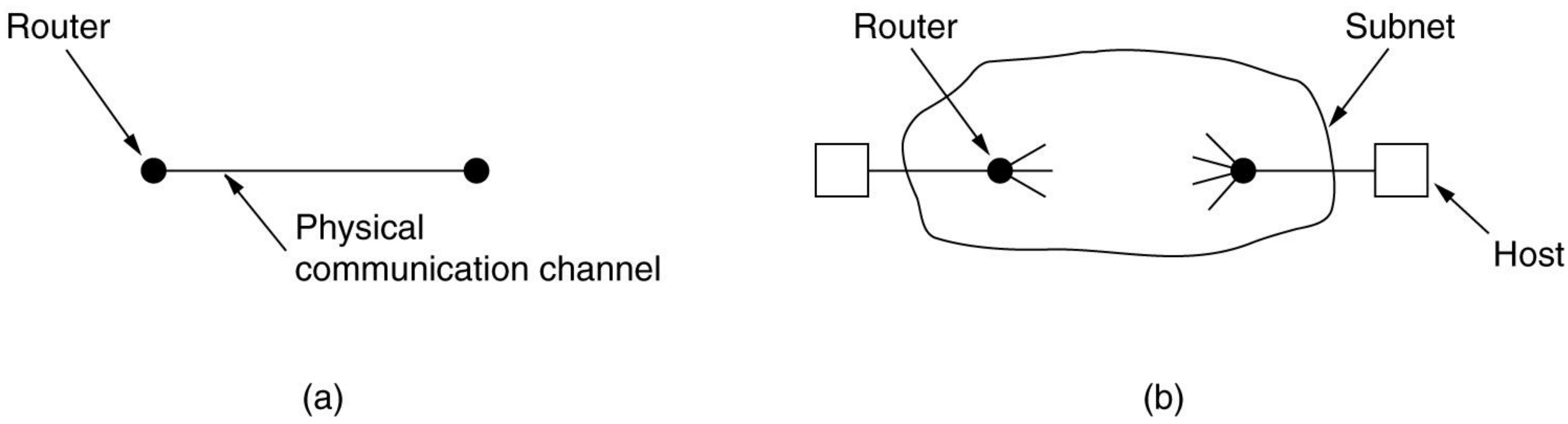
- Notion of “sockets” as transport endpoints
- Like simple set plus SOCKET, BIND, and ACCEPT

Primitive	Meaning
SOCKET	Create a new communication end point
BIND	Associate a local address with a socket
LISTEN	Announce willingness to accept connections; give queue size
ACCEPT	Passively establish an incoming connection
CONNECT	Actively attempt to establish a connection
SEND	Send some data over the connection
RECEIVE	Receive some data from the connection
CLOSE	Release the connection

Elements of Transport Protocols

- Addressing »
- Connection establishment »
- Connection release »
- Error control and flow control »
- Multiplexing »
- Crash recovery »

Transport Protocol



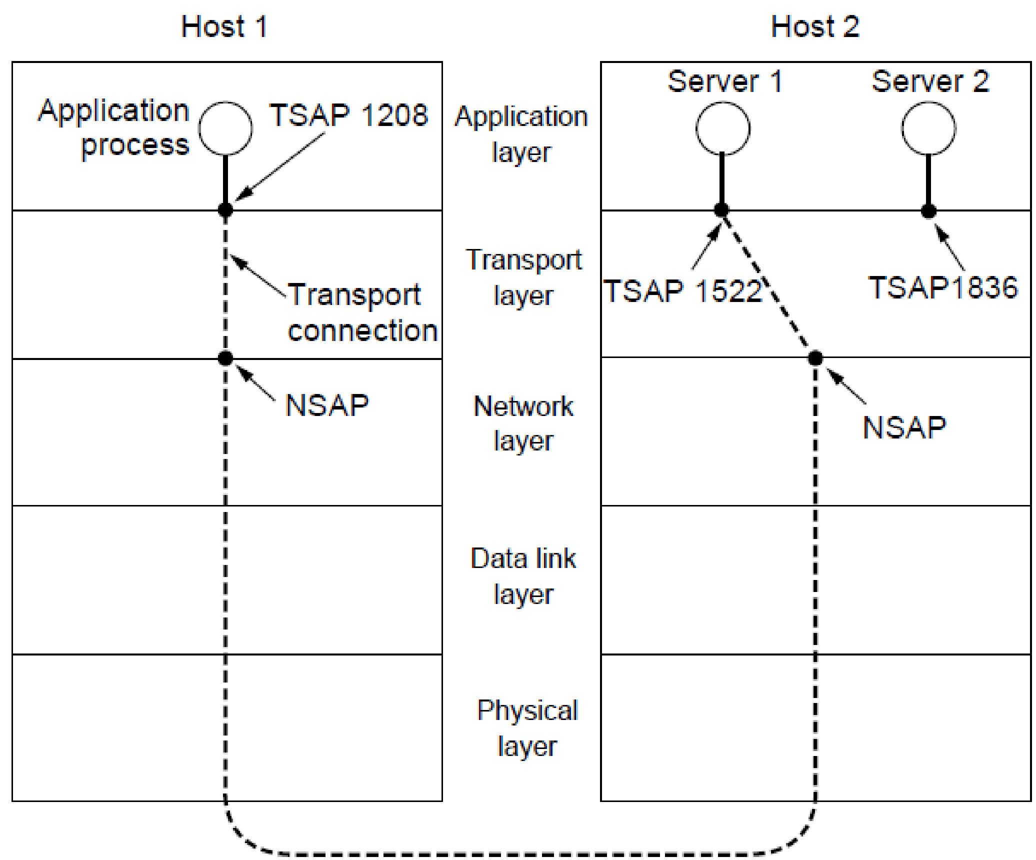
(a) Environment of the data link layer.

(b) Environment of the transport layer.

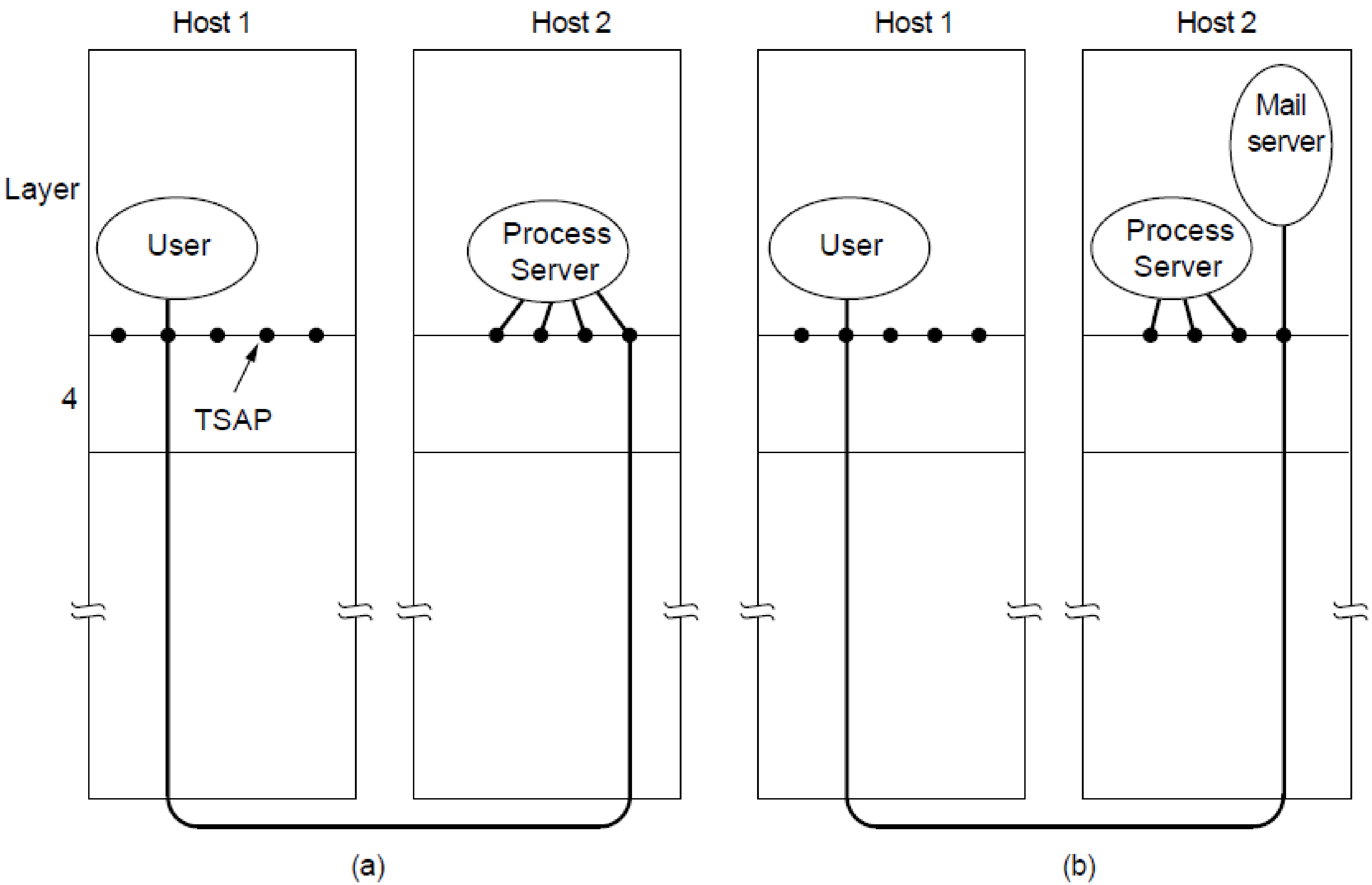
Packets may be lost, delayed, duplicated, out of order, etc.

Addressing

- Transport layer adds TSAPs
- Multiple clients and servers can run on a host with a single network (IP) address
- TSAPs are ports for TCP/UDP



Addressing (2)



How a user process in host 1 establishes a connection with a mail server in host 2 via a process server.

Connection Establishment

- Connect
  - Expect confirmation that one and only one connection has been established
  - In a reasonable time
- Send Connection Request
- Expect Connection Accepted
- Segments (Packets) may be lost, corrupted, delayed, out of order, duplicated.

Connection Establishment (1)

Key problem is to ensure reliability even though packets may be lost, corrupted, <u>delayed</u>, and <u>duplicated</u>

- Detect corrupted packets (error detection – e.g. checksum)
- Detect lost packets (Time Out)
- Identify retransmitted packets (sequence numbers)

Sequence Numbers

- An integer number with finite number of bits
  - 8 bits => 256
  - 16 bits => 64K
- Roll Over
  - How long does it take?
  - Can there be old duplicates in the system?

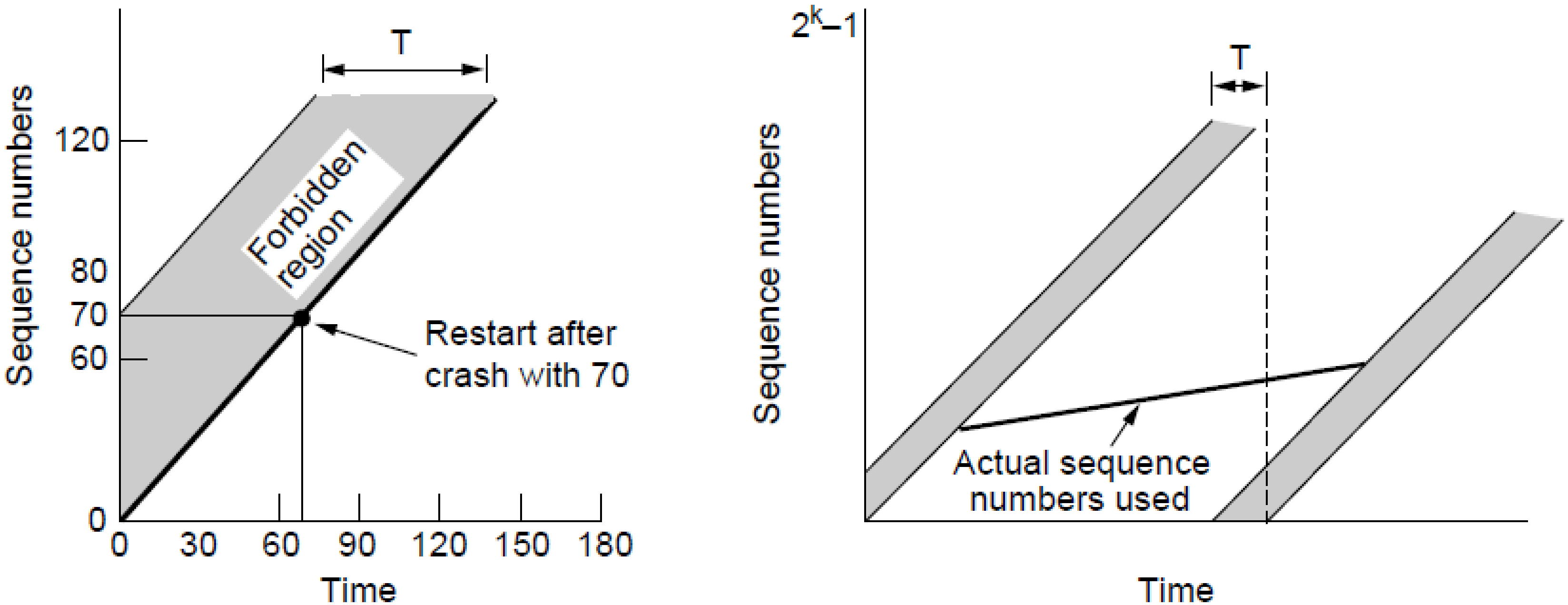
Approach:

- Don't reuse sequence numbers within twice the MSL (Maximum Segment Lifetime) of 2T=240 secs
- Three-way handshake for establishing connection

Connection Establishment (2)

Use a sequence number space large enough that it will not wrap, even when sending at full rate

- Clock (high bits) advances & keeps state over crash



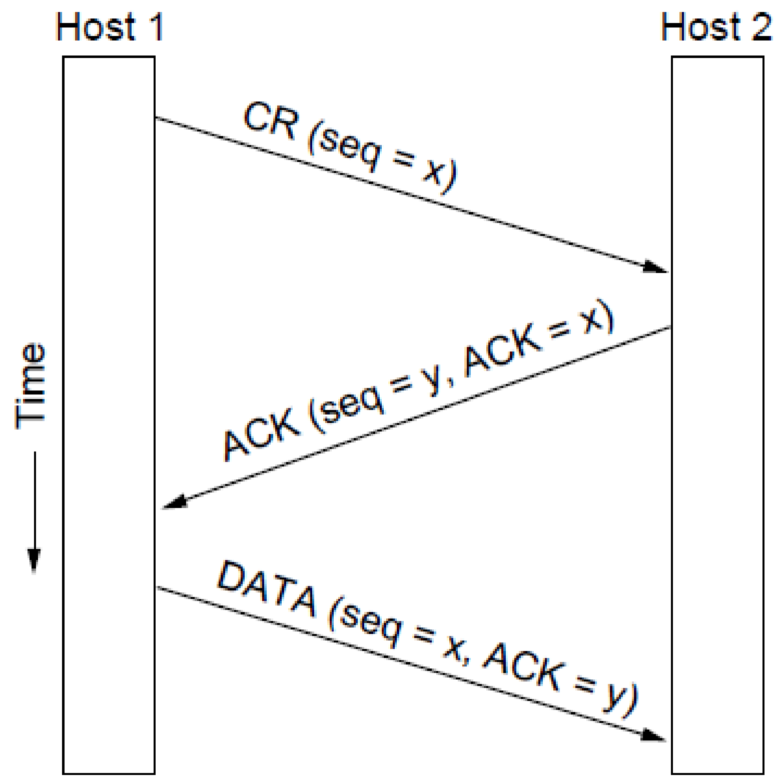
Need seq. number not to wrap within T seconds

Need seq. number not to climb too slowly for too long

Connection Establishment (3)

Three-way handshake used for initial packet

- Since no state from previous connection
- Both hosts contribute fresh seq. numbers
- CR = Connect Request



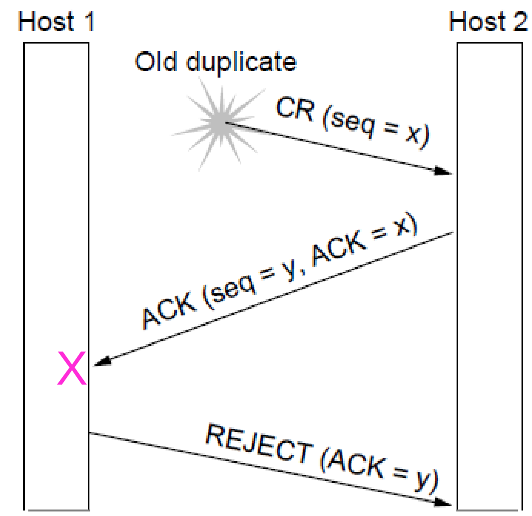
Connection Establishment (4)

Three-way handshake protects against odd cases:

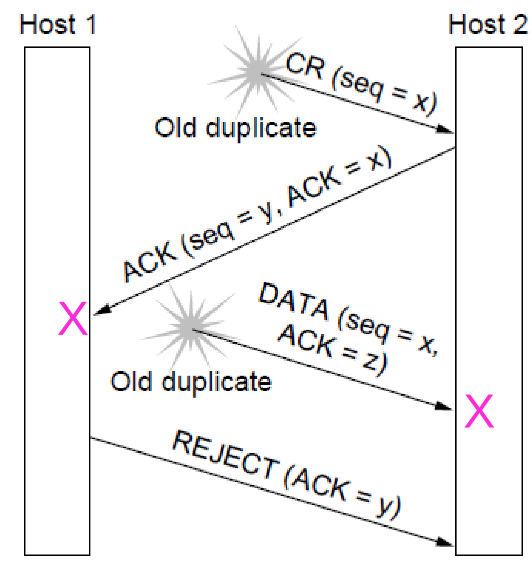
a) Duplicate CR. Spurious ACK does not connect

b) Duplicate CR and DATA. Same plus DATA will be rejected (wrong ACK).

a)



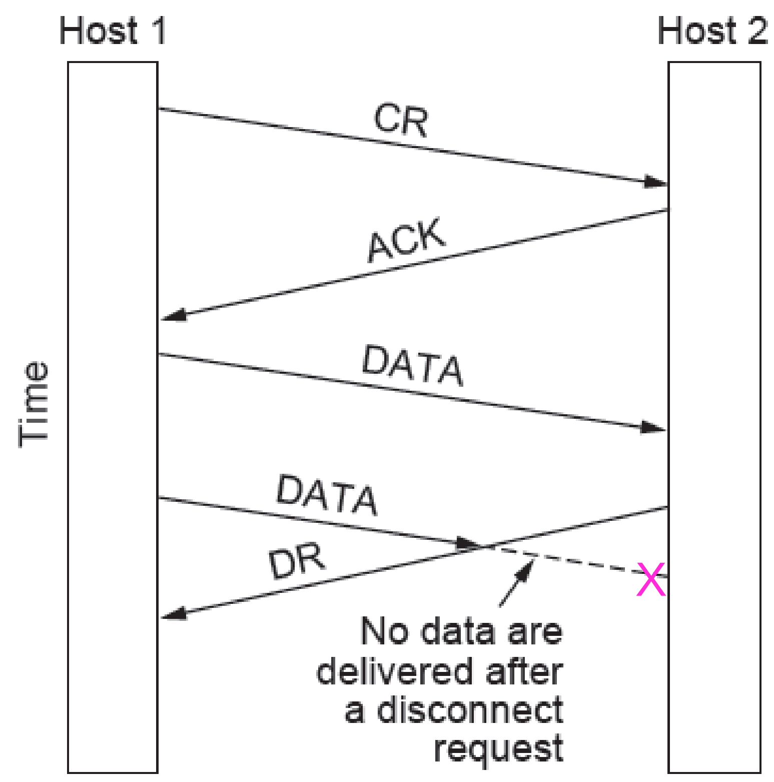
b)



Connection Release (1)

Key problem is to ensure reliability while releasing

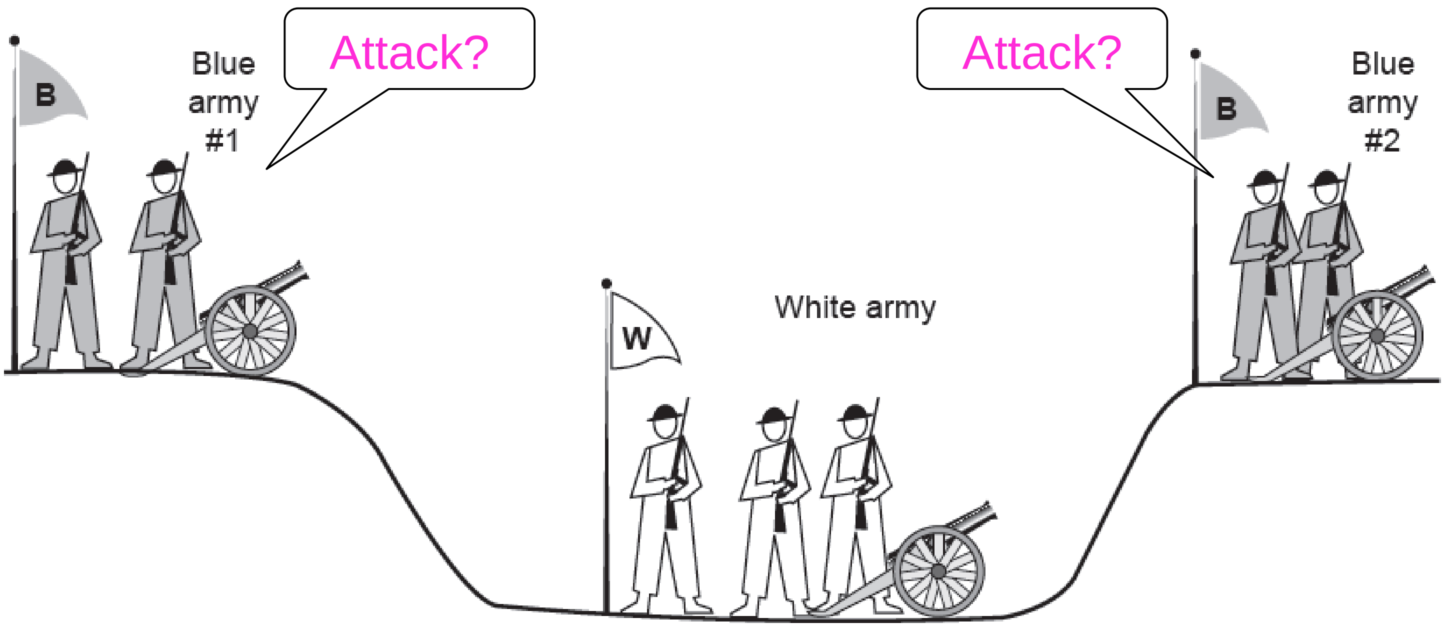
Asymmetric release (when one side breaks connection) is abrupt and may lose data



Connection Release (2)

Symmetric release (both sides agree to release) can't be handled solely by the transport layer

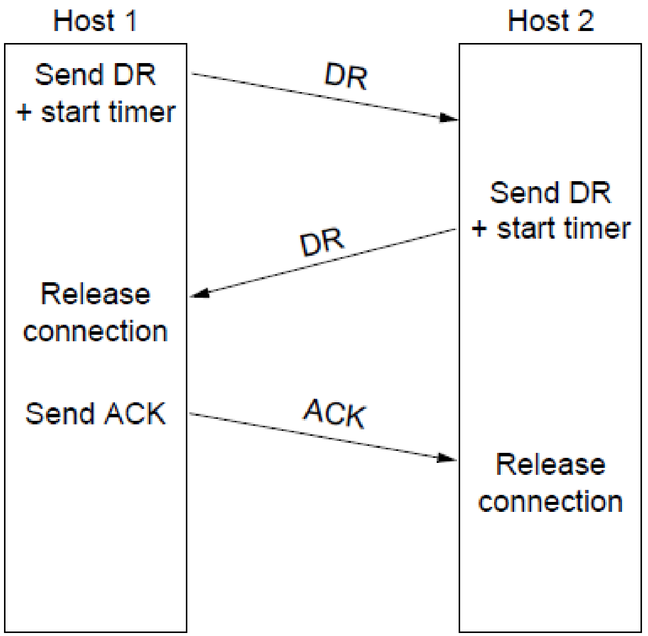
- Two-army problem shows pitfall of agreement



Connection Release (3)

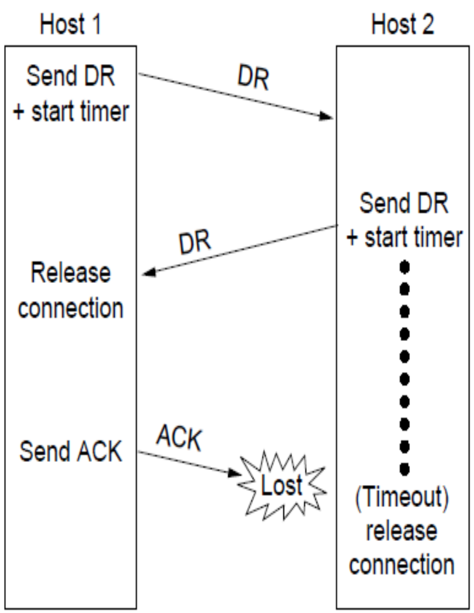
Normal release sequence, initiated by transport user on Host 1

- DR=Disconnect Request
- Both DRs are ACKed by the other side

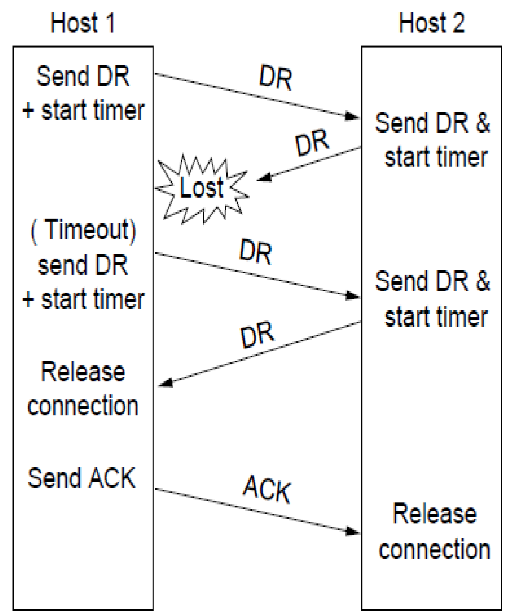


Connection Release (4)

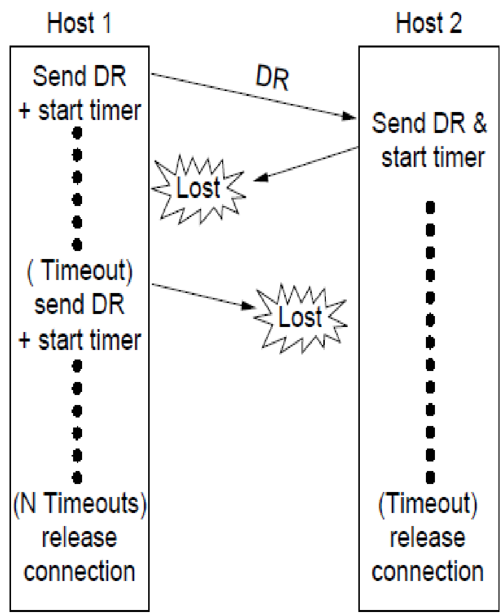
Error cases are handled with timer and retransmission



Final ACK lost,
Host 2 times out



Lost DR causes
retransmissions



Extreme: Many lost
DRs cause both
hosts to timeout

Flow Control

- Use Sliding Window
- Buffering
  - Sender buffers all TPDUs until acknowledged
  - TPDU lost by the network
    - Unreliable service
    - Receiver not having buffer
- How should buffers be managed
  - Dedicate
  - Acquire when needed
- Traffic
  - Low bandwidth, Bursty – buffer at sender – acquire at receiver
  - High bandwidth, smooth – Buffer at both ends
- Exchange Buffer information

Error Control and Flow Control (1)

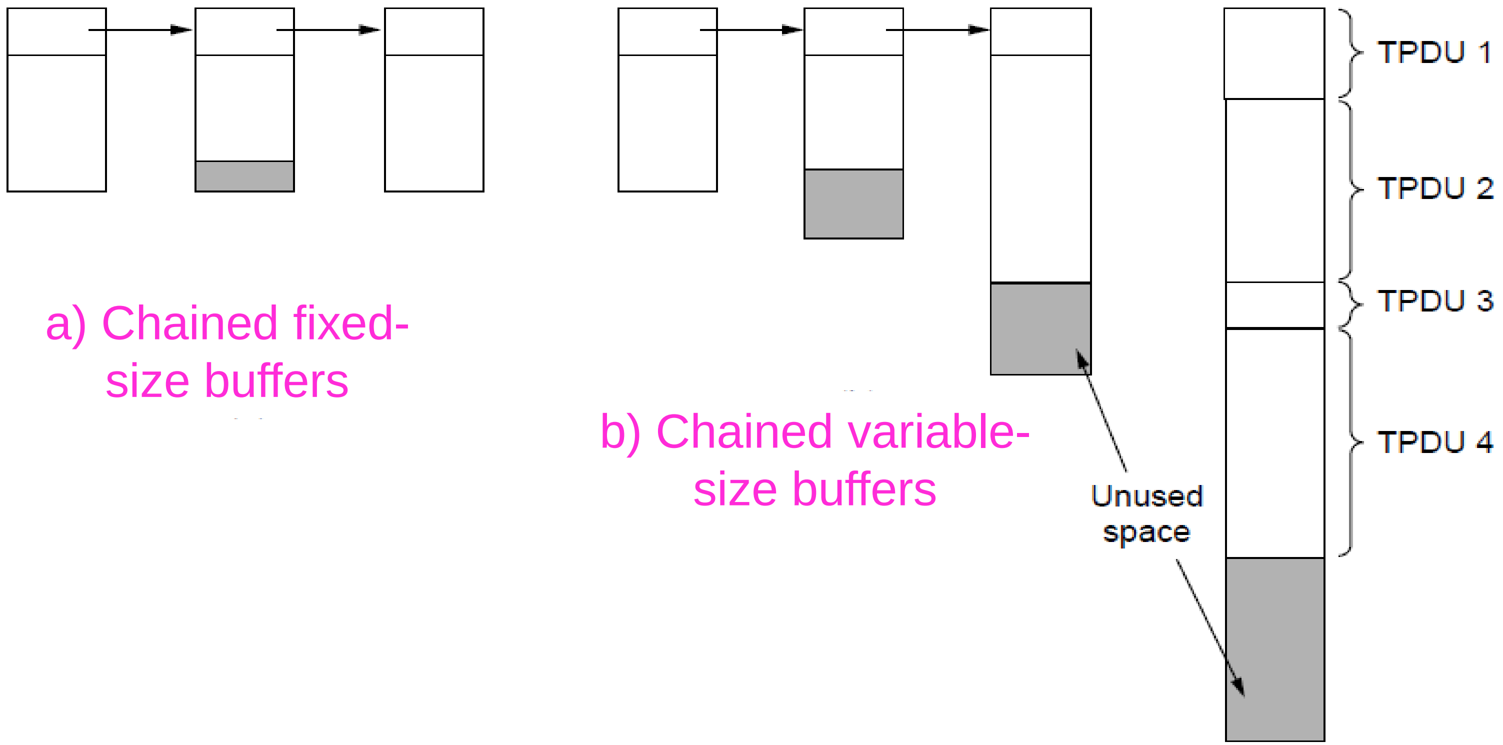
Foundation for error control is a sliding window (from Link layer) with checksums and retransmissions

Flow control manages buffering at sender/receiver

- Issue is that data goes to/from the network and applications at different times
- Window tells sender available buffering at receiver
- Makes a variable-size sliding window

Error Control and Flow Control (2)

Different buffer strategies trade efficiency / complexity



a) Chained fixed-size buffers

b) Chained variable-size buffers

c) One large circular buffer

Error Control and Flow Control (3)

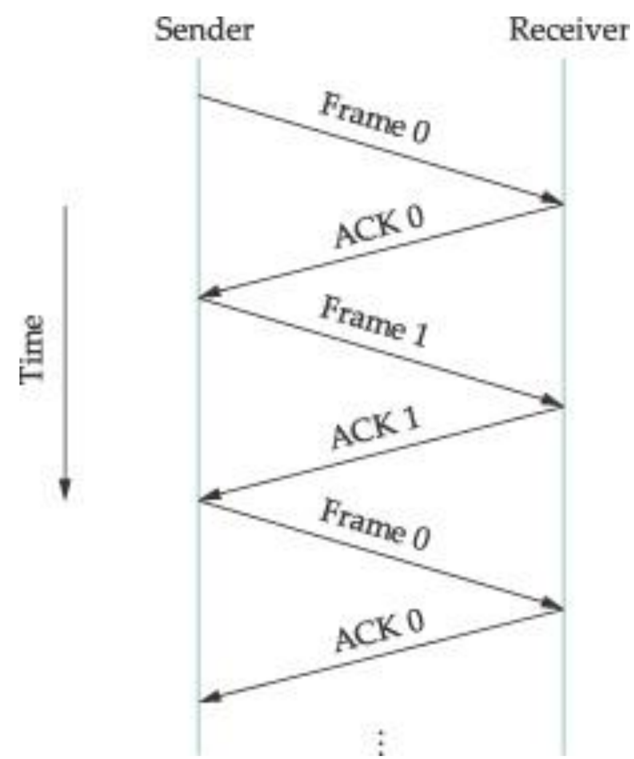
Flow control example: A's data is limited by B's

	A	Message	B	B's Buffer	Comments
1	→	< request 8 buffers>	→		A wants 8 buffers
2	←	<ack = 15, buf = 4>	←	0 1 2 3	B grants messages 0-3 only
3	→	<seq = 0, data = m0>	→	0 1 2 3	A has 3 buffers left now
4	→	<seq = 1, data = m1>	→	0 1 2 3	A has 2 buffers left now
5	→	<seq = 2, data = m2>	• • •	0 1 2 3	Message lost but A thinks it has 1 left
6	←	<ack = 1, buf = 3>	←	1 2 3 4	B acknowledges 0 and 1, permits 2-4
7	→	<seq = 3, data = m3>	→	1 2 3 4	A has 1 buffer left
8	→	<seq = 4, data = m4>	→	1 2 3 4	A has 0 buffers left, and must stop
9	→	<seq = 2, data = m2>	→	1 2 3 4	A times out and retransmits
10	←	<ack = 4, buf = 0>	←	1 2 3 4	Everything acknowledged, but A still blocked
11	←	<ack = 4, buf = 1>	←	2 3 4 5	A may now send 5
12	←	<ack = 4, buf = 2>	←	3 4 5 6	B found a new buffer somewhere
13	→	<seq = 5, data = m5>	→	3 4 5 6	A has 1 buffer left
14	→	<seq = 6, data = m6>	→	3 4 5 6	A is now blocked again
15	←	<ack = 6, buf = 0>	←	3 4 5 6	A is still blocked
16	• • •	<ack = 6, buf = 4>	←	7 8 9 10	Potential deadlock

Reliable Transmission

- Transfer frames without errors
  - Error Correction
  - Error Detection
  - Discard frames with error
- Acknowledgements and Timeouts
- Retransmission
- ARQ – Automatic Repeat Request

Stop and Wait with 1-bit Seq No

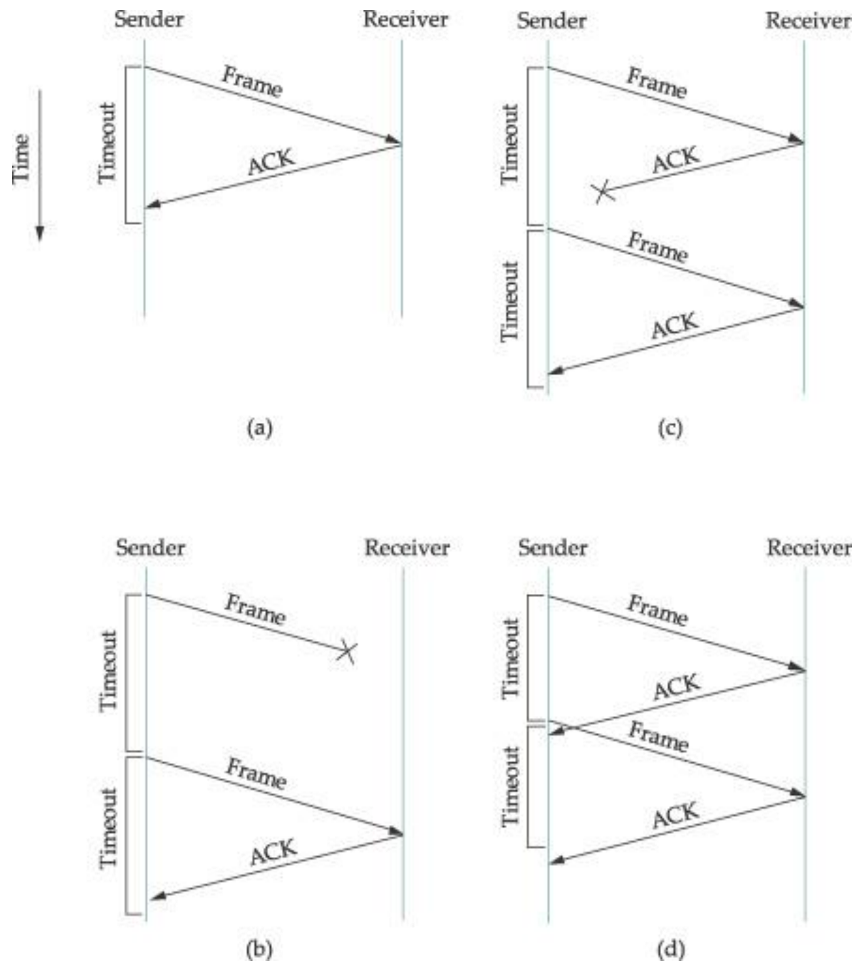


Stop and Wait Protocols

- Simple
- Low Throughput
  - One Frame per RTT

- Increase throughput by having more frames in flight
  - Sliding Window Protocol

Stop and Wait



Duplicate Frames

Stop-and-Wait – Error-free channel

Protocol (p2) ensures sender can't outpace receiver:

- Receiver returns a dummy frame (ack) when ready
- Only one frame out at a time – called stop-and-wait
- We added flow control!

```
void sender2(void)
{
  frame s;
  packet buffer;
  event_type event;

  while (true) {
       from_network_layer(&buffer);
       s.info = buffer;
       to_physical_layer(&s);
       wait_for_event(&event);
  }
}
```

Sender waits to for ack after passing frame to physical layer

```
void receiver2(void)
{
  frame r, s;
  event_type event;
  while (true) {
       wait_for_event(&event);
       from_physical_layer(&r);
       to_network_layer(&r.info);
       to_physical_layer(&s);
  }
}
```

Receiver sends ack after passing frame to network layer

Stop-and-Wait – Noisy channel (1)

ARQ (Automatic Repeat reQuest) adds error control

- Receiver acks frames that are correctly delivered
- Sender sets timer and resends frame if no ack)

For correctness, frames and acks must be numbered

- Else receiver can't tell retransmission (due to lost ack or early timer) from new frame
- For stop-and-wait, 2 numbers (1 bit) are sufficient

Stop-and-Wait – Noisy channel (2)

Sender loop (p3):

Send frame (or retransmission)
Set timer for retransmission
Wait for ack or timeout

If a good ack then set up for the next frame to send (else the old frame will be retransmitted)

```
void sender3(void) {
  seq_nr next_frame_to_send;
  frame s;
  packet buffer;
  event_type event;

  next_frame_to_send = 0;
  from_network_layer(&buffer);
  while (true) {
      s.info = buffer;
      s.seq = next_frame_to_send;
      to_physical_layer(&s);
      start_timer(s.seq);
      wait_for_event(&event);
      if (event == frame_arrival) {
          from_physical_layer(&s);
          if (s.ack == next_frame_to_send) {
              stop_timer(s.ack);
              from_network_layer(&buffer);
              inc(next_frame_to_send);
          }
      }
  }
}
```

Stop-and-Wait – Noisy channel (3)

Receiver loop (p3):

Wait for a frame

If it's new then take it and advance expected frame

Ack current frame

```
void receiver3(void)
{
  seq_nr frame_expected;
  frame r, s;
  event_type event;

  frame_expected = 0;
  while (true) {
       wait_for_event(&event);
       if (event == frame_arrival) {
              from_physical_layer(&r);
              if (r.seq == frame_expected) {
                     to_network_layer(&r.info);
                     inc(frame_expected);
              }
              s.ack = 1 – frame_expected;
              to_physical_layer(&s);
       }
  }
}
```

Sliding Window Protocols

– Sliding Window concept »
– One-bit Sliding Window »
– Go-Back-N »
– Selective Repeat »

Sliding Window concept (1)

Sender maintains window of frames it can send

- Needs to buffer them for possible retransmission
- Window advances with next acknowledgements

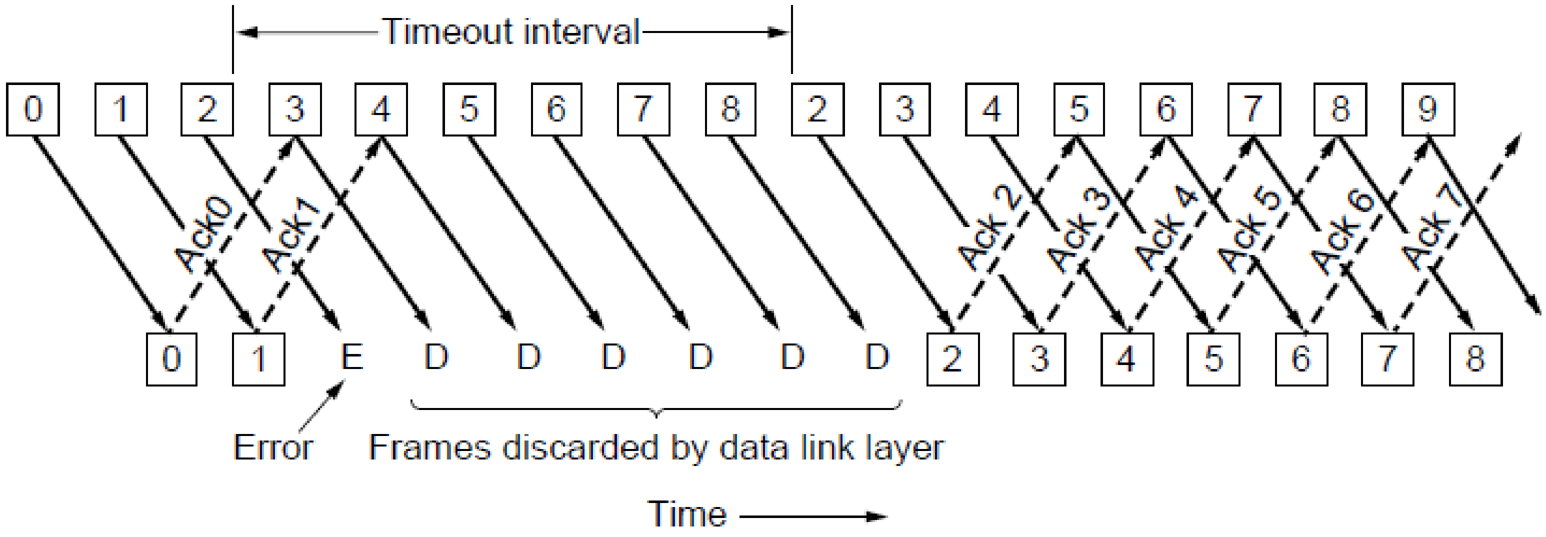
Receiver maintains window of frames it can receive

- Needs to keep buffer space for arrivals
- Window advances with in-order arrivals

Go-Back-N (1)

Receiver only accepts/acks frames that arrive in order:

- Discards frames that follow a missing/errored frame
- Sender times out and resends all outstanding frames



Go-Back-N (2)

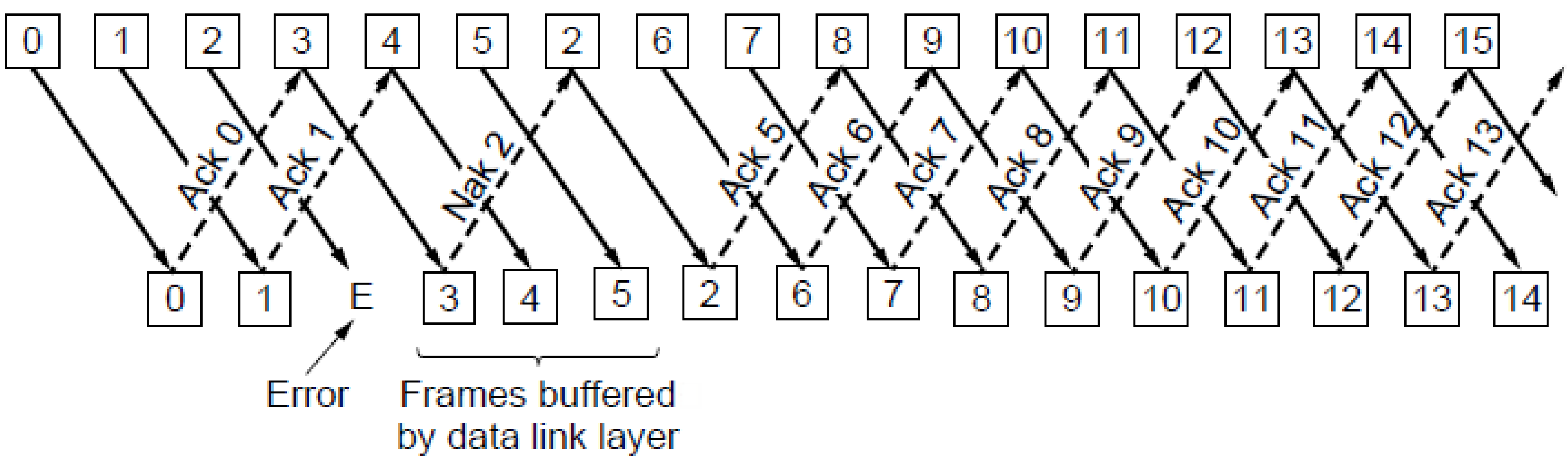
Tradeoff made for Go-Back-N:

- Simple strategy for receiver; needs only 1 frame
- Wastes link bandwidth for errors with large windows; entire window is retransmitted

Selective Repeat (1)

Receiver accepts frames anywhere in receive window

- Cumulative ack indicates highest in-order frame
- NAK (negative ack) causes sender retransmission of a



Selective Repeat (2)

Tradeoff made for Selective Repeat:

- More complex than Go-Back-N due to buffering at receiver and multiple timers at sender
- More efficient use of link bandwidth as only lost frames are resent (with low error rates)

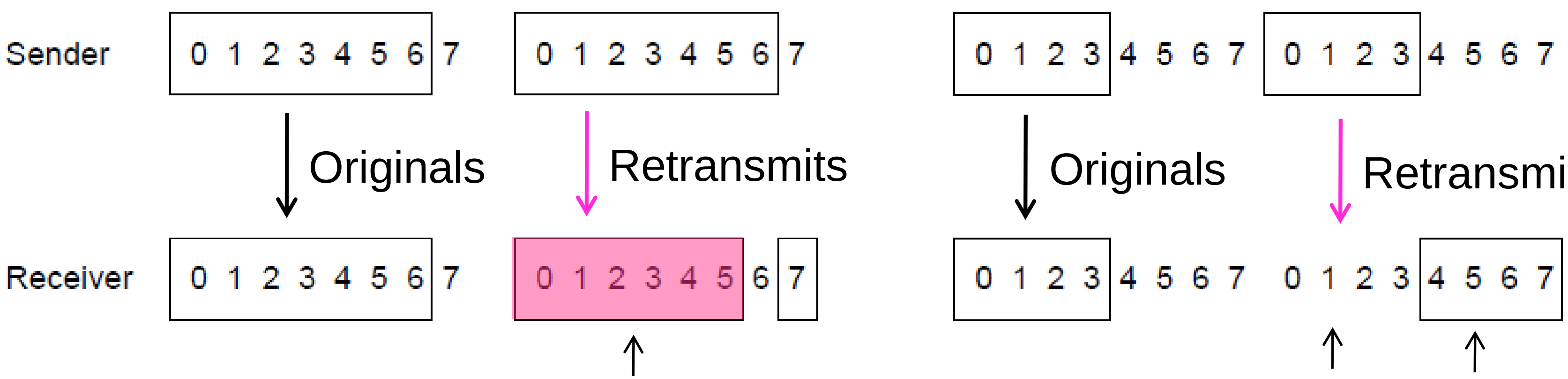
Selective Repeat (3)

For correctness, we require:

- Sequence numbers (s) at least twice the window (w)

Error case (s=8, w=7) – too few sequence numbers

Correct (s=8, w=4) – enough sequence numbers



New receive window overlaps old – retransmits ambiguous

New and old receive window don’t overlap – no ambiguity

Sliding Window Protocols

http://www.ccs-labs.org/teaching/rn/animations/gbn_sr/

http://www.cs.stir.ac.uk/~kjt/software/comms/jasper/SWP3.html

http://www.cs.stir.ac.uk/~kjt/software/comms/jasper/SWP5.html

http://www2.rad.com/networks/2004/sliding_window/

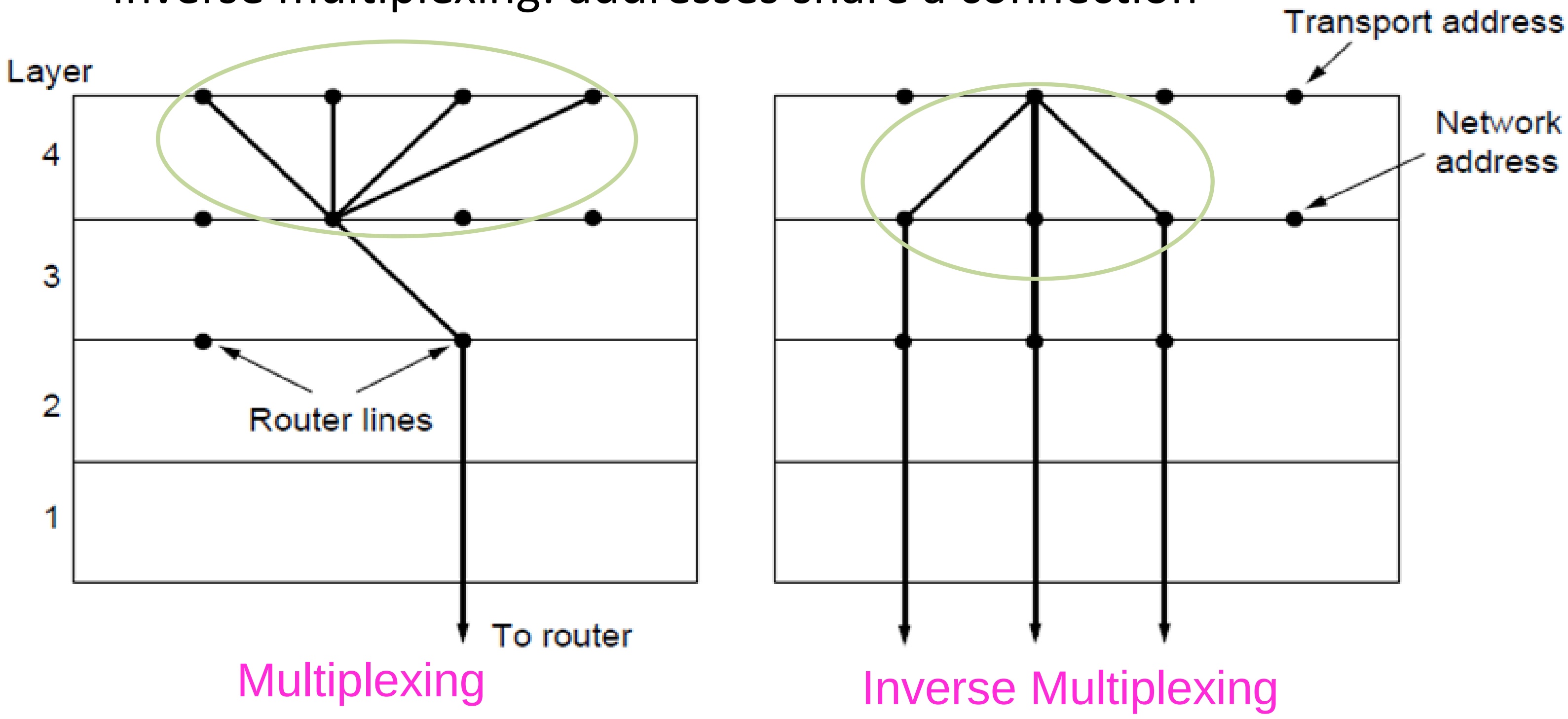
Throughput limits

- Buffers
- Bandwidth – subnet's carrying capacity
  - K TPDUs per second
  - X paths then total of XK

- Flow control to manage
  - Manage window size
    - If network can handle c TPDUs/sec and Cycle time is r then the window size should be cr

Multiplexing

- Kinds of transport / network sharing that can occur:
  - Multiplexing: connections share a network address
  - Inverse multiplexing: addresses share a connection



Crash Recovery

- Network Failures
  - Transport layer handles
    - Connectionless
    - Connection oriented
- Host Crashes
  - Server crash and may reboot
    - Send broadcast asking clients to inform of prior connections (stop and wait protocol)
      - Client – one TPDU outstanding or none outstanding

Crash Recovery

- Application needs to help recovering from a crash
  - Transport can fail since A(ck) / W(rite) not atomic

Strategy used by receiving host

Strategy used by sending host	First ACK, then write			First write, then ACK		
	AC(W)	AWC	C(AW)	C(WA)	W AC	WC(A)
Always retransmit	OK	DUP	OK	OK	DUP	DUP
Never retransmit	LOST	OK	LOST	LOST	OK	OK
Retransmit in S0	OK	DUP	LOST	LOST	DUP	OK
Retransmit in S1	LOST	OK	OK	OK	OK	DUP

OK = Protocol functions correctly
DUP = Protocol generates a duplicate message
LOST = Protocol loses a message

Congestion Control

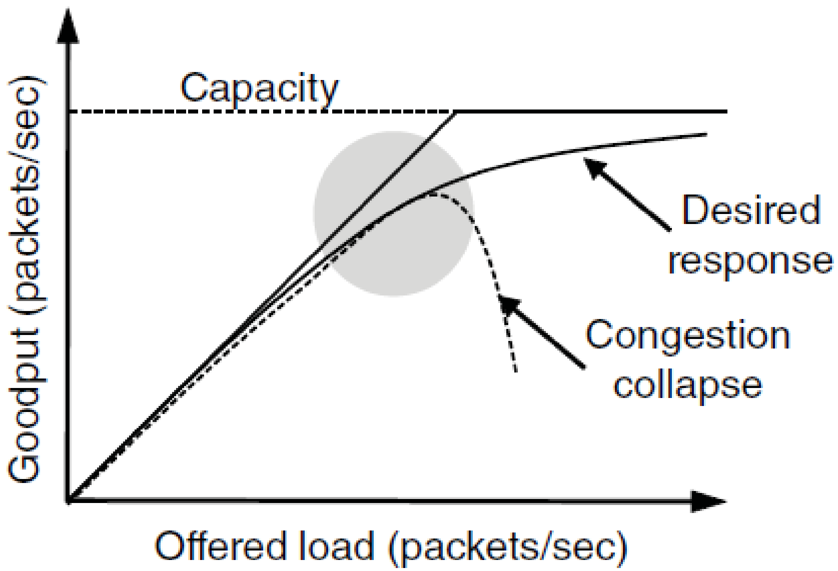
Two layers are responsible for congestion control:

- Transport layer, controls the offered load [here]
- Network layer, experiences congestion [previous]

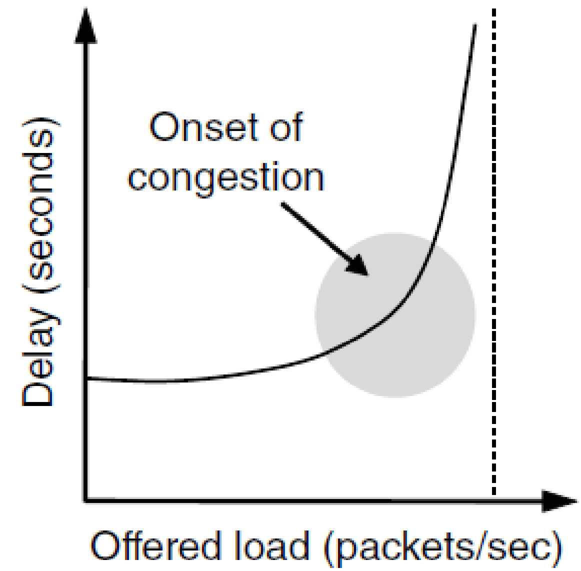
– Desirable bandwidth allocation »
– Regulating the sending rate »
– Wireless issues »

Desirable Bandwidth Allocation (1)

Efficient use of bandwidth gives high goodput, low delay



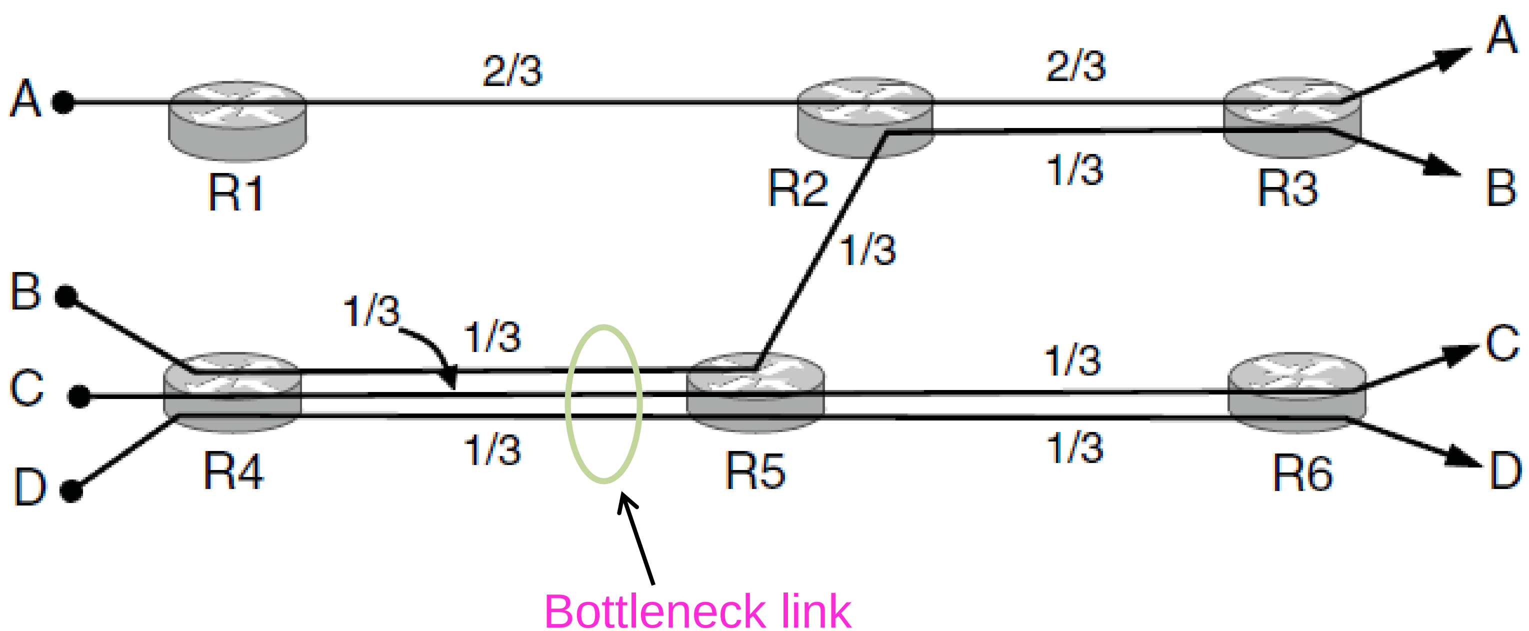
Goodput rises more slowly than load when congestion sets in



Delay begins to rise sharply when congestion sets in

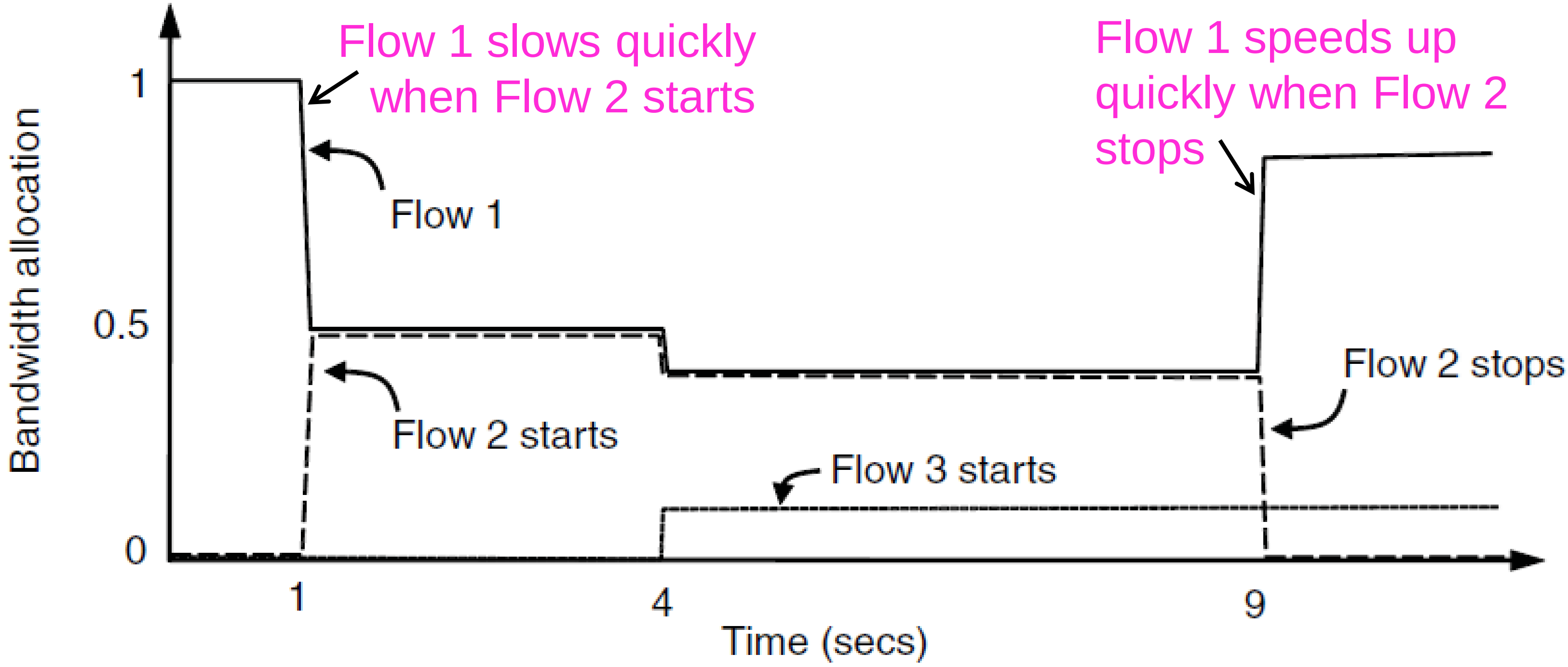
Desirable Bandwidth Allocation (2)

- Fair use gives bandwidth to all flows (no starvation)
  - Max-min fairness gives equal shares of bottleneck



Desirable Bandwidth Allocation (3)

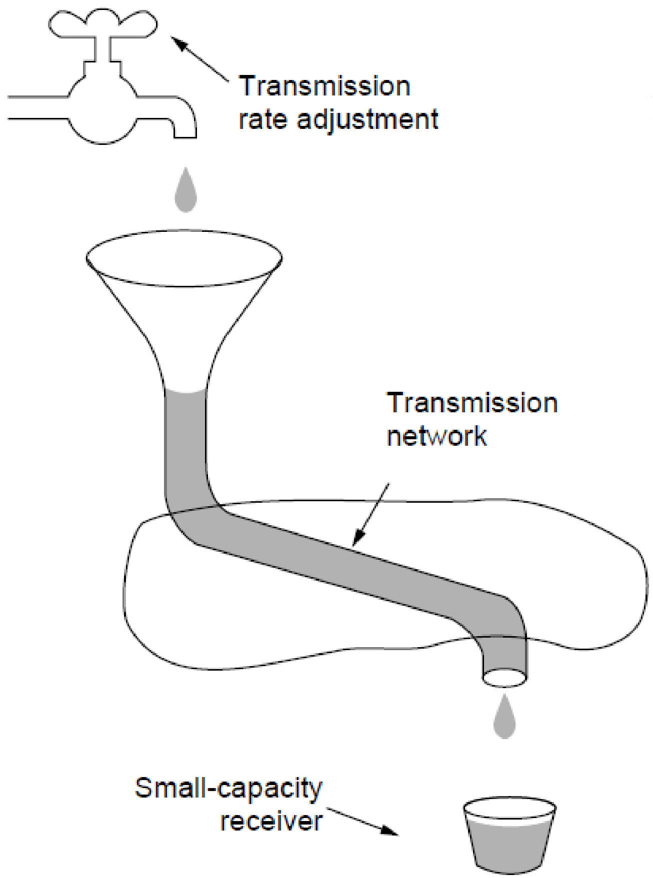
We want bandwidth levels to converge quickly when traffic patterns change



Regulating the Sending Rate (1)

Sender may need to slow down for different reasons:

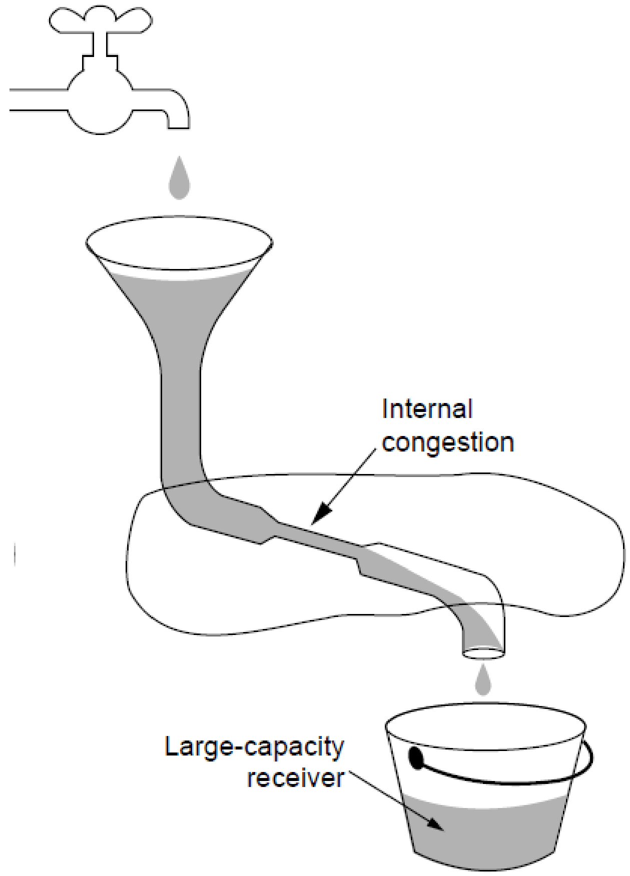
- Flow control, when the receiver is not fast enough [right]
- Congestion, when the network is not fast enough [over]



A fast network feeding a low-capacity receiver → flow control is needed

Regulating the Sending Rate (2)

Our focus is dealing with this problem – congestion



A slow network feeding a high-capacity receiver
→ congestion control is needed

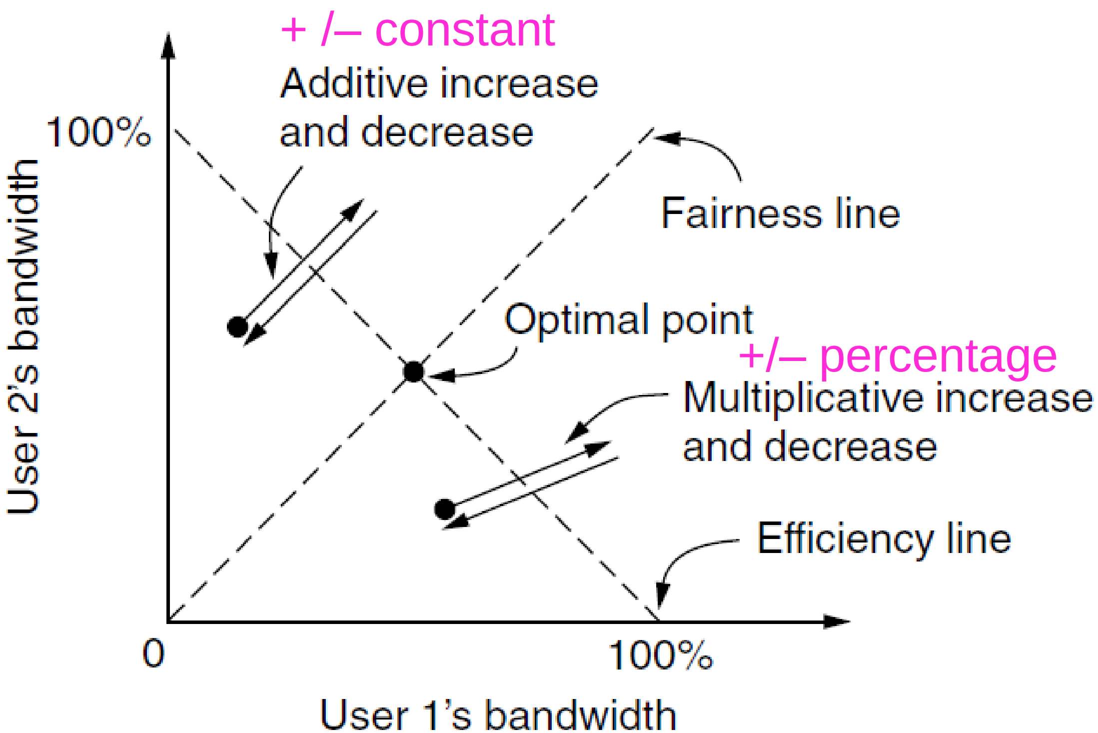
Regulating the Sending Rate (3)

Different congestion signals the network may use to tell the transport endpoint to slow down (or speed up)

Protocol	Signal	Explicit?	Precise?
XCP	Rate to use	Yes	Yes
TCP with ECN	Congestion warning	Yes	No
FAST TCP	End-to-end delay	No	Yes
CUBIC TCP	Packet loss	No	No
TCP	Packet loss	No	No

Regulating the Sending Rate (3)

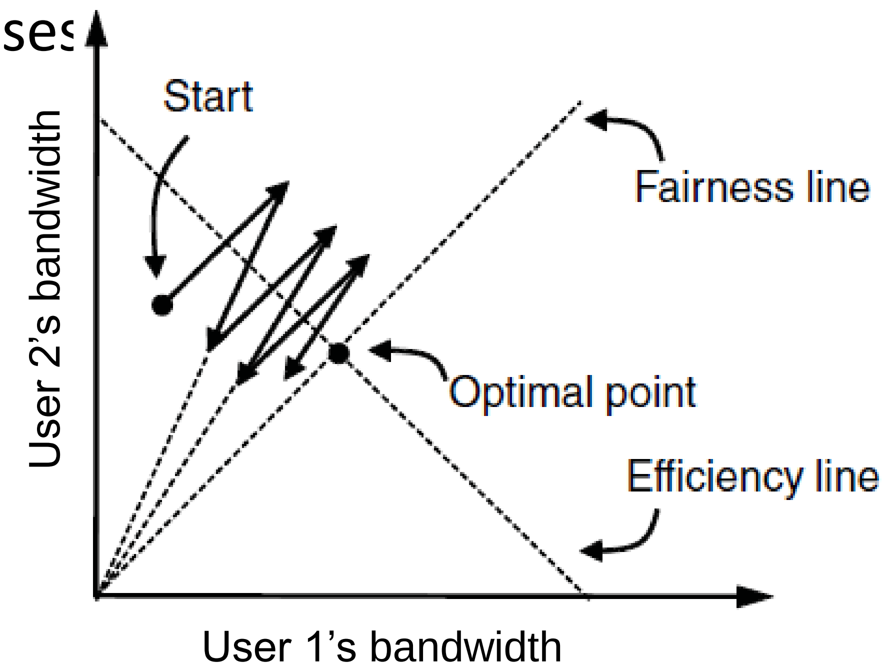
If two flows increase/decrease their bandwidth in the same way when the network signals free/busy they will not converge to a fair allocation



Regulating the Sending Rate (4)

The AIMD (Additive Increase Multiplicative Decrease) control law does converge to a fair and efficient point!

– TCP uses



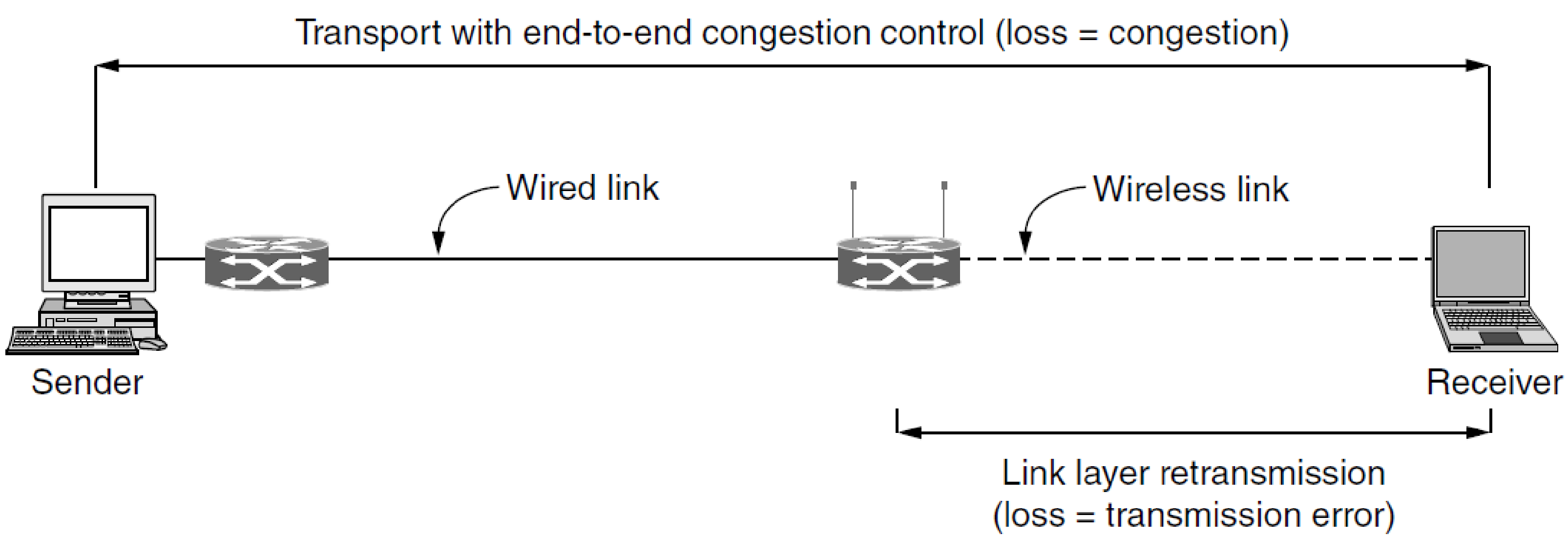
Wireless Issues

Wireless links lose packets due to transmission errors

- Do not want to confuse this loss with congestion
- Or connection will run slowly over wireless links!

Strategy:

- Wireless links use ARQ, which masks errors



A Simple Transport Protocol

- The Example Service Primitives
- The Example Transport Entity
- The Example as a Finite State Machine

Similar to TCP but simpler

Service Primitives

- Connect
  - Parameters – local and remote TSAPs
  - Caller is blocked
  - If connection succeeds the caller is unblocked and transmission starts
- Listen – specifies a TSAP to listen to
- Disconnect
- Send
- Receive
- ** Library procedures

Service Primitives

- Connum=LISTEN(local)
- Connum=Connect(local,remote)
- Status = Send(Connum, buffer,bytes)
  - No Connection, illegal buffer address, negative count
- Status = Receive(Connum, buffer, bytes)
- Status = Disconnect(Connum)

The Transport Entity

- Use connection oriented, reliable network service
- Transport Entity is part of the user process
- Network Layer interface
  - To_net and from_net
  - Parameters –
    - Connection Identifier
    - Q bit – control message
    - M bit – more data from this message to follow
    - Packet Type
    - Pointer to data

The Example Transport Entity

The network layer packets used in our example.

Network packet	Meaning
CALL REQUEST	Sent to establish a connection
CALL ACCEPTED	Response to CALL REQUEST
CLEAR REQUEST	Sent to release a connection
CLEAR CONFIRMATION	Response to CLEAR REQUEST
DATA	Used to transport data
CREDIT	Control packet for managing the window

The Example Transport Entity (2)

Each connection is in one of seven states:

1. Idle – Connection not established yet.
2. Waiting – CONNECT has been executed, CALL REQUEST sent.
3. Queued – A CALL REQUEST has arrived; no LISTEN yet.
4. Established – The connection has been established.
5. Sending – The user is waiting for permission to send a packet.
6. Receiving – A RECEIVE has been done.
7. DISCONNECTING – a DISCONNECT has been done locally.

State Transitions

- A primitive is executed
- A packet arrives
- A timer expires

Internet Protocols – UDP

– Introduction to UDP »
– Remote Procedure Call »
– Real-Time Transport »

User Datagram Protocol

- Connectionless
- Does not do
  - Flow control
  - Error control
  - Retransmissions
- Useful in client-server situations
- Sends segments consisting of an 8-byte header followed by the payload

Introduction to UDP (1)

UDP (User Datagram Protocol) is a shim over IP

- Header has ports (TSAPs), length and checksum.

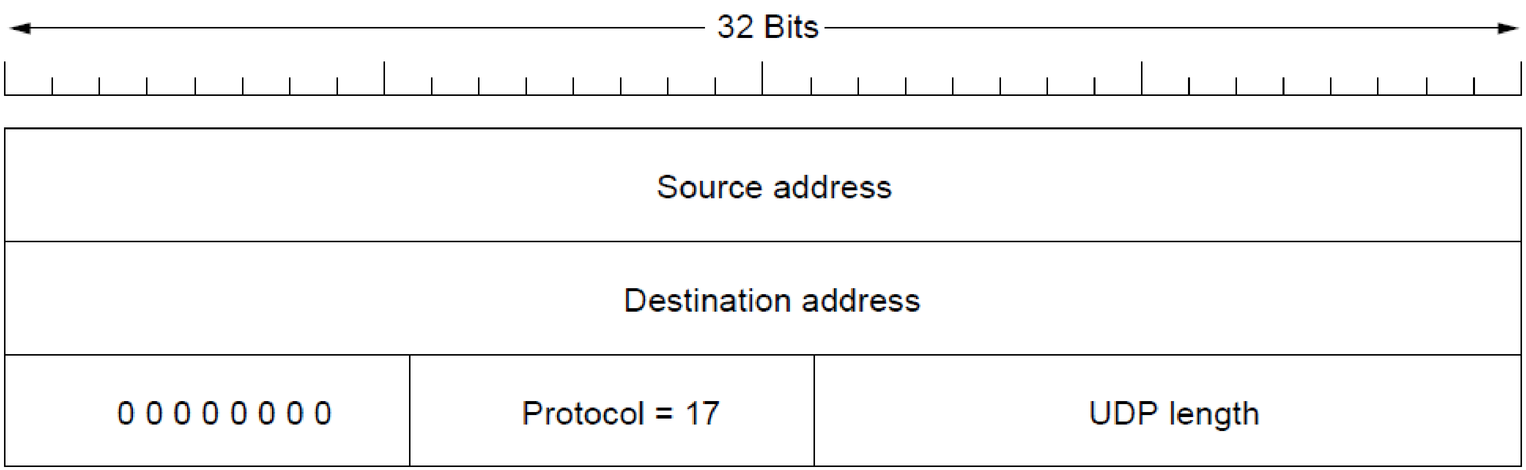
32 Bits

Source port	Destination port
UDP length	UDP checksum

Introduction to UDP (2)

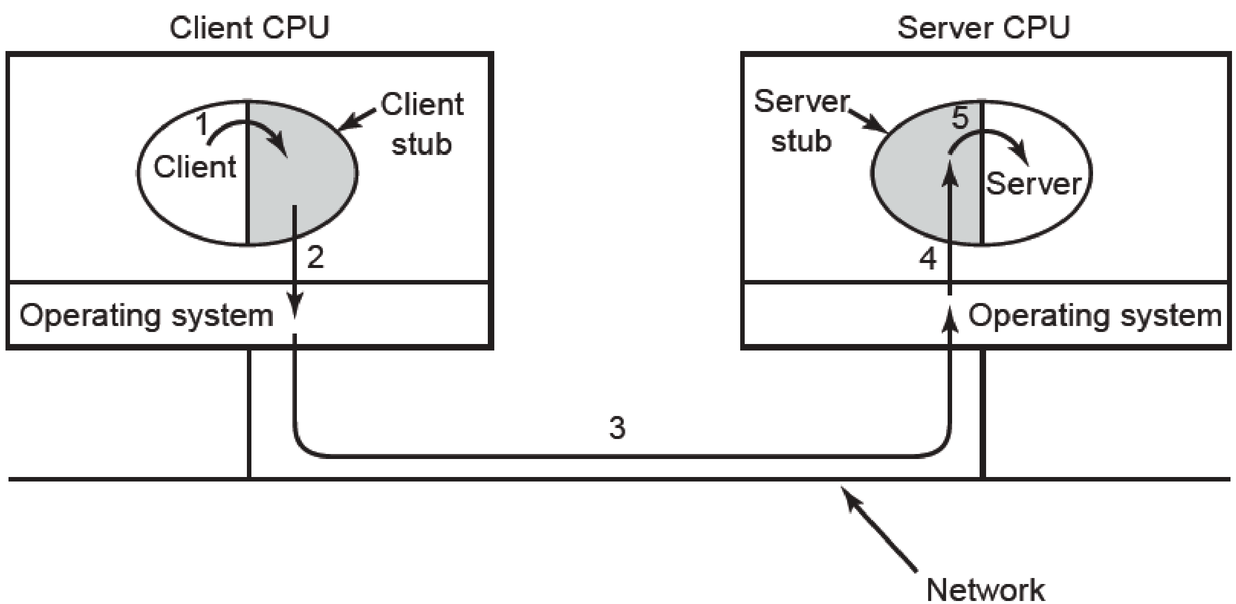
Checksum covers UDP segment and IP pseudoheader

- Fields that change in the network are zeroed out
- Provides an end-to-end delivery check



RPC (Remote Procedure Call)

- RPC connects applications over the network with the familiar abstraction of procedure calls
  - Stubs package parameters/results into a message
  - UDP with retransmissions is a low-latency transport



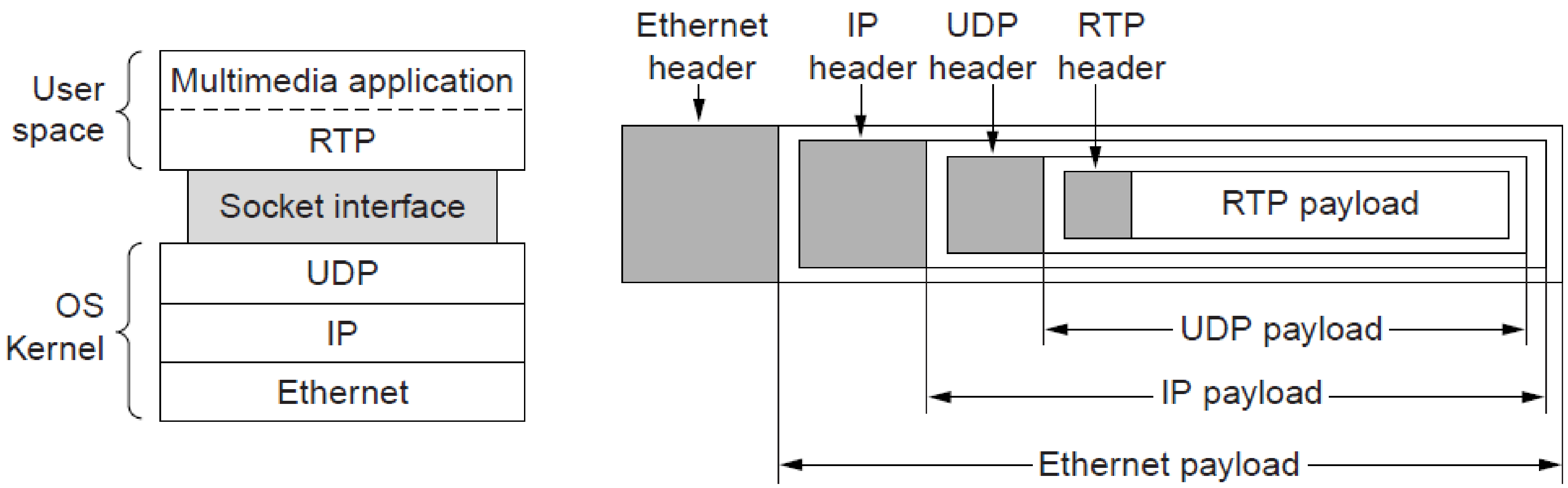
Limitations of RPC

- Pointers
- Weakly Typed languages – variable length arrays
- Not possible always to deduce parameter types
- Global variables

Real-Time Transport (1)

RTP (Real-time Transport Protocol) provides support for sending real-time media over UDP

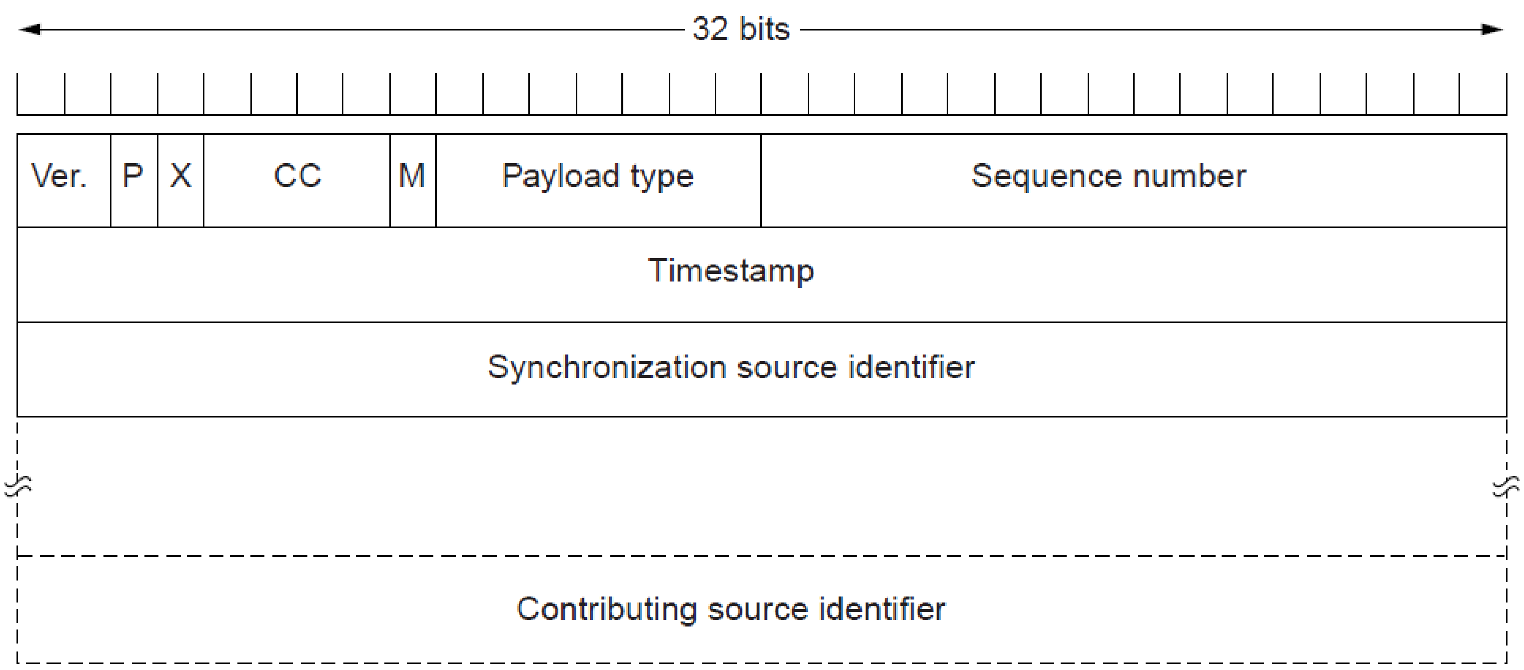
– Often implemented as part of the application



Real-Time Transport (2)

RTP header contains fields to describe the type of media and synchronize it across multiple streams

- RTCP sister protocol helps with management tasks



RTP Header Fields

- Ver – 2
- P – Packet padded to multiple of 4 bytes
- X – extension header present
- CC – number of contributing sources
- M bit – Application specific marker
- Payload Type – encoding used
- Sequence Number
- Time stamp – produced by the source
- Synchronizations Source Identifier – which stream the packet belongs to

RTP Profiles

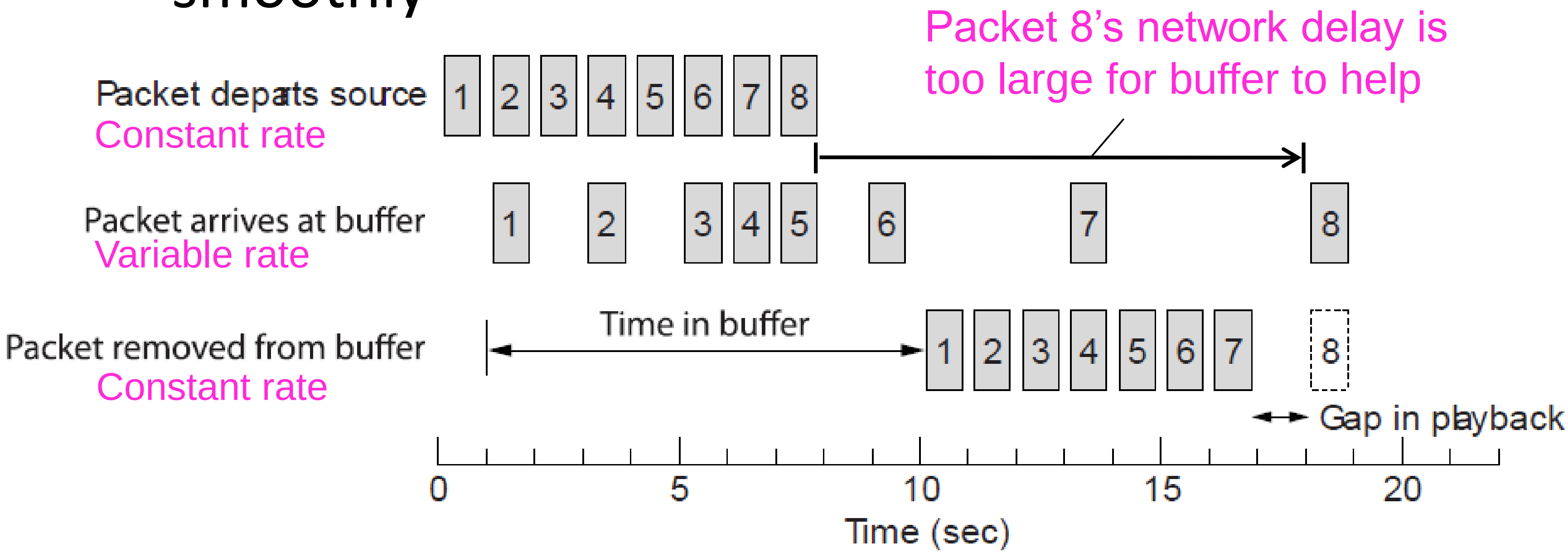
- RTP payloads may contain multiple samples coded in any way the application wants
- Profiles – to support interworking
  - Single Audio Stream
  - Multiple encoding formats may be supported
    - 8-bit pcm samples at 8KHz
    - Delta encoding
    - Predictive encoding
    - MP3
    - …

RTCP – Real-time Transport Control Protocol

- Control Protocol for RTP
- Does not transport any data
- Handles:
  - Feedback
    - Delay
    - Jitter
    - Bandwidth
    - Congestion, etc.
  - Synchronization
    - Interstream Synchronization – Different clocks, drifts, etc.
  - User Interface

Real-Time Transport (3)

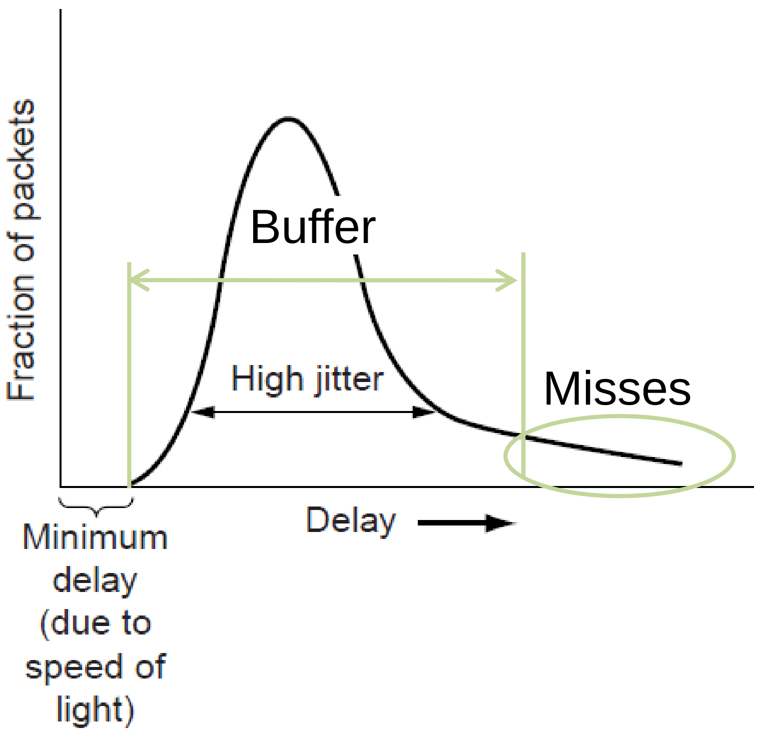
Buffer at receiver is used to delay packets and absorb jitter so that streaming media is played out smoothly

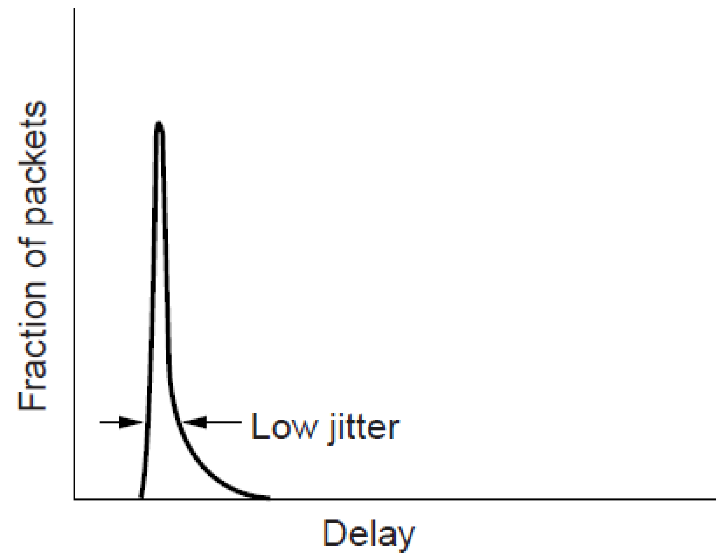


Real-Time Transport (3)

High jitter, or more variation in delay, requires a larger playout buffer to avoid playout misses

- Propagation delay does not affect buffer size





Internet Protocols – TCP

- The TCP service model »
- The TCP segment header »
- TCP connection establishment »
- TCP connection state modeling »
- TCP sliding window »
- TCP timer management »
- TCP congestion control »

TCP

- Reliable end-to-end byte stream over an unreliable internetwork
- Dynamically adapt to the properties of the network
- Robust

- RFC 793 – 1122 and 1323

TCP Transport Entity

- Implemented as
  - Library procedures
  - User process, or
  - Part of the kernel
- Accepts data streams from local processes
- Breaks them into segments of >64KB (usually 1460 bytes)
- Sends each piece in a separate IP packet
- On receive side – reconstruct the original byte stream and give to a process.
- Must recover from errors – time outs, retransmissions, etc.

TCP Service Model

- End points called sockets
  - Socket number
    - IP address of the host
    - 16-bit number (called the *port*)
- Connection Oriented – Full Duplex, Point-to-Point
  - Establish a connection between sockets
  - An socket may be used for multiple connections at the same time
  - Connection (*socket1, socket2*)

Berkeley Sockets

The socket primitives for TCP.

Primitive	Meaning
SOCKET	Create a new communication end point
BIND	Attach a local address to a socket
LISTEN	Announce willingness to accept connections; give queue size
ACCEPT	Block the caller until a connection attempt arrives
CONNECT	Actively attempt to establish a connection
SEND	Send some data over the connection
RECEIVE	Receive some data from the connection
CLOSE	Release the connection

The TCP Service Model (1)

TCP provides applications with a reliable byte stream between processes; it is the workhorse of the Internet

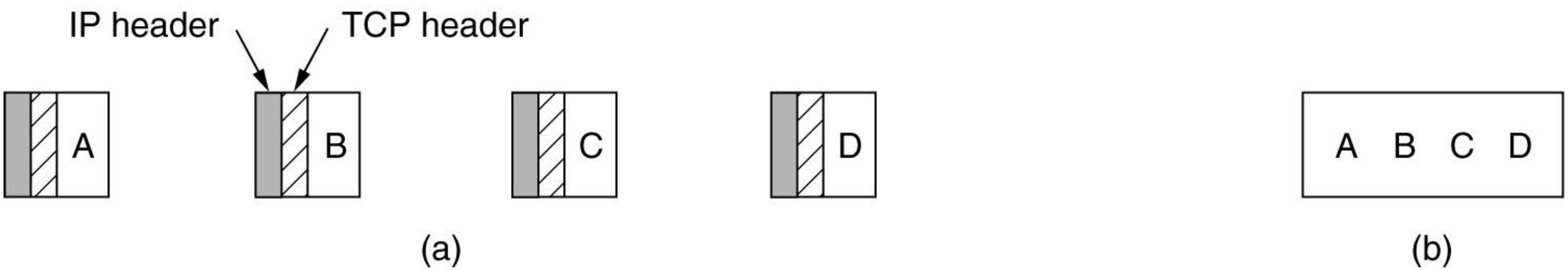
- Popular servers run on well-known ports

Port	Protocol	Use
20, 21	FTP	File transfer
22	SSH	Remote login, replacement for Telnet
25	SMTP	Email
80	HTTP	World Wide Web
110	POP-3	Remote email access
143	IMAP	Remote email access
443	HTTPS	Secure Web (HTTP over SSL/TLS)
543	RTSP	Media player control
631	IPP	Printer sharing

Internet Daemon

- Attaches to multiple well-known ports and waits
- When a connection comes in it forks off a new process and executes the appropriate daemon
- That daemon handles the request

The TCP Service Model



(a) Four 512-byte segments sent as separate IP datagrams.

(b) The 2048 bytes of data delivered to the application in a single READ CALL.

TCP Service

- No message boundaries are preserved
- Send data as received or buffer it
- PUSH Flag
  - Send data now
  - Useful in sending command from terminal
- Urgent Data
  - DEL or CTRL-C to break off a remote computation
  - Use URGENT flag – Transmit everything right now
    - Receiving application is interrupted

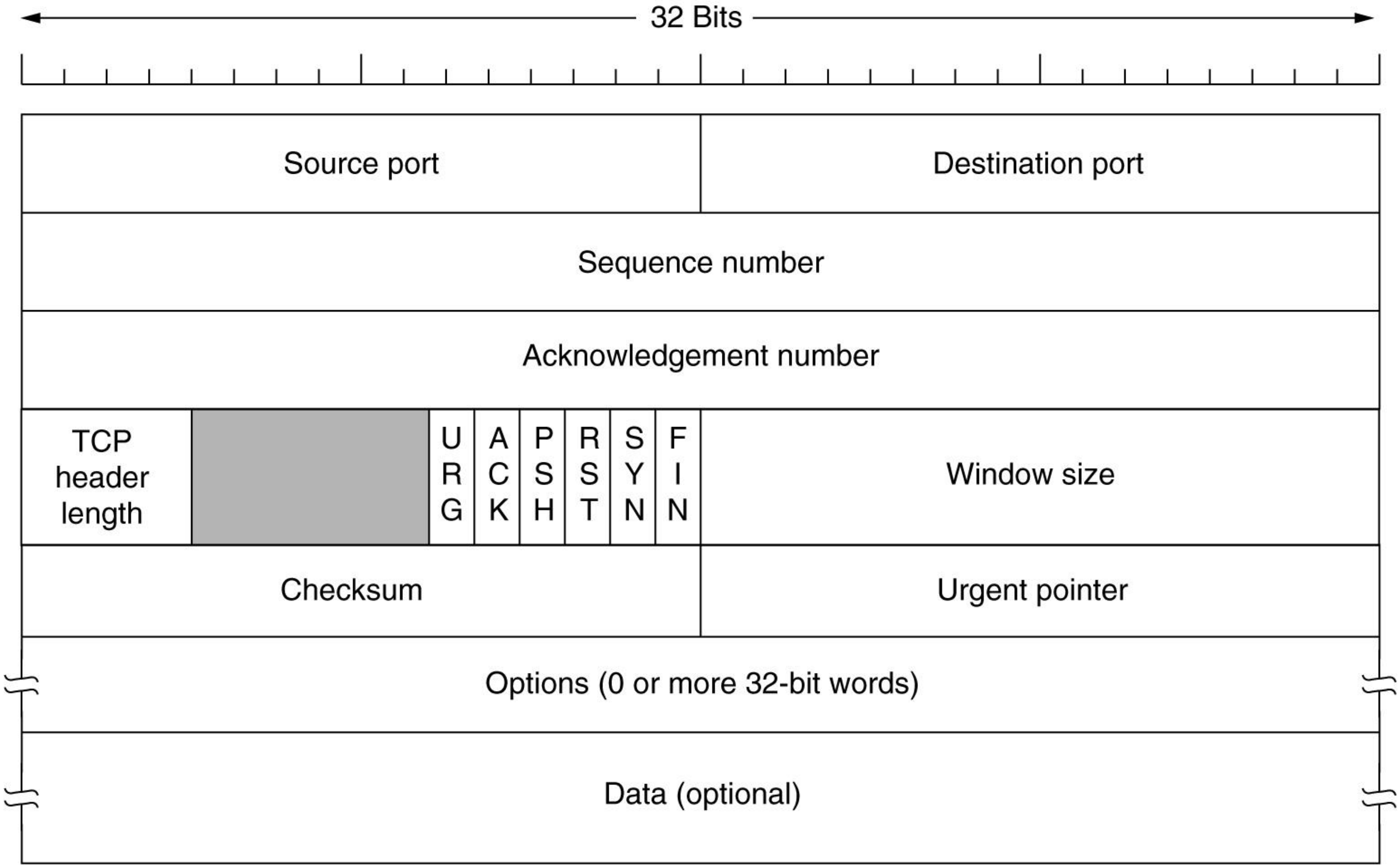
TCP Protocol

- Exchange segments
  - 20 byte header (plus optional parts)
  - 0 or more data bytes
- Accumulate data from several writes into one segment
- May split data from one write into multiple segments
- Each segment, including the header, must be <65515 byte IP payload
- Each network has MTU- Maximum Transfer Unit
  - Each segment must be less than or equal to MTU

TCP Protocol

- TCP uses sliding window protocol
- Sequence numbers are for bytes not segments
- Sending – start a timer
- Receiving – send an ack with sequence number = next sequence number expected

The TCP Segment Header



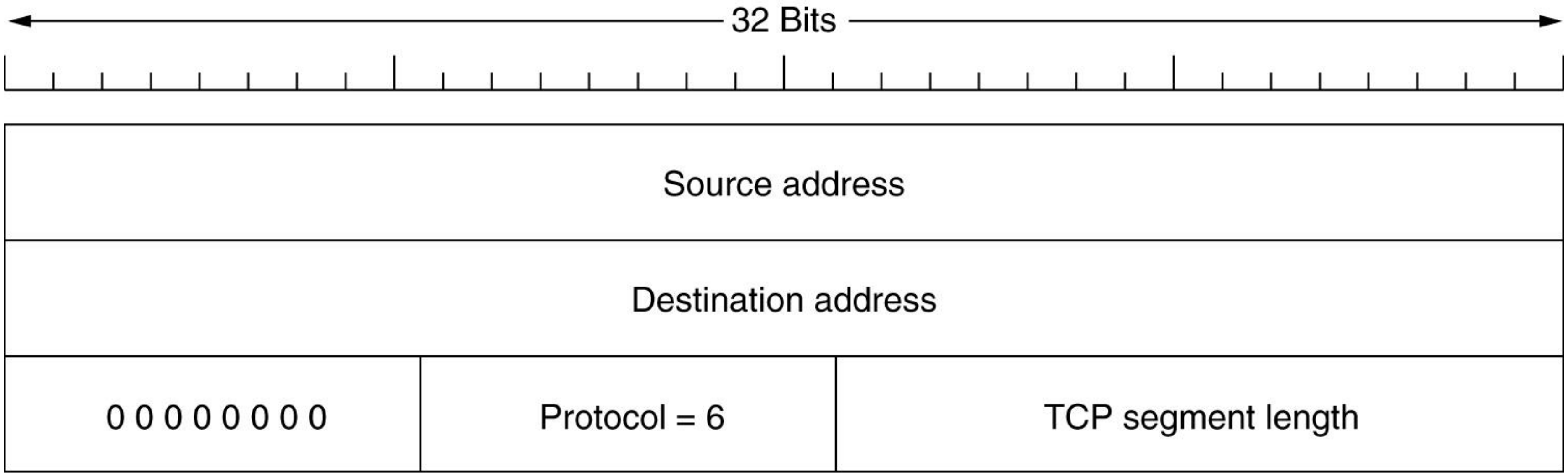
TCP Header.

TCP Segment Header

- Source and Destination Port
- Sequence number
- Acknowledgement Number
  - Next byte expected
- TCP Header Length – Number of 32 bit words in TCP header
- Flags
  - URG – set to 1 if Urgent Pointer is in use
    - Used to indicate a byte offset from the current seq no at which urgent data care there
  - ACK – set to 1 when ack no is valid
  - PSH – Push bit
  - RST – Reset
  - SYN – Used for connection establishment
  - FIN – Used to close a connection

The TCP Segment Header (2)

- The pseudoheader included in the TCP checksum
- Checksum the header, the data and the pseudoheader
- Add all 16-bit words in 1's complement and then take 1's complement of the sum
- To check calculate on the entire segment and result should be 0.



TCP Window

- Window size tells how many bytes may be sent starting at the byte acknowledged
  - If 0 means do not send now.
  - May send a segment with same ack no and non-zero window size.

Maximum segment Size

- All hosts are required to accept TCP segments of 536+20 =556 bytes
- May negotiate max segment size using options.
- Another negotiable parameter – Window Scale
  - May shift to the left by up to 14 bits
  - Giving a max window size of 2^{30} bytes

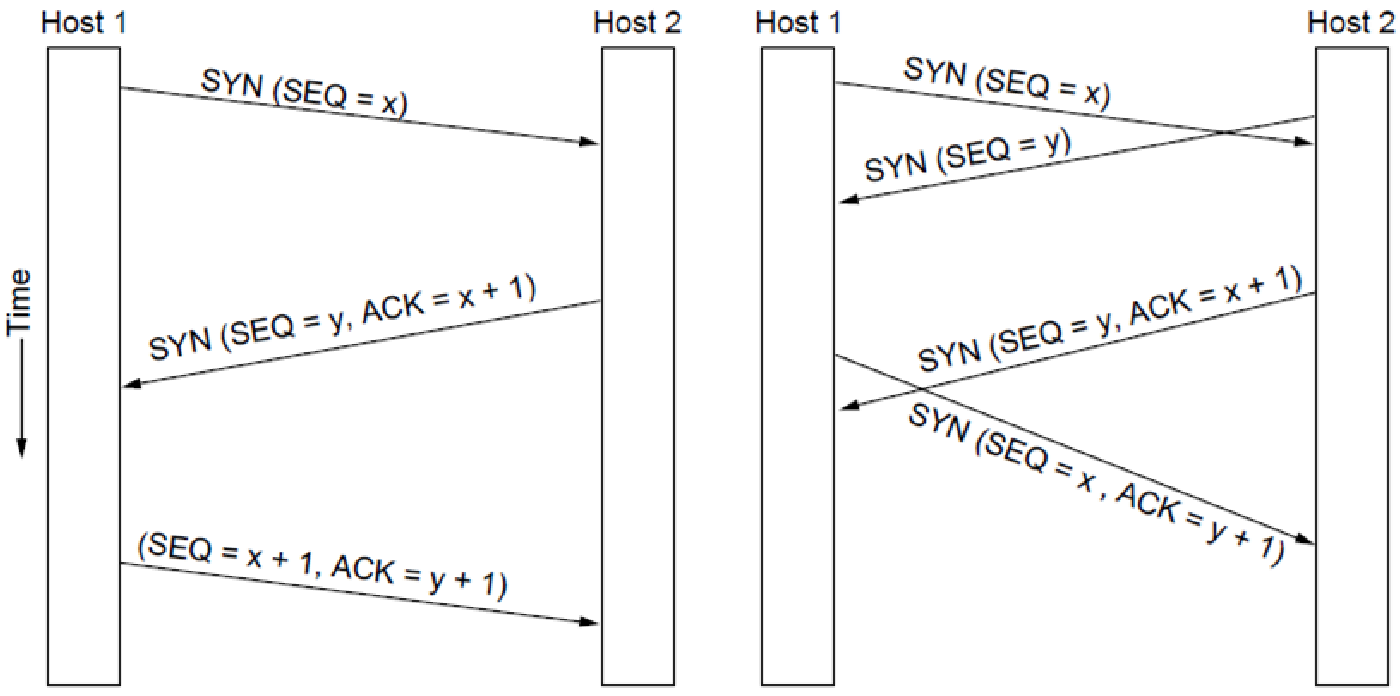
Connection Establishment

- Uses three-way handshake
- Server passively listens
  - LISTEN and ACCEPT primitives
- Client executes CONNECT
  - Send a TCP Segment with SYN bit on and ACK bit off.
- Check to see if there is a process listening
  - If not send a RST
  - If yes, then give the segment to the process
  - If accepted – send an ACK message

TCP Connection Establishment

TCP sets up connections with the three-way handshake

- Release is symmetric, also as described before



Normal case

Simultaneous connect

TCP Connection State Modeling (1)

The TCP connection finite state machine has more states than our simple example from earlier.

State	Description
CLOSED	No connection is active or pending
LISTEN	The server is waiting for an incoming call
SYN RCVD	A connection request has arrived; wait for ACK
SYN SENT	The application has started to open a connection
ESTABLISHED	The normal data transfer state
FIN WAIT 1	The application has said it is finished
FIN WAIT 2	The other side has agreed to release
TIME WAIT	Wait for all packets to die off
CLOSING	Both sides have tried to close simultaneously
CLOSE WAIT	The other side has initiated a release
LAST ACK	Wait for all packets to die off

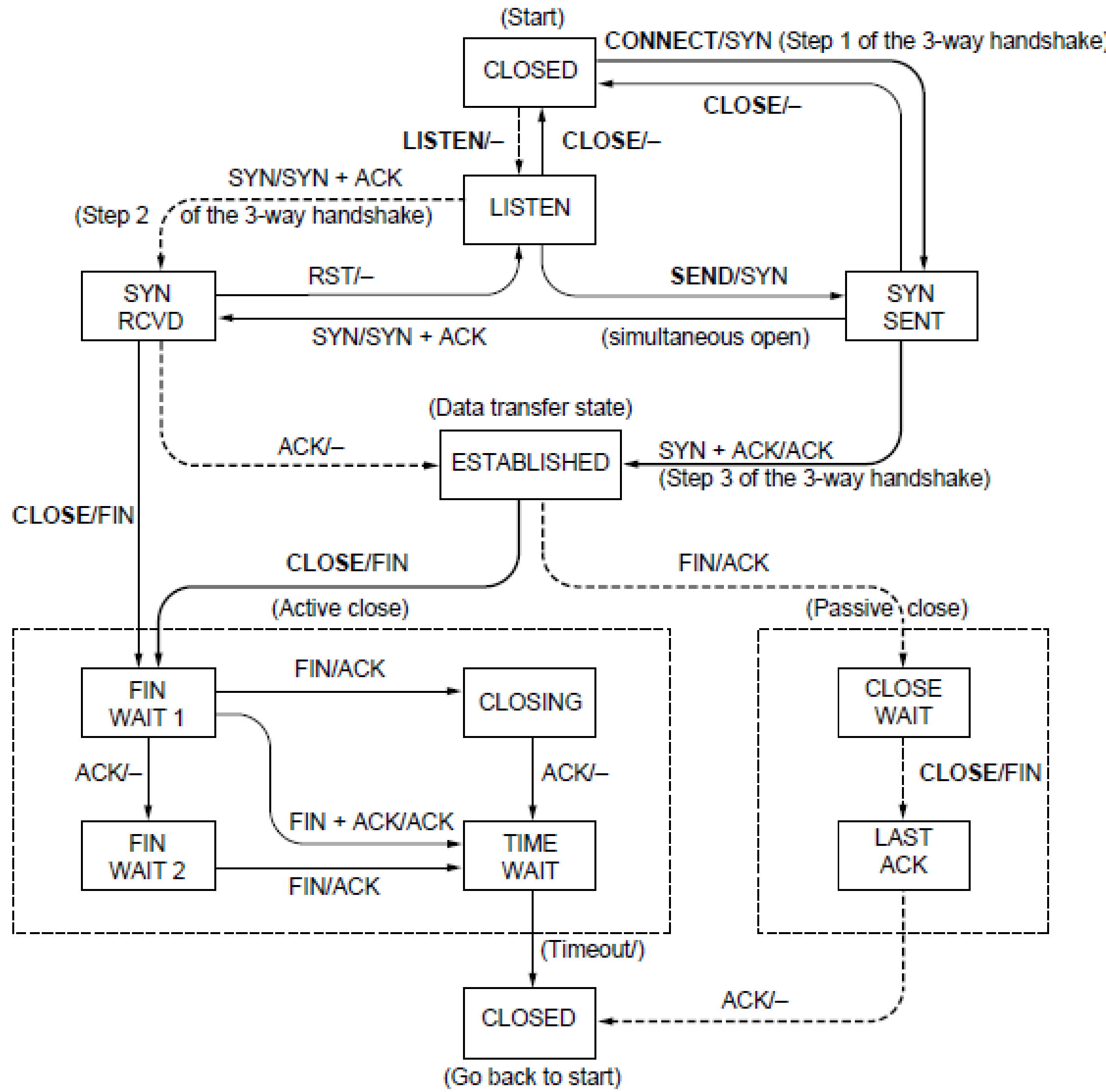
TCP Connection State Modeling (2)

Solid line is the normal path for a client.

Dashed line is the normal path for a server.

Light lines are unusual events.

Transitions are labeled by the cause and action, separated by a slash.



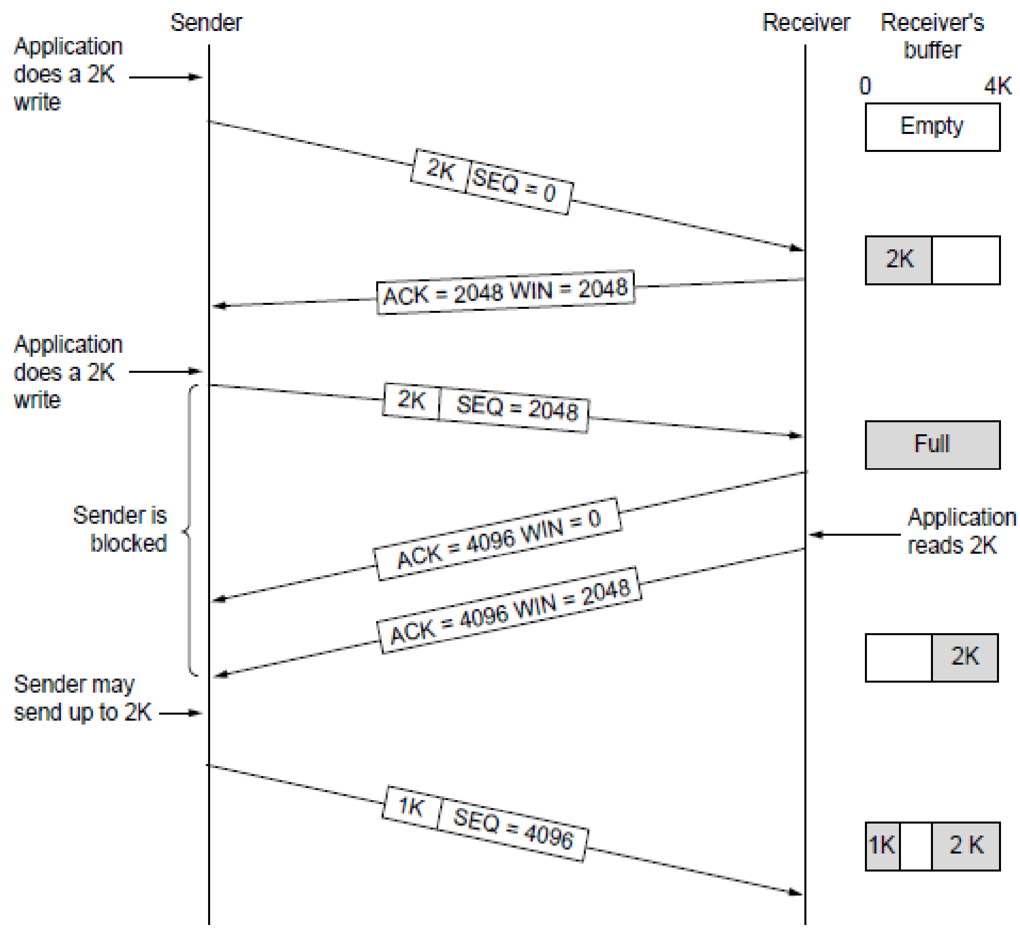
Connection Release

- Think of the connection as a pair of simplex connections
  - Each is released independently
- Either party sends a segment with FIN bit set
- When FIN acked that direction is shut down

TCP Sliding Window (1)

TCP adds flow control to the sliding window as before

- ACK + WIN is the sender's limit



Sliding Window

When Window Size = 0

- Sender stops sending, except
  - Urgent Data
  - Window Probe –
    - One Byte Segment – forcing receiver to re-announce the next byte expected as Window size.

Example Situation Telnet

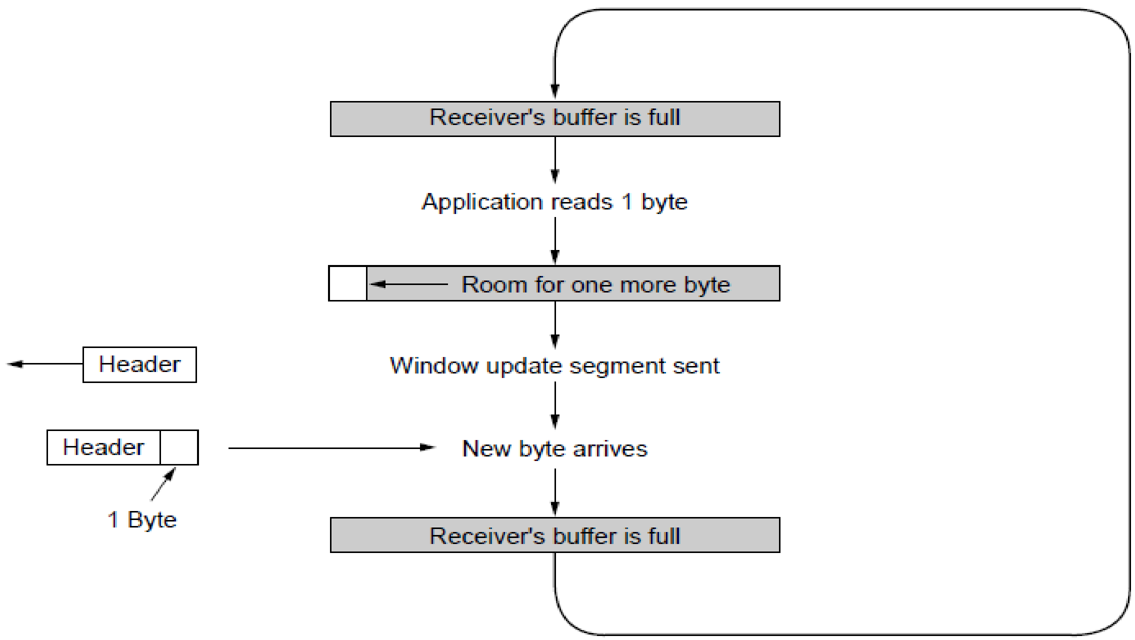
- 1 Byte
  - 21 Byte Segment
  - 41 Byte Packet
- 40 Byte Ack
- 41 Byte Echo
- 40 Byte Ack

162 Bytes on the network for one byte data

Delayed Ack - 500 ms

TCP Sliding Window (2)

Need to add special cases to avoid unwanted behavior



Receiver application reads single bytes, so sender always sends one byte segments

Handling Silly Window Problem

- Delay ack and window updates for 500 ms.
- Nagle's Algorithm
  - When data comes in one byte at a time
    - Send the first byte and buffer the rest till the outstanding byte is acknowledged
    - Then send all the buffered characters in one TCP segment
  - Mouse movements have to be sent – Burst does not work well.
- Clark's Solution
  - Wait until decent amount of space available then advertise
    - Max segment size or half buffer
  - Sender not send tiny segments

Timer Management

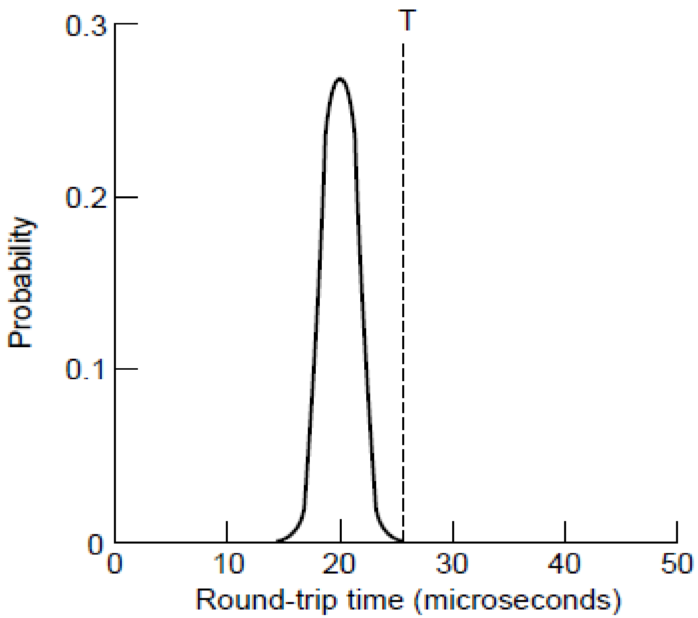
- TCP uses multiple timers
  - Most important is the *retransmission timer*
  - What value to set it at??
- Round Trip time
  - Highly variable
  - Varies over time
  - Have to track it
  - Estimate it
  - M is a new measurement
  - α = 7/8
  - Use βRTT for retransmission timer
  - Initial values of β were 2 – make is proportional to standard deviation of M

$$RTT = \alpha RTT + (1-\alpha)M$$

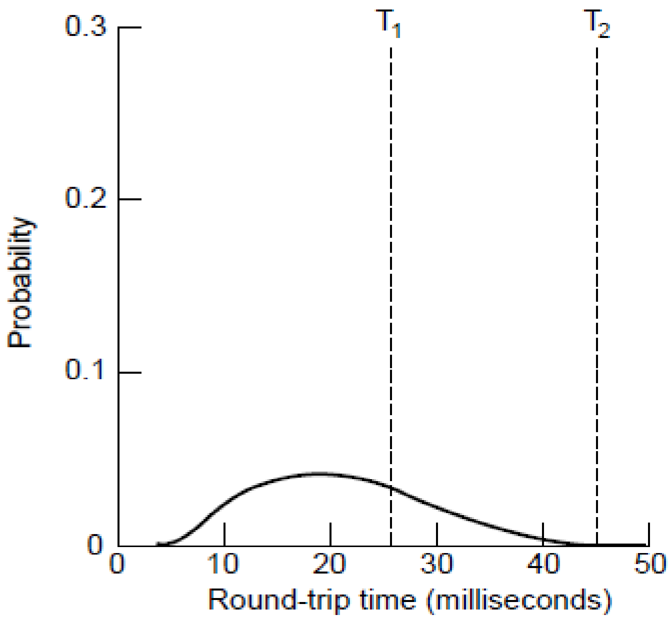
TCP Timer Management

TCP estimates retransmit timer from segment RTTs

- Tracks both average and variance (for Internet case)
- Timeout is set to average plus 4 x variance



LAN case – small, regular RTT



Internet case – large, varied RTT

Timer Management

- Jacobson Approach
- Mean deviation estimate
- $D = \gamma D + (1-\gamma)\ |RTT - M|$
- Timeout = RTT + 4 D

- What to do on retransmissions
  - Do not know if the ack is for the first or second
- Karn Algorithm
  - Do not update RTT on any segments that have been retransmitted

Persistence Timer

- Receiver sends a window of 0
- Later sends a window size but that packet is lost
  - Both wait
- Persistence Timer
  - When it goes off – sender sends a probe request to receiver to get a window size
  - If still zero – continue to wait and reset persistence timer
- Keepalive Timer
  - When a connection is idle for a long time – check if the other side is still there

TCP Congestion Control

- Congestion – a function of total number of packets in the network, and where they are
- First step – detection
  – Is packet loss an indication of congestion??
  – All TCP algorithms assume timeouts are caused by congestion
- Initial steps
  – When connection is established – use suitable window size
    - Loss will not occur due to buffers at receiver
- Two issues
  – Network Capacity
  – Receiver Capacity

TCP Congestion Control

- Network Capacity and Receiver Capacity
- Maintain two windows
  - Receiver window
  - Congestion window
  - Use the min (Receiver window and Congestion window)
- Initially
  - Sender sets congestion window to MSS (Max Seg Size)
  - If acked add one more MSS – 2 now
  - Repeat for each acked MSS
  - Congestion window grows exponentially
  - If timeout – go back to previous window size
  - SLOW START

Internet Congestion Control

- Use a Threshold – initially 64 KB
- When a timeout occurs set threshold to half the current congestion window and reset congestion window to 1 MSS
- Use slow start till the threshold is reached
- Then successful transmissions grow congestion window linearly

TCP Congestion Control (1)

TCP uses AIMD with loss signal to control congestion

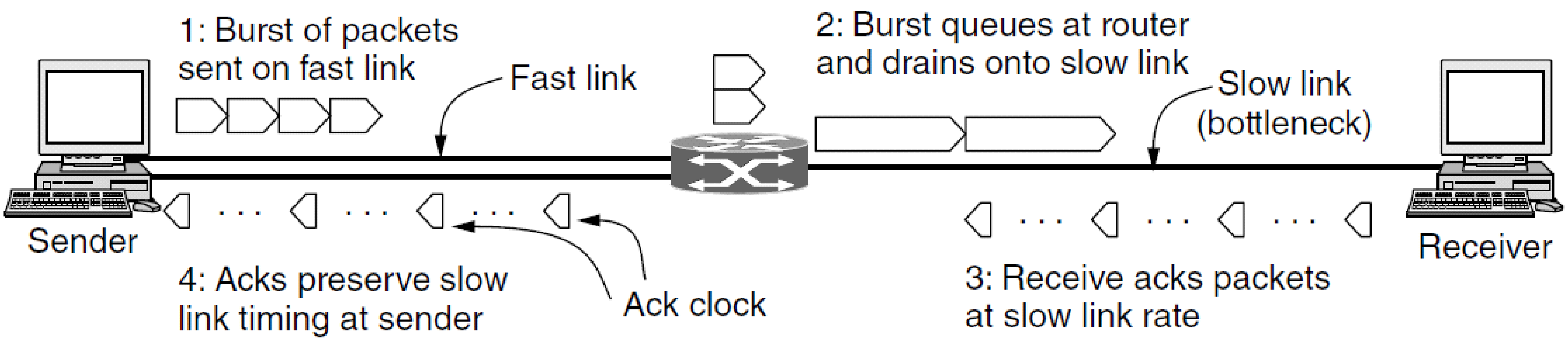
- Implemented as a congestion window (cwnd) for the number of segments that may be in the network
- Uses several mechanisms that work together

Name	**Mechanism**	**Purpose**
ACK clock	Congestion window (cwnd)	Smooth out packet bursts
Slow-start	Double cwnd each RTT	Rapidly increase send rate to reach roughly the right level
Additive Increase	Increase cwnd by 1 packet each RTT	Slowly increase send rate to probe at about the right level
Fast retransmit / recovery	Resend lost packet after 3 duplicate ACKs; send new packet for each new ACK	Recover from a lost packet without stopping ACK clock

TCP Congestion Control (2)

Congestion window controls the sending rate

- Rate is cwnd / RTT; window can stop sender quickly
- ACK clock (regular receipt of ACKs) paces traffic and smoothes out sender bursts



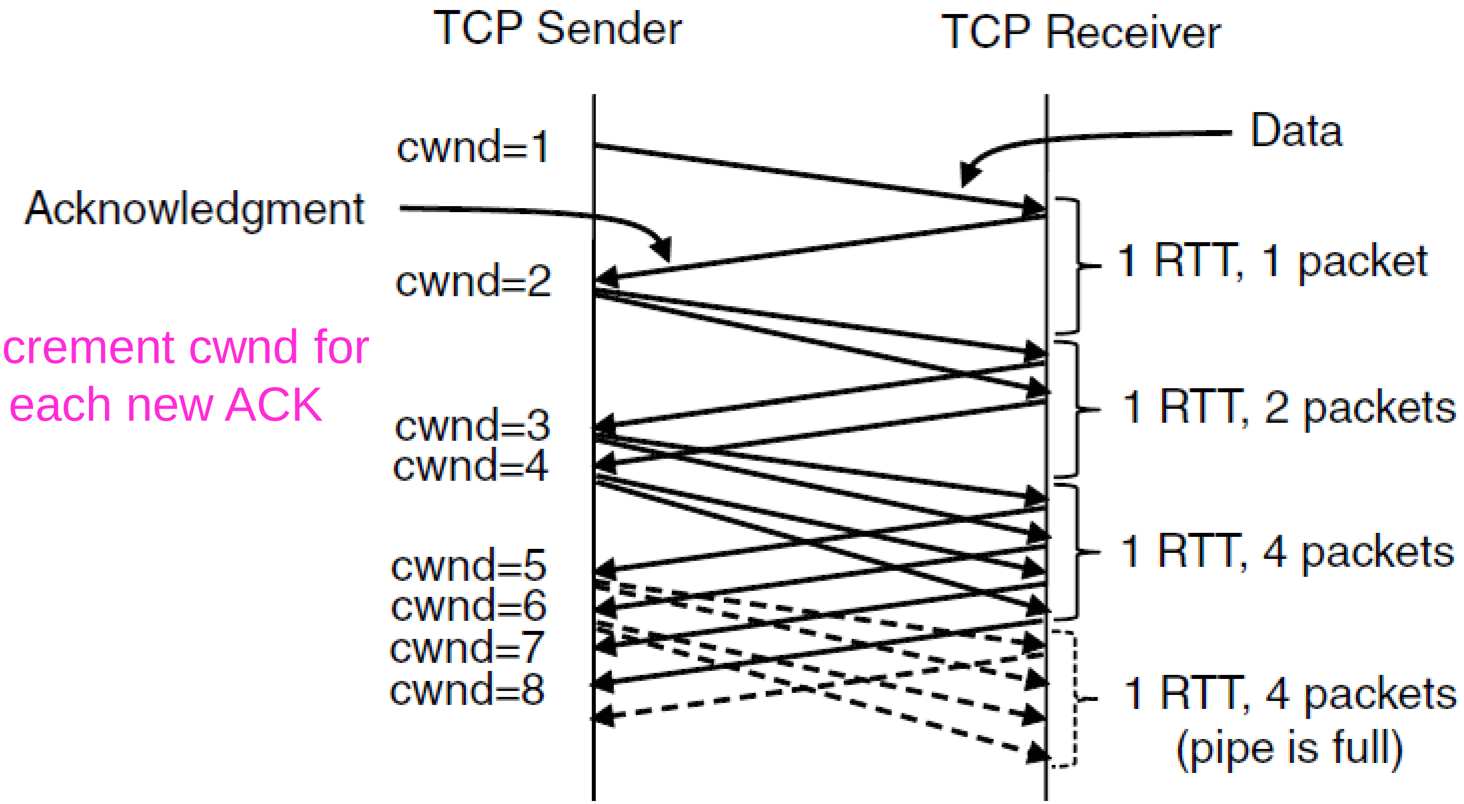
ACKs pace new segments into the network and smooth bursts

TCP Congestion Control (3)

Slow start grows congestion window exponentially

- Doubles every RTT while keeping ACK clock going

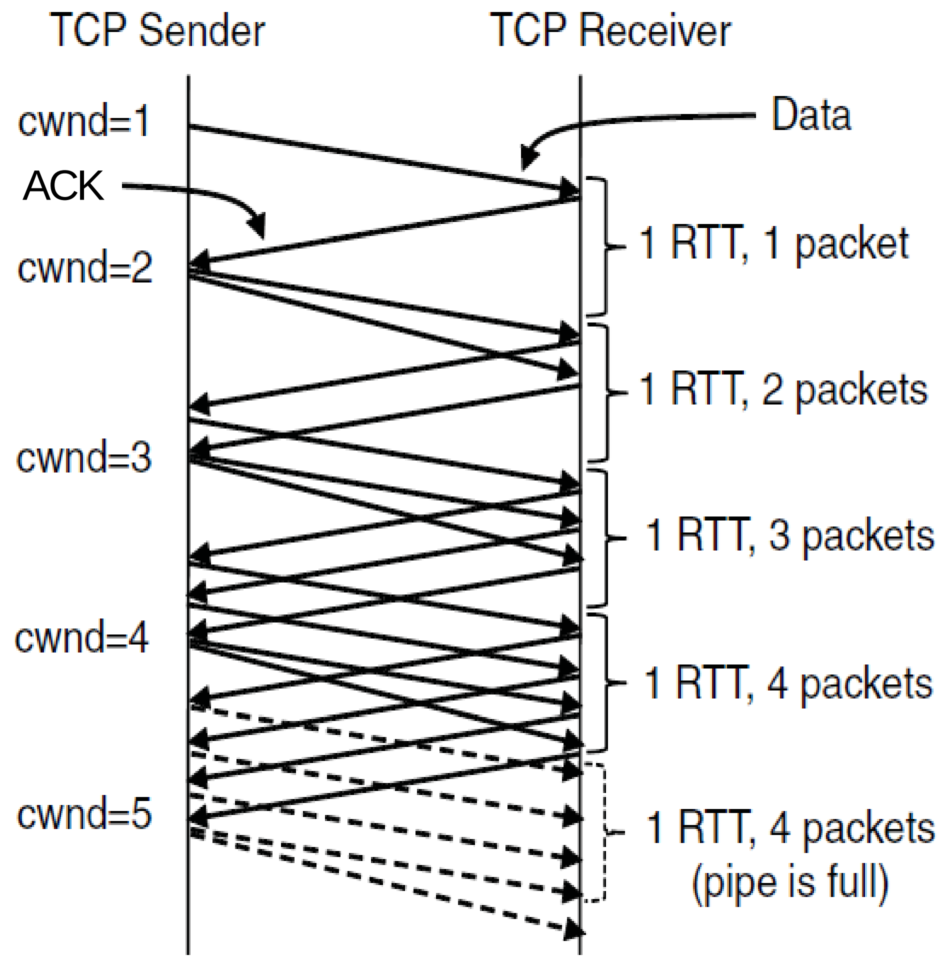
Increment cwnd for each new ACK



TCP Congestion Control (4)

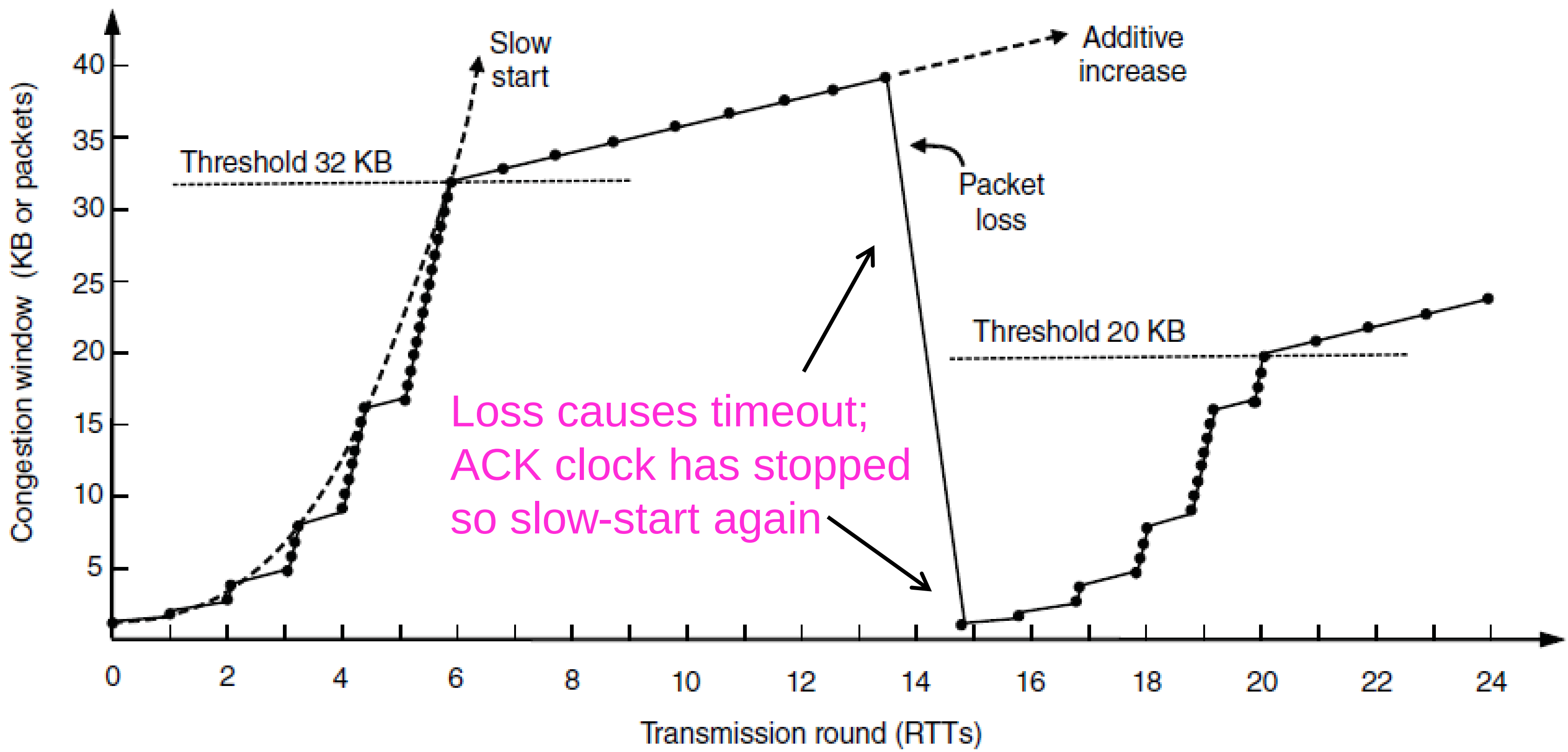
Additive increase grows cwnd slowly

- Adds 1 every RTT
- Keeps ACK clock



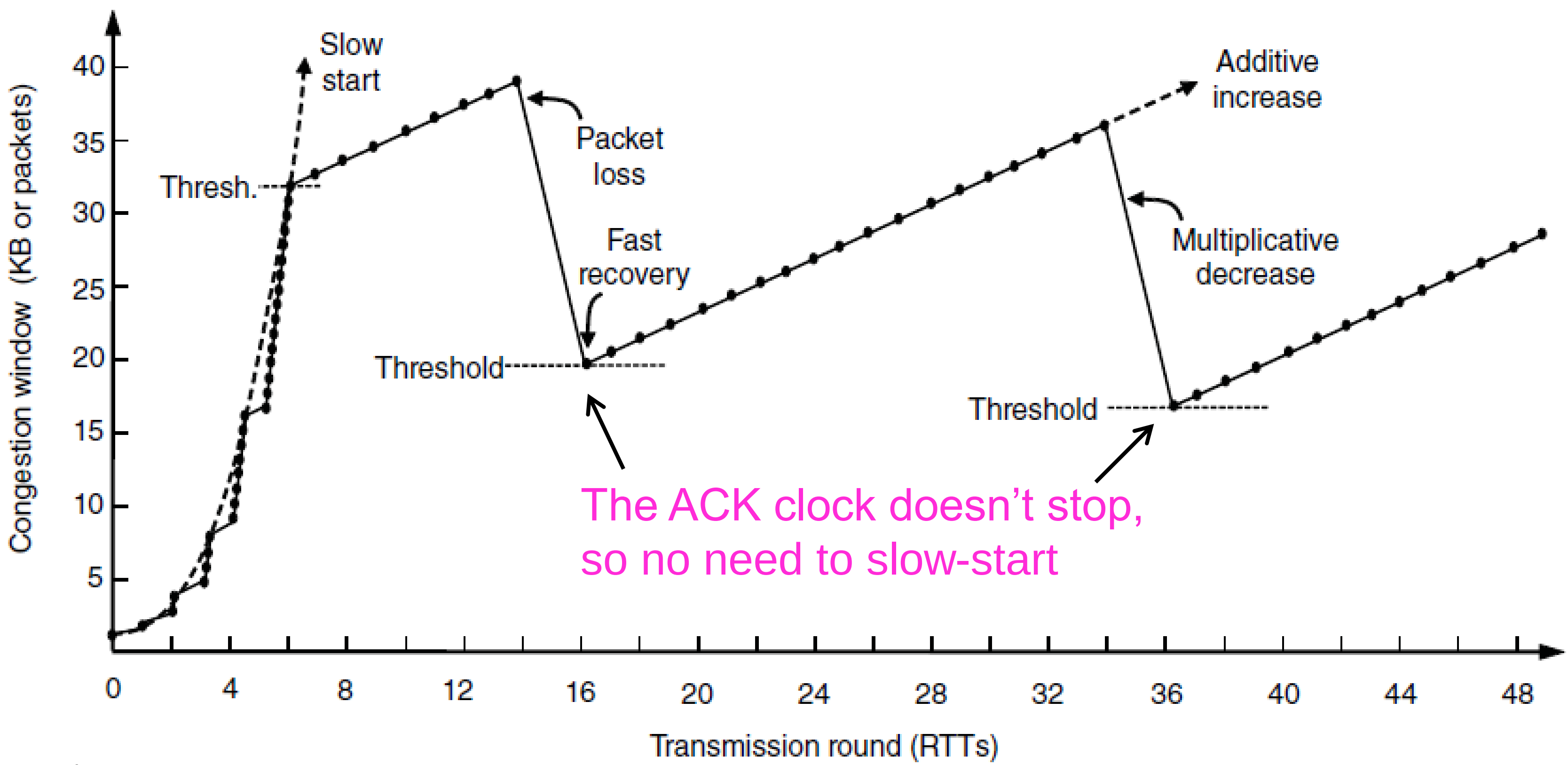
TCP Congestion Control (5)

- Slow start followed by additive increase (TCP Tahoe)
  - Threshold is half of previous loss cwnd



TCP Congestion Control (6)

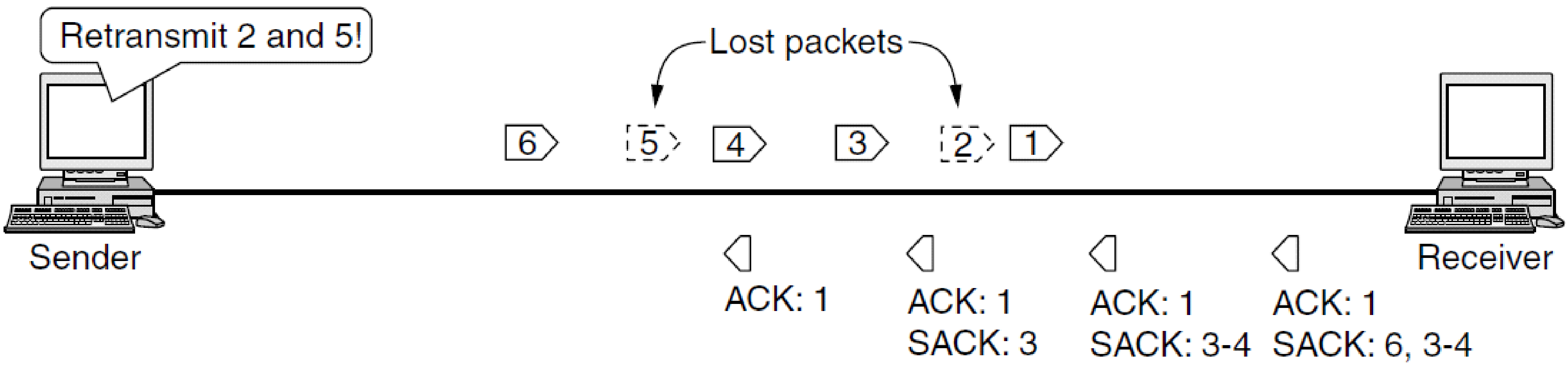
- With fast recovery, we get the classic sawtooth (TCP Reno)
  - Retransmit lost packet after 3 duplicate ACKs



TCP Congestion Control (7)

SACK (Selective ACKs) extend ACKs with a vector to describe received segments and hence losses

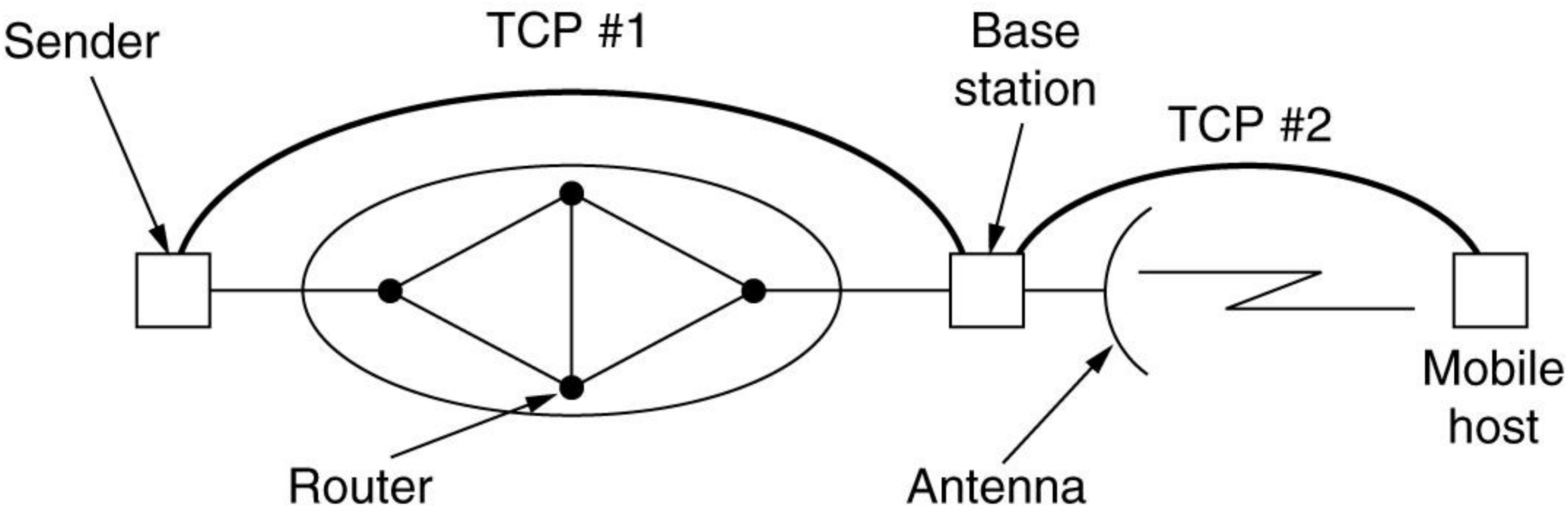
- Allows for more accurate retransmissions / recovery



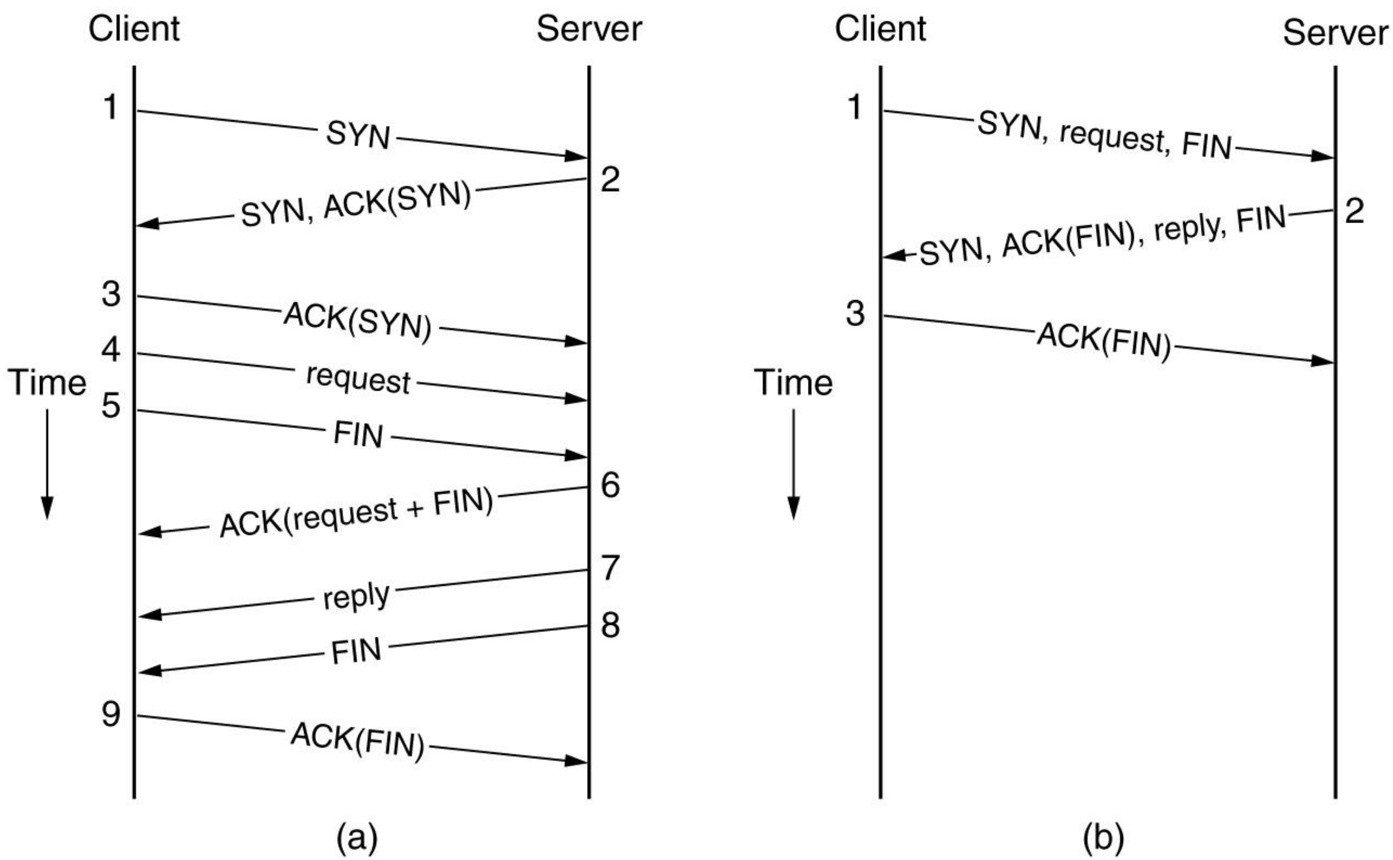
No way for us to know that 2 and 5 were lost with only ACKs

Wireless TCP and UDP

Splitting a TCP connection into two connections.



Transactional TCP



(a) RPC using normal TPC.

(b) RPC using T/TCP.

Performance Issues

Many strategies for getting good performance have been learned over time

- Performance problems »
- Measuring network performance »
- Host design for fast networks »
- Fast segment processing »
- Header compression »
- Protocols for "long fat" networks »

Performance Problems

Unexpected loads often interact with protocols to cause performance problems

- Need to find the situations and improve the protocols

Examples:

- Broadcast storm: one broadcast triggers another
- Synchronization: a building of computers all contact the DHCP server together after a power failure
- Tiny packets: some situations can cause TCP to send many small packets instead of few large ones

Host Design for Fast Networks

Poor host software can greatly slow down networks.

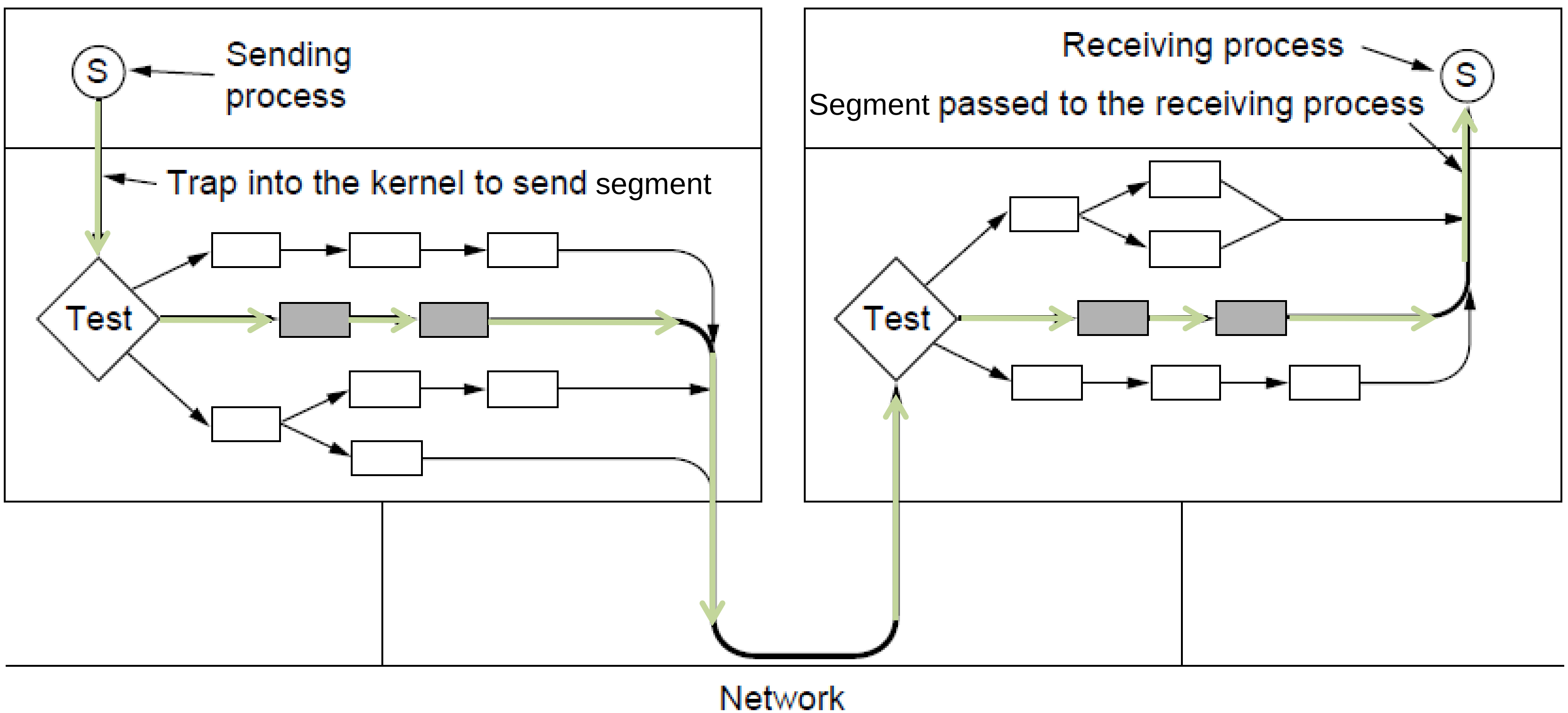
Rules of thumb for fast host software:

- Host speed more important than network speed
- Reduce packet count to reduce overhead
- Minimize data touching
- Minimize context switches
- Avoiding congestion is better than recovering from it
- Avoid timeouts

Fast Segment Processing (1)

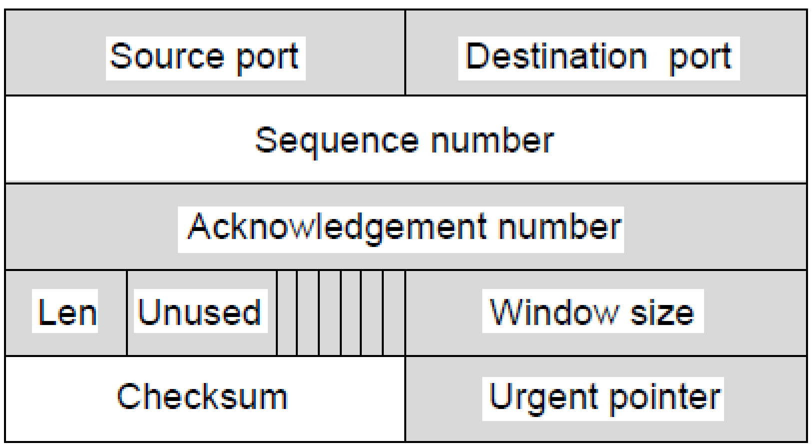
Speed up the common case with a fast path [pink]

- Handles packets with expected header; OK for others to run slowly

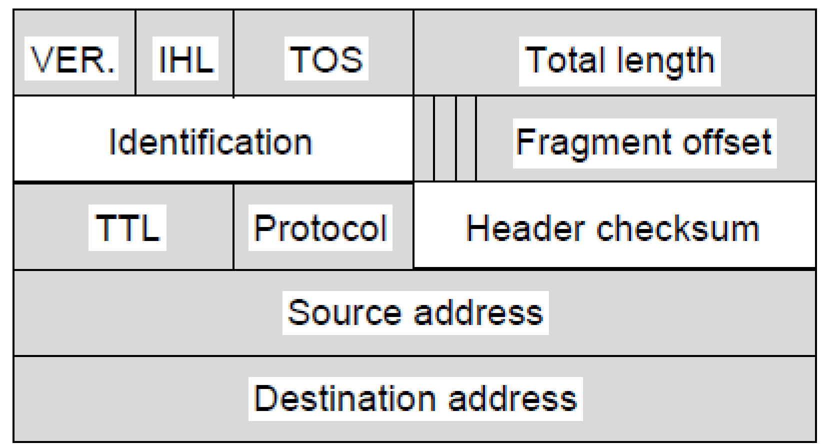


Fast Segment Processing (2)

Header fields are often the same from one packet to the next for a flow; copy/check them to speed up processing

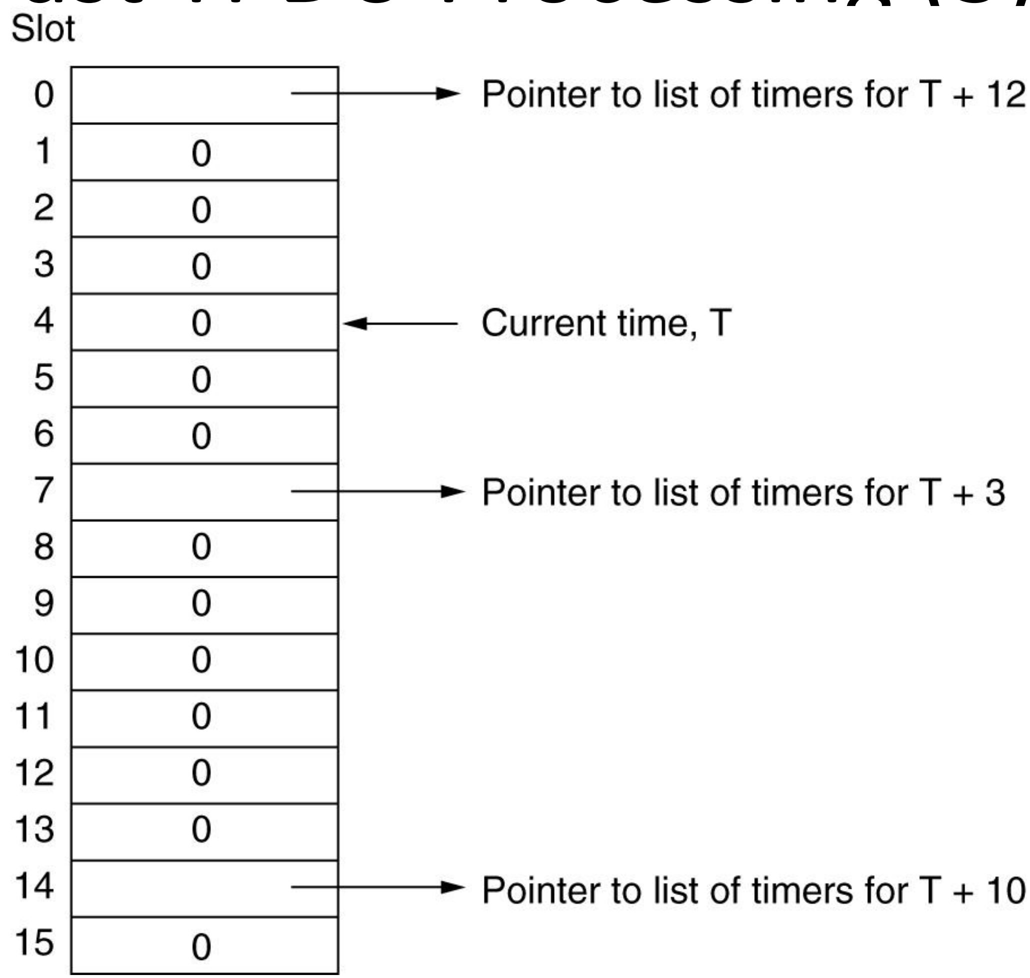


TCP header fields that stay the same for a one-way flow (shaded)



IP header fields that are often the same for a one-way flow (shaded)

Fast TPDU Processing (3)



A timing wheel.

Header Compression

Overhead can be very large for small packets

- 40 bytes of header for RTP/UDP/IP VoIP packet
- Problematic for slow links, especially wireless

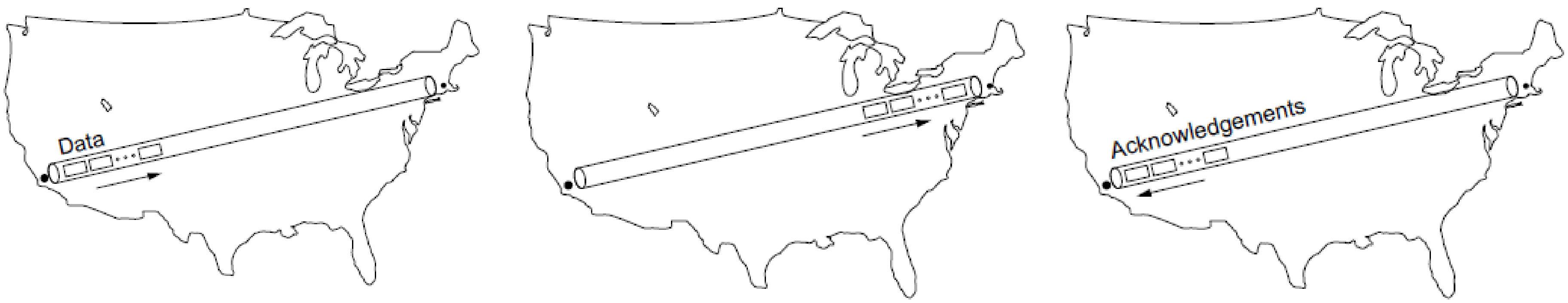
Header compression mitigates this problem

- Runs between Link and Network layer
- Omits fields that don't change or change predictably
  - 40 byte TCP/IP header → 3 bytes of information
- Gives simple high-layer headers and efficient links

Protocols for "Long Fat" Networks (1)

Networks with high bandwidth ("Fat") and high delay ("Long") can store much information inside the network

– Requires protocols with ample buffering and few RTTs, rather than reducing the bits on the wire



Starting to send 1 Mbit San Diego → Boston

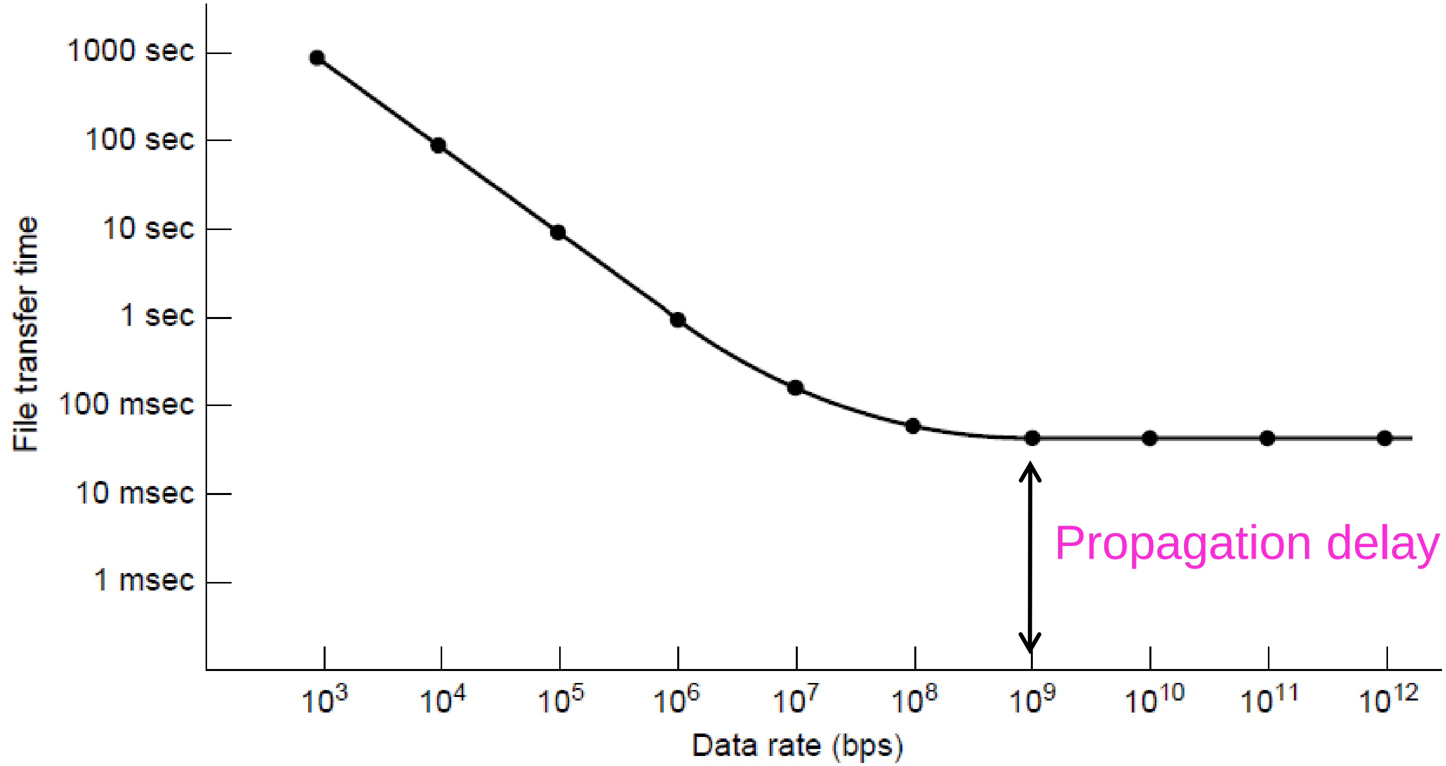
20ms after start

40ms after start

Protocols for "Long Fat" Networks (2)

You can buy more bandwidth but not lower delay

- Need to shift ends (e.g., into cloud) to lower further



Minimum time to send and ACK a 1-Mbit file over a 4000-km line

Delay Tolerant Networking

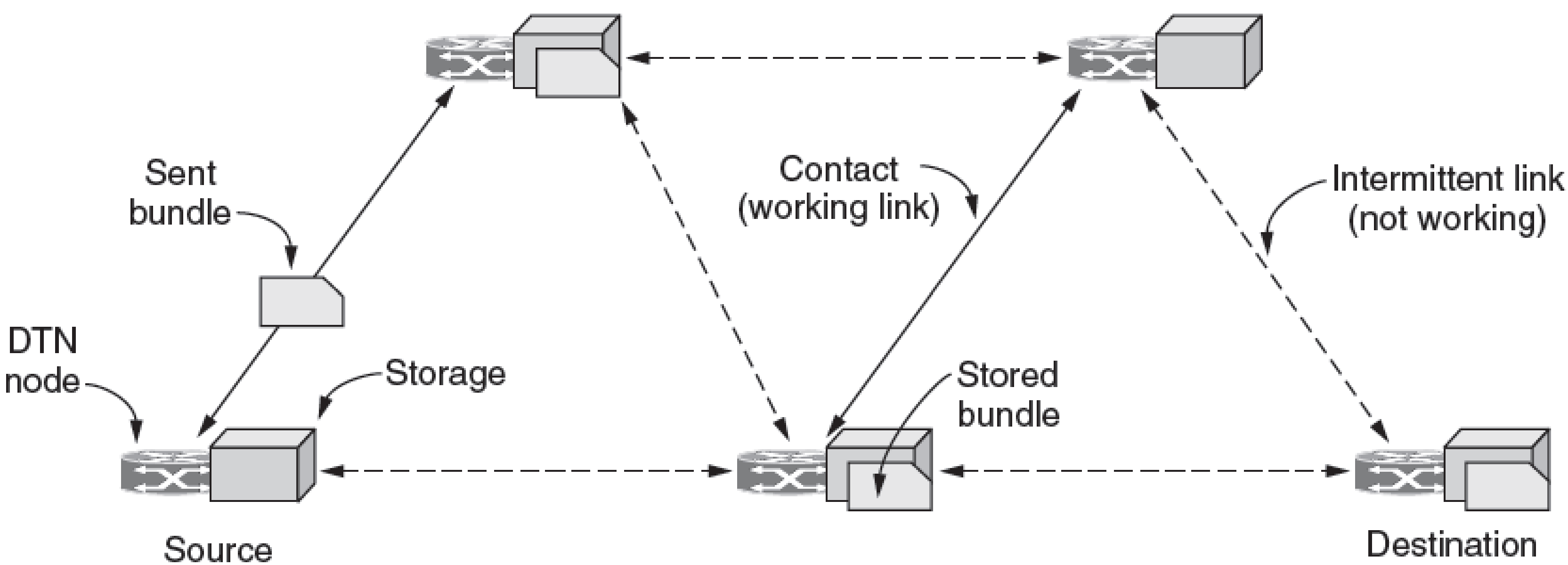
DTNs (Delay Tolerant Networks) store messages inside the network until they can be delivered

- DTN Architecture »
- Bundle Protocol »

DTN Architecture (1)

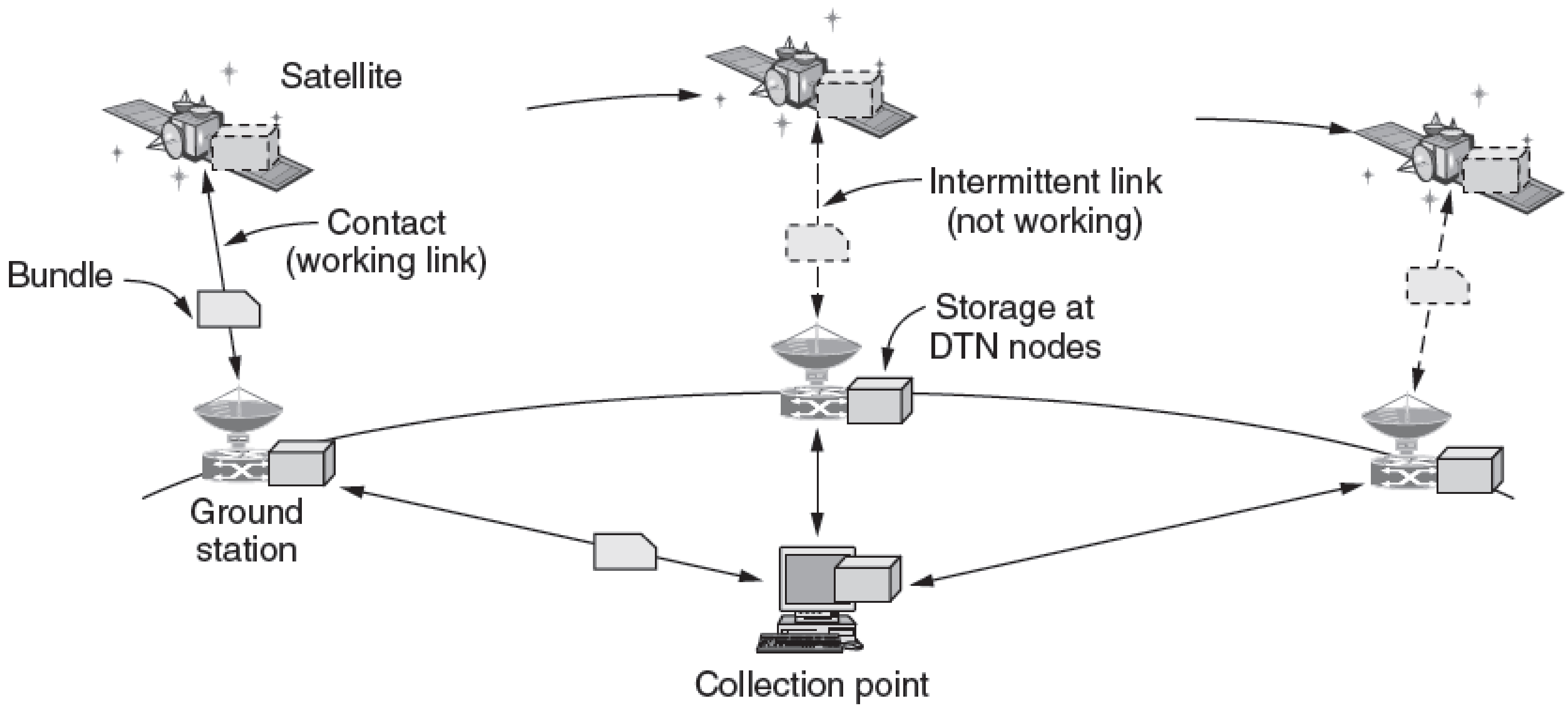
Messages called <u>bundles</u> are stored at DTN nodes while waiting for an intermittent link to become a contact

– Bundles might wait hours, not milliseconds in routers



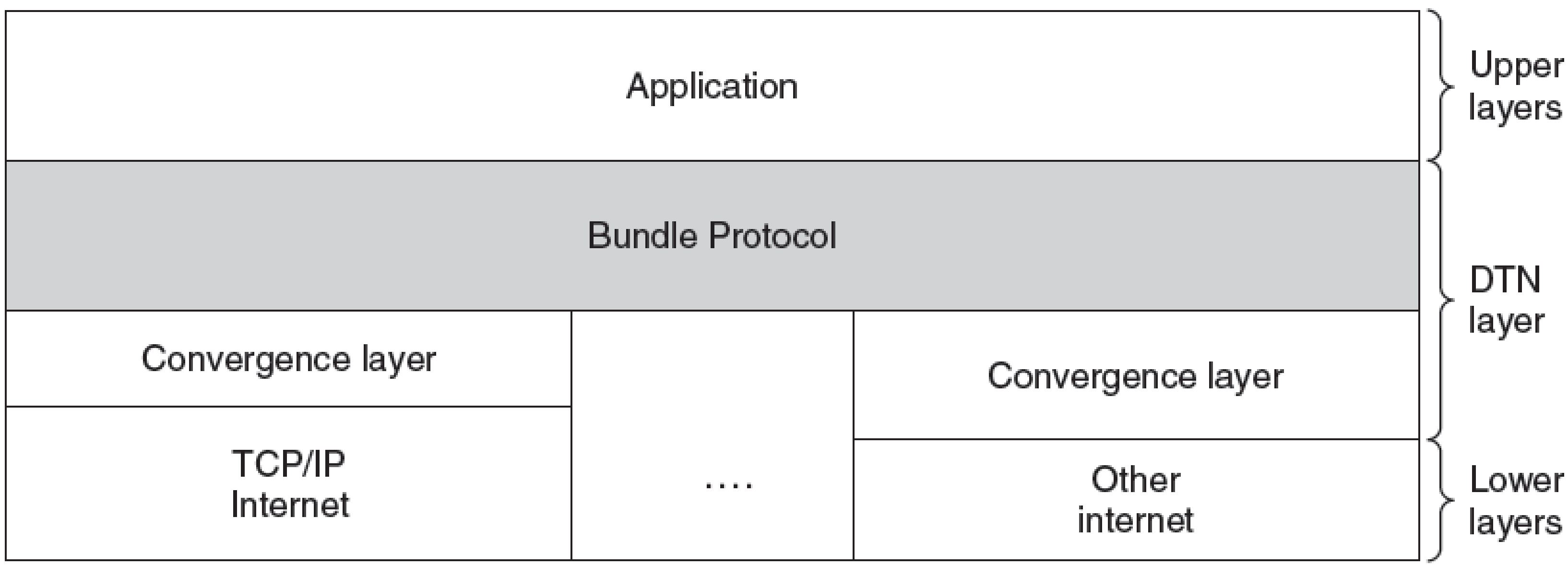
DTN Architecture (2)

Example DTN connecting a satellite to a collection point



Bundle Protocol (1)

The Bundle protocol uses TCP or other transports and provides a DTN service to applications



Bundle Protocol (2)

Features of the bundle message format:

- Dest./source add high-level addresses (not port/IP)
- Custody transfer shifts delivery responsibility
- Dictionary provides compression for efficiency