# Transport Layer

## Chapter 6

- Transport Service
- Elements of Transport Protocols
- Congestion Control
- Internet Protocols – UDP
- Internet Protocols – TCP
- Performance Issues
- Delay-Tolerant Networking



Revised: August 2011

# The Transport Layer

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

| |
|---|
| Application |
| **Transport** |
| Network |
| Link |
| Physical |

# Transport Service

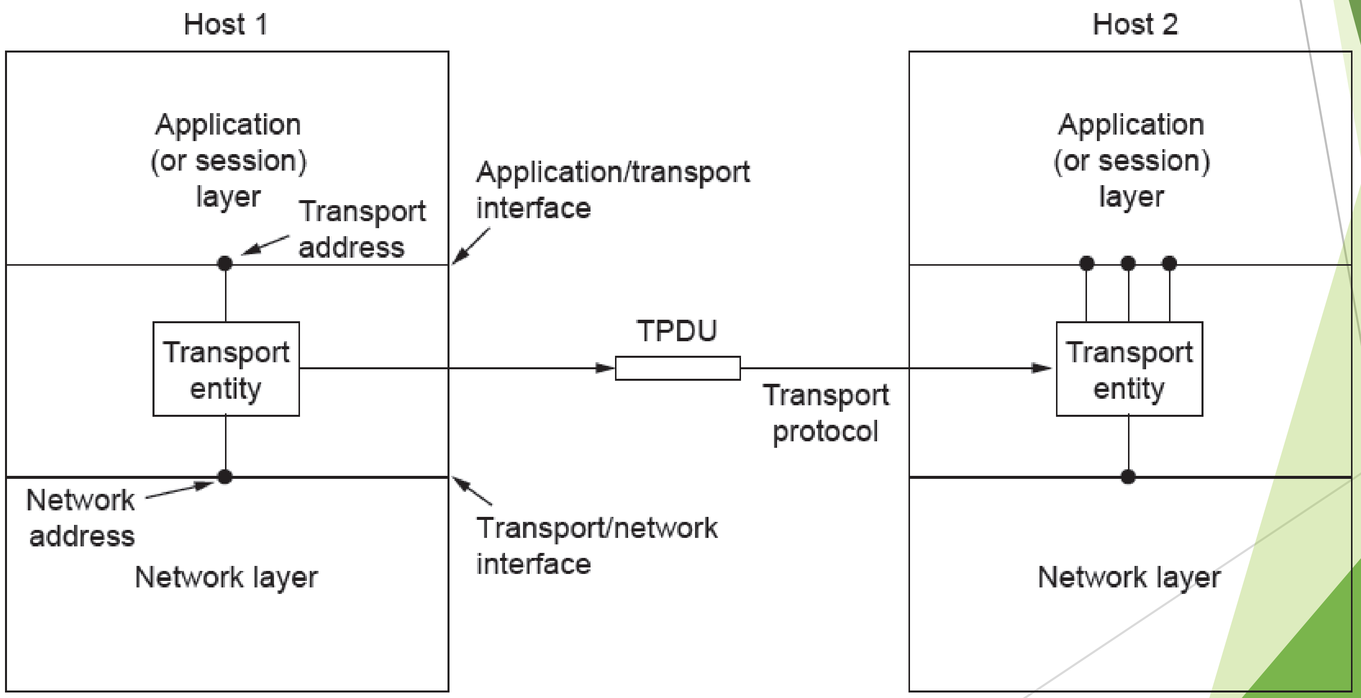
- Services Provided to the Upper Layer »
- Transport Service Primitives »
- Berkeley Sockets »
- Socket Example: Internet File Server »

Transport entity: software o hardware che implementa il transport layer, solitamente all’interno del kernel del sistema operativo.Cosí come nel network layer, es

# Services Provided to the Upper Layers (1)

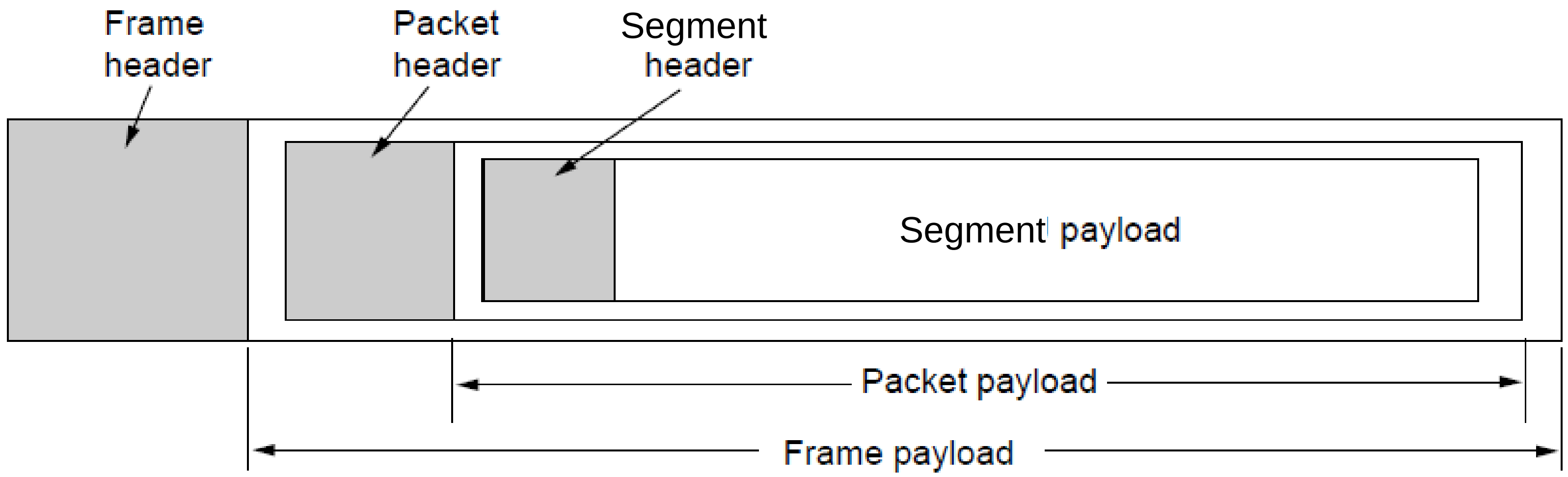
Transport layer adds reliability to the network layer

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



# Services Provided to the Upper Layers (2)

Transport layer sends segments in packets (in frames)



Le reti reali, sono prone ad errori ed é compito del transport layer di renderle piú affidabili.I messaggi inoltrati da transport entity a transport entity sono chiamati segmetoi (se

# Transport Service Primitives (1)

Primitives that applications might call to transport data for a simple connection-oriented service:

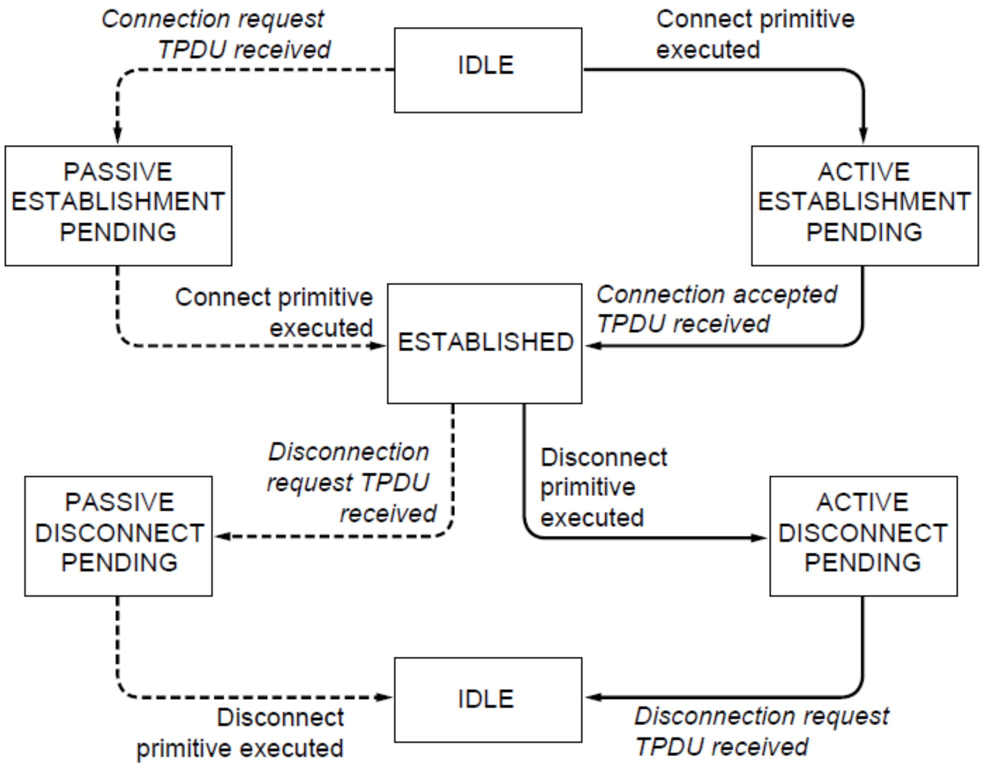
- Client calls CONNECT, SEND, RECEIVE, DISCONNECT
- Server calls LISTEN, RECEIVE, SEND, DISCONNECT

Le primitive minime ed essenziali sono: LISTEN, CONNECT, SEND, RECIVE e DISCONNECT. Tutti i protocolli di transport layer hanno queste primitieve in una forma o nell

| Primitive | Segment sent | Meaning |
|---|---|---|
| LISTEN | (none) | Block until some process tries to connect |
| CONNECT | CONNECTION REQ. | Actively attempt to establish a connection |
| SEND | DATA | Send information |
| RECEIVE | (none) | Block until a DATA packet arrives |
| DISCONNECT | DISCONNECTION REQ. | This side wants to release the connection |

# Transport Service Primitives (2)

State diagram for a simple connection-oriented service




Solid lines (right) show client state sequence

Dashed lines (left) show server state sequence

Transitions in italics are due to segment arrivals.

# 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

Sia il server side che il client side, le primitive sono eseguite in quest'ordine.

| | Primitive | Meaning |
|---|---|---|
| Server side | 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 |
| Client Side | 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 |

# Socket Example – Internet File Server (1)

Client code

```
. . .

if (argc != 3) fatal("Usage: client server-name file-name");
h = gethostbyname(argv[1]);
if (!h) fatal("gethostbyname failed");

s = socket(PF_INET, SOCK_STREAM, IPPROTO_TCP);
if (s <0) fatal("socket");
memset(&channel, 0, sizeof(channel));
channel.sin_family= AF_INET;
memcpy(&channel.sin_addr.s_addr, h->h_addr, h->h_length);
channel.sin_port= htons(SERVER_PORT);

c = connect(s, (struct sockaddr *) &channel, sizeof(channel));
if (c < 0) fatal("connect failed");

. . .
```

Get server's IP address

Make a socket

Try to connect

# Socket Example – Internet File Server (2)

Client code (cont.)

. . .

```
write(s, argv[2], strlen(argv[2])+1);

while (1) {
     bytes = read(s, buf, BUF_SIZE);
     if (bytes <= 0) exit(0);
     write(1, buf, bytes);
}
}
```

Write data (equivalent to send)

Loop reading (equivalent to receive) until no more data; exit implicitly calls close

# Socket Example – Internet File Server (3)

Server code

```
. . .

memset(&channel, 0, sizeof(channel));
channel.sin_family = AF_INET;
channel.sin_addr.s_addr = htonl(INADDR_ANY);
channel.sin_port = htons(SERVER_PORT);

s = socket(AF_INET, SOCK_STREAM, IPPROTO_TCP);
if (s < 0) fatal("socket failed");
setsockopt(s, SOL_SOCKET, SO_REUSEADDR, (char *) &on, sizeof(on));

b = bind(s, (struct sockaddr *) &channel, sizeof(channel));
if (b < 0) fatal("bind failed");

l = listen(s, QUEUE_SIZE);
if (l < 0) fatal("listen failed");

. . .
```

Make a socket

Assign address

Prepare for incoming connections

# Socket Example – Internet File Server (4)

Server code

. . .

```
while (1) {
      sa = accept(s, 0, 0);
      if (sa < 0) fatal("accept failed");

      read(sa, buf, BUF_SIZE);

      /* Get and return the file. */
      fd = open(buf, O_RDONLY);
      if (fd < 0) fatal("open failed");

      while (1) {
            bytes = read(fd, buf, BUF_SIZE);
            if (bytes <= 0) break;
            write(sa, buf, bytes);
      }
      close(fd);
      close(sa);
  }
}
```

Block waiting for the next connection

Read (receive) request and treat as file name

Write (send) all file data

Done, so close this connection

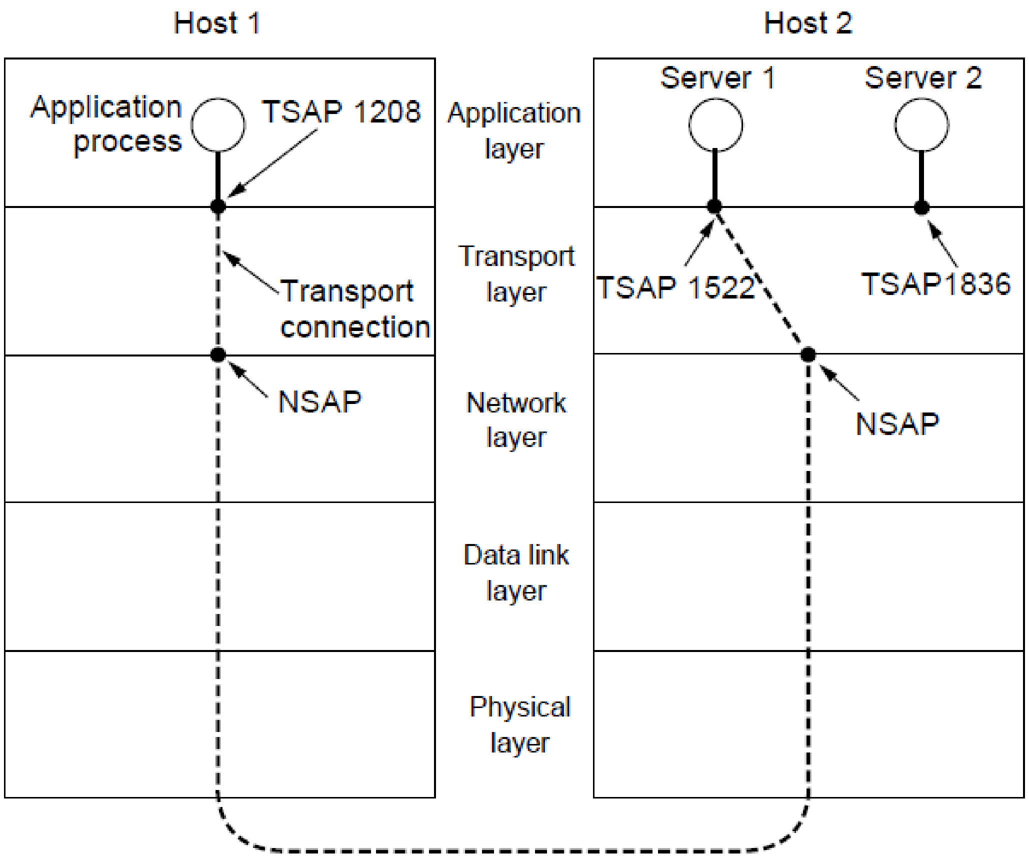
# Elements of Transport Protocols

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

# 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

Le porte sono indirizzi di transport layer ai quali i processi si mettor



# Connection Establishment (1)

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

- Don't treat an old or duplicate packet as new
- (Use ARQ and checksums for loss/corruption)

Approach:

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

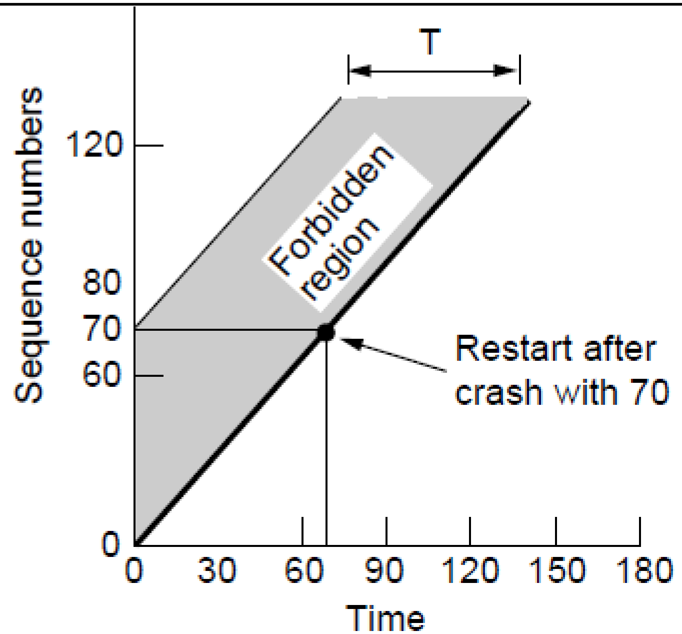
Un protocollo deve essere costruito per essere corretto in tutte le curcostanze; soltanto il caso comune deve essere implementato in modo efficiente per ottenere buone perform

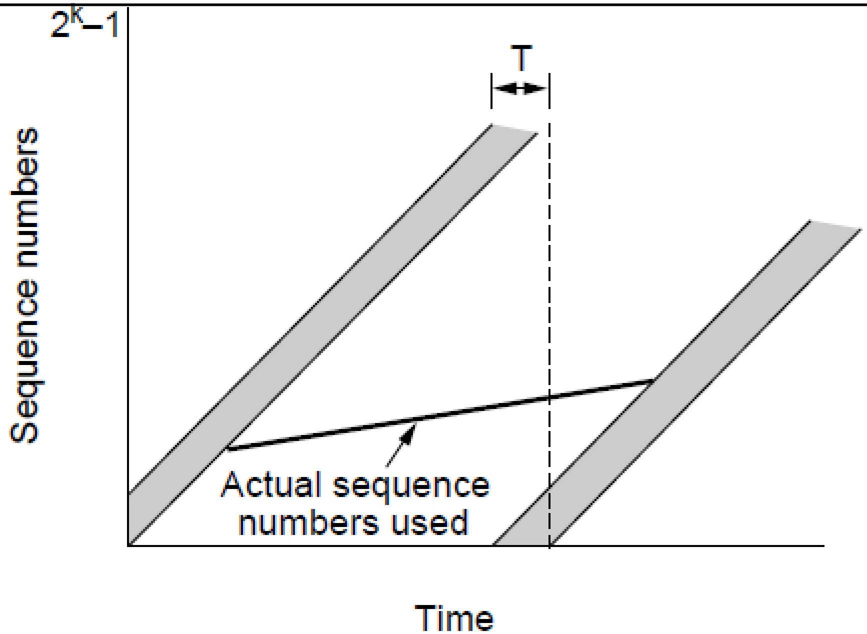
# Connection Establishment (2)

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

É stato proposto di utilizzare un orologio (il quale non deve essere sincoronizzato) da parte degli host. Ogni sequence number viene computato in base all'orologio e ogni nuo



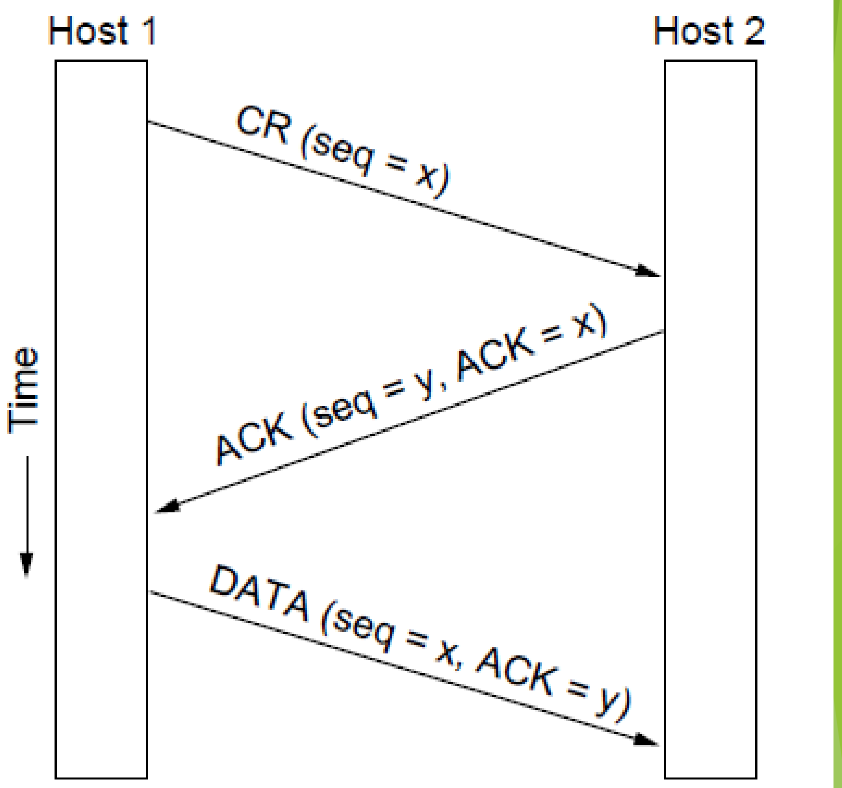

Need seq. number not to wrap within T seconds

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




Ogni qual volta un host riceve un segmento, chiede la conferma all'altro host. Se quest'ultim

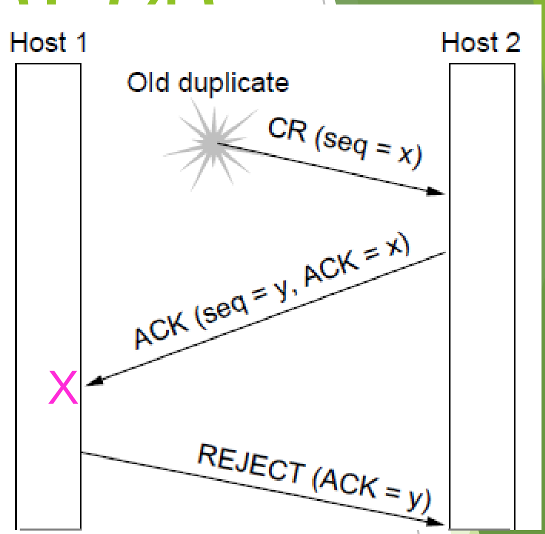
# Connection Establishment (4)

Three-way handshake protects against odd cases:

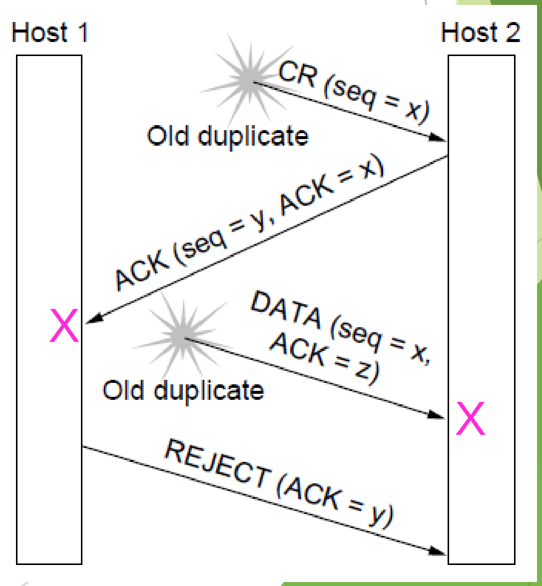
a) Duplicate CR. Spurious ACK does not connect

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

a)




b)

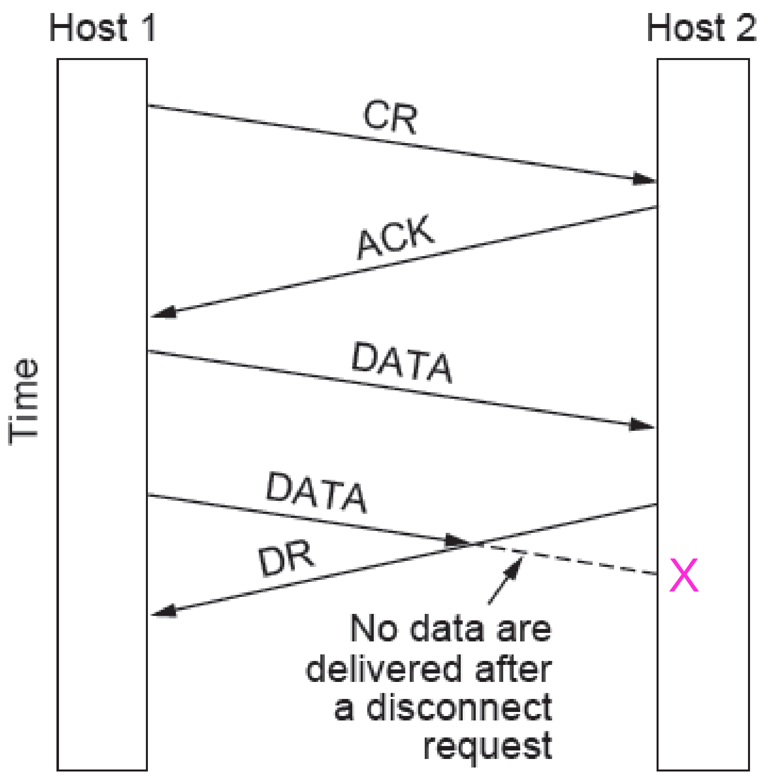


# Connection Release (1)

Key problem is to ensure reliability while releasing

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

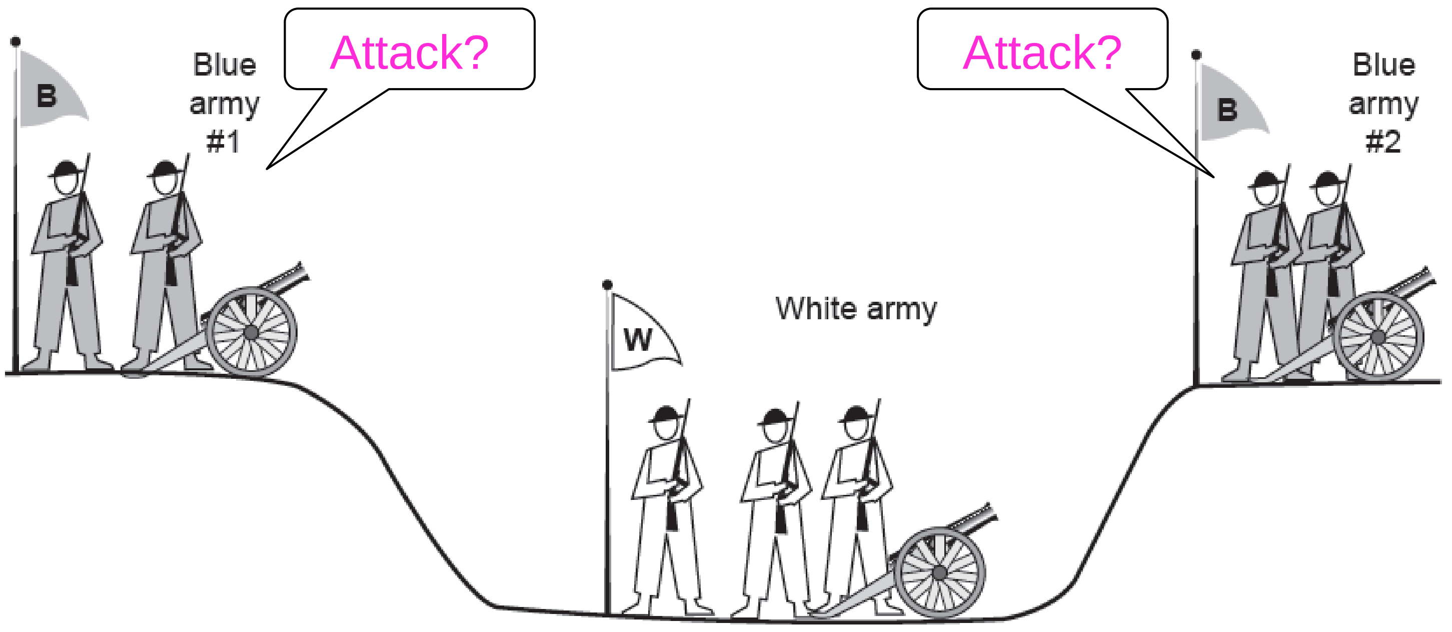
Asimmetrico: come nel sistema telefonico, quando uno dei due rilascia la conne



# 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

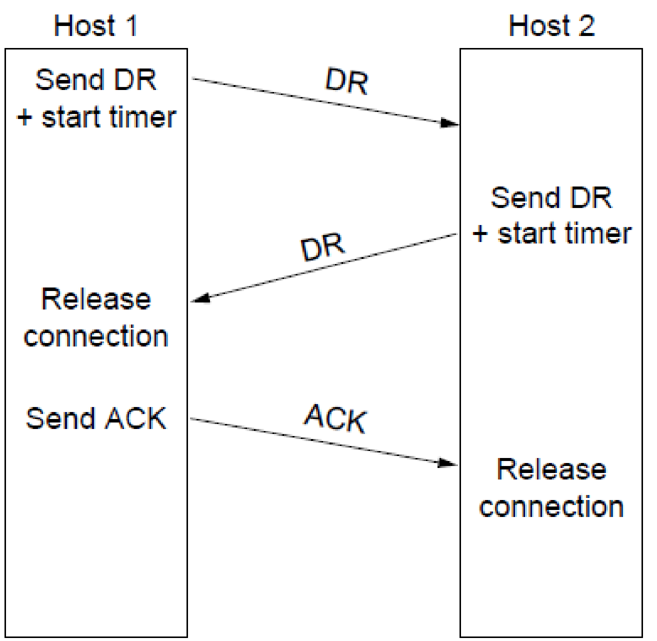


# Connection Release (3)

Normal release sequence, initiated by transport user on Host 1

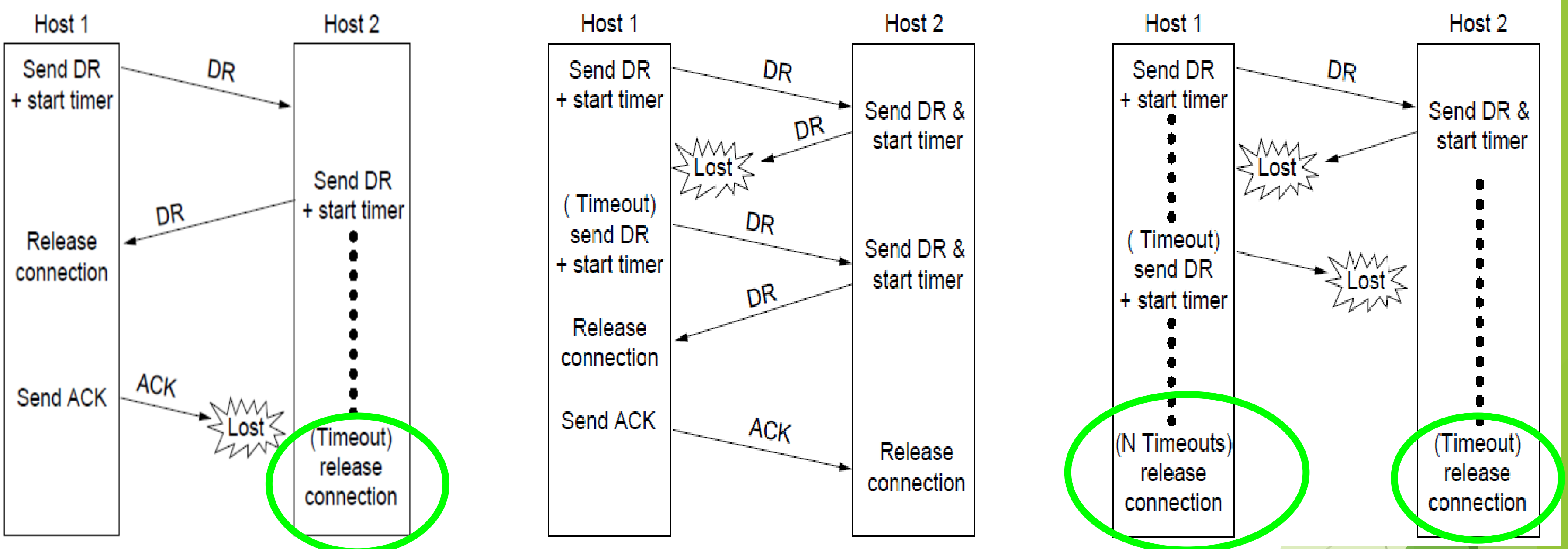
- DR=Disconnect Request
- Both DRs are ACKed by the other side

Le parti possono eseguono un massimo di N tentativi di trasmissione, e lo stesso vale in caso



# 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

# Error Control and Flow Control (1)

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

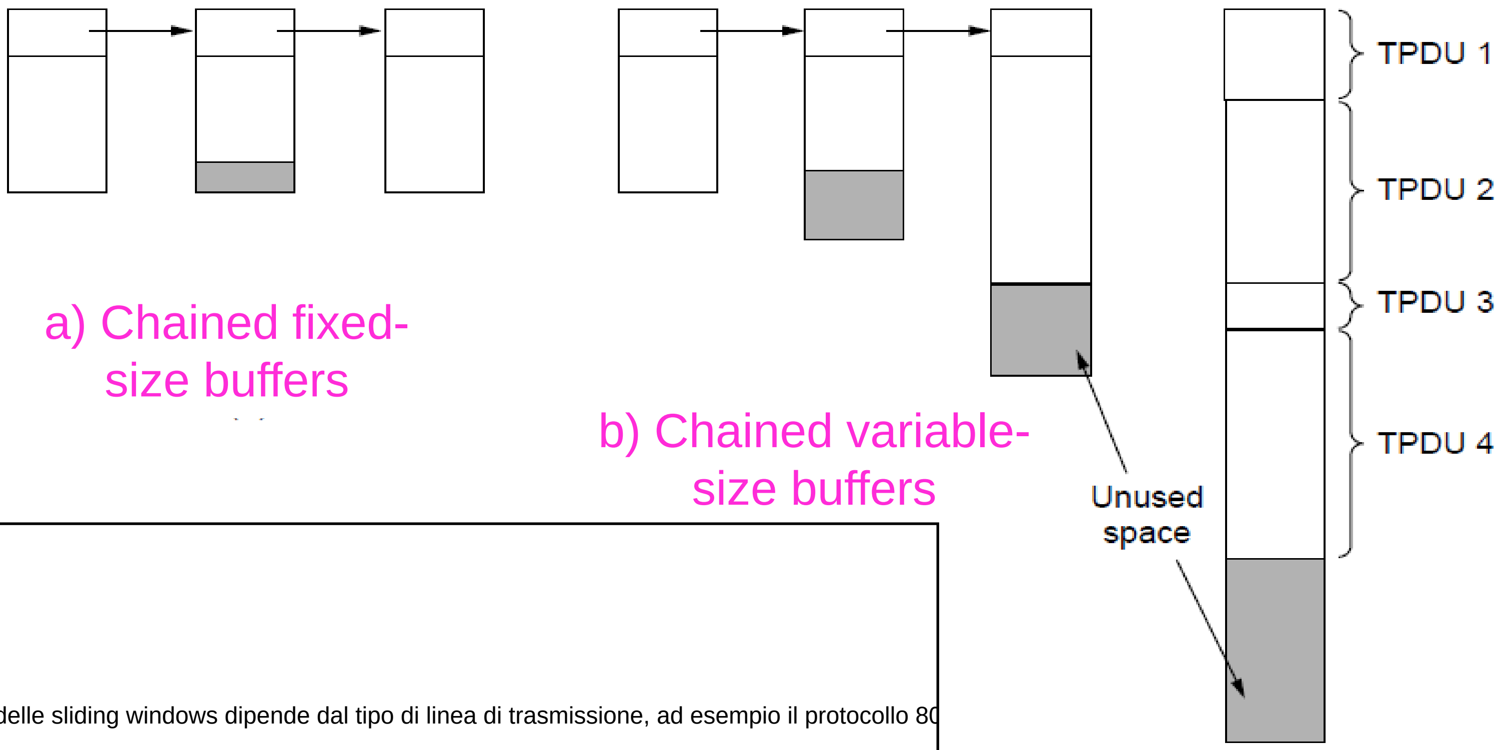
Flow control manages buffering at sender/receiver

- 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: si sddicura che i dari vengono consegnati con il desigderio di affidabilitá (solitamente si vuole senza errori).Flow Control: non saturare gli slow transmitter.Ripc

# 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

La grandezza delle sliding windows dipende dal tipo di linea di trasmissione, ad esempio il protocollo 80

# Error Control and Flow Control (3)

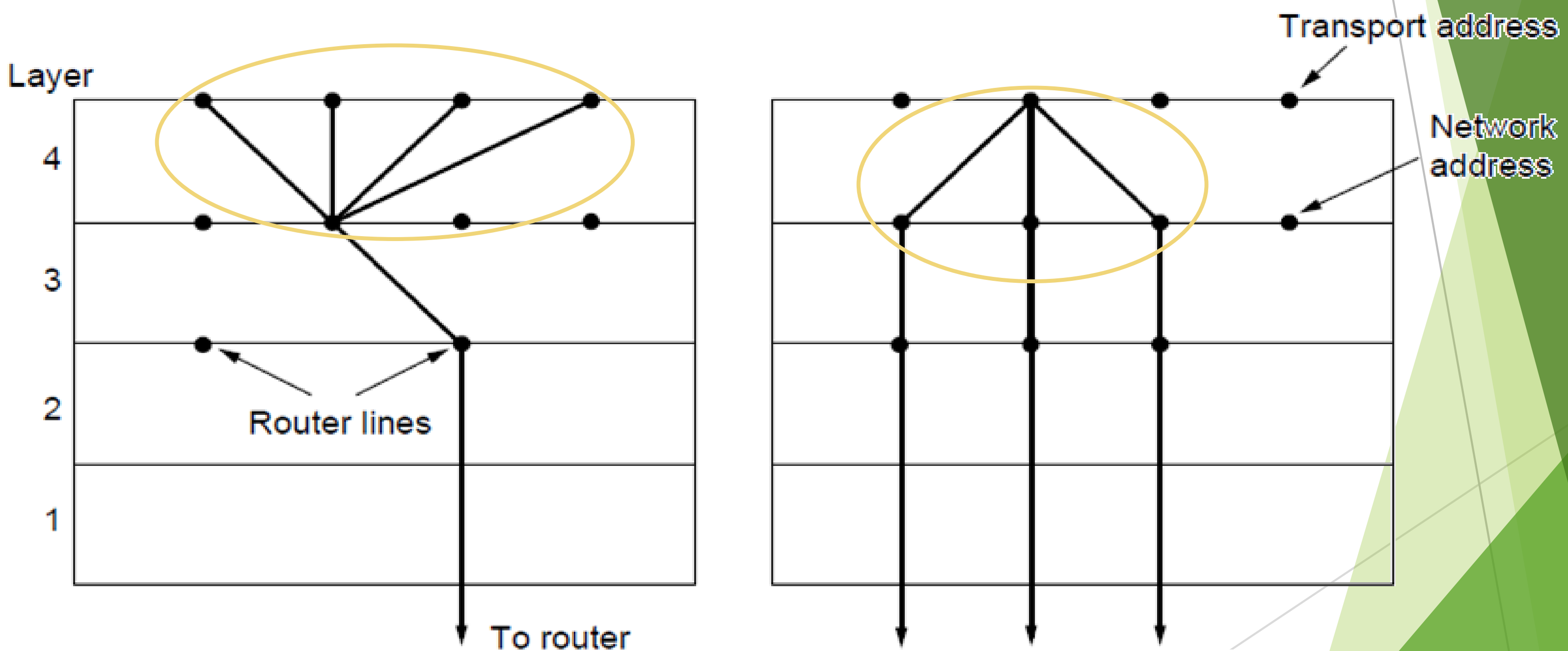
Per prevenire casi di deadlock, tutti i gli host dovrebbero mandare s

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

| | 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 |

# Multiplexing

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




Multiplexing

Inverse Multiplexing

# Crash Recovery

Quando un host crasha e viene ripristinato, non sa piú il punto a cui era rimasto. Gli host a cui era

Application needs to help recovering from a crash

- Transport can fail since A(ck) / W(rite) not atomic

A: AcknownedgmentW

| | Strategy used by receiving host | | | | | |
|---|---|---|---|---|---|---|
| | First ACK, then write | | | First write, then ACK | | |
| Strategy used by sending host | 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

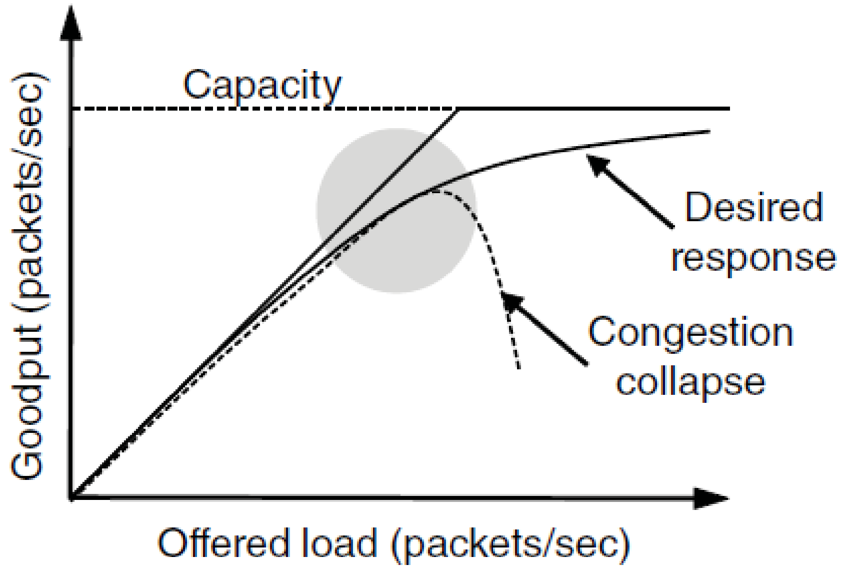
Two layers are responsible for congestion control:

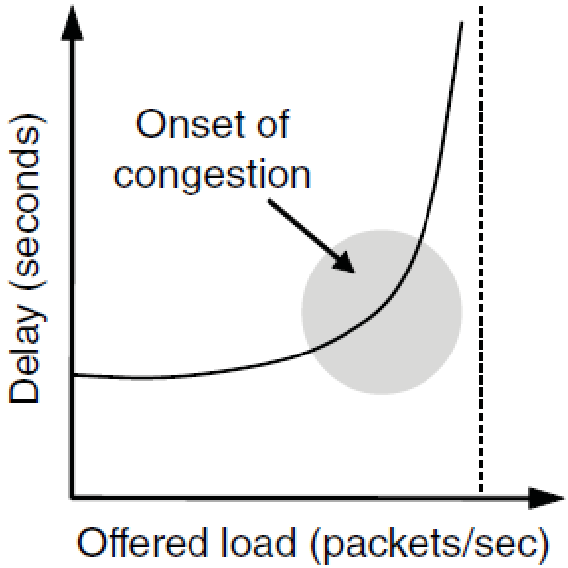
- Transport layer, controls the offered load [here]
- Network layer, experiences congestion [previous]

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

# Desirable Bandwidth Allocation (1)

Efficient use of bandwidth gives high goodput, low delay



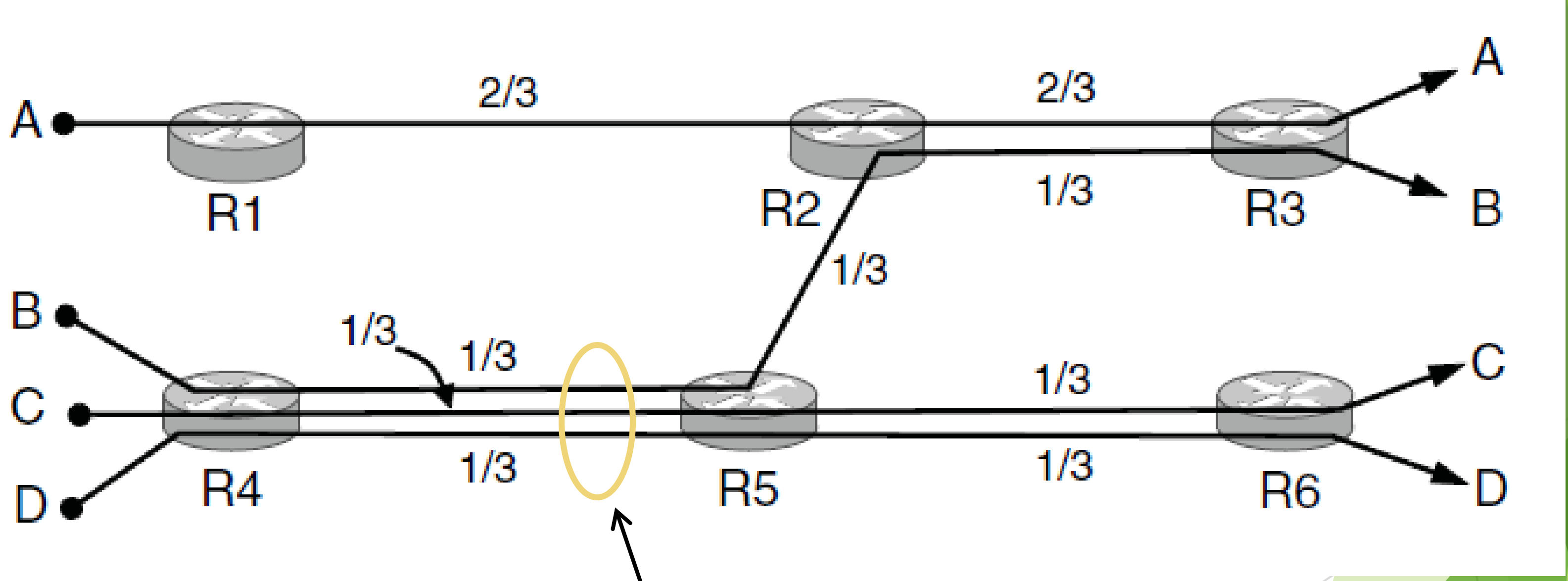

Goodput rises more slowly than load when congestion sets in

Delay begins to rise sharply when congestion sets in

# Desirable Bandwidth Allocation (2)

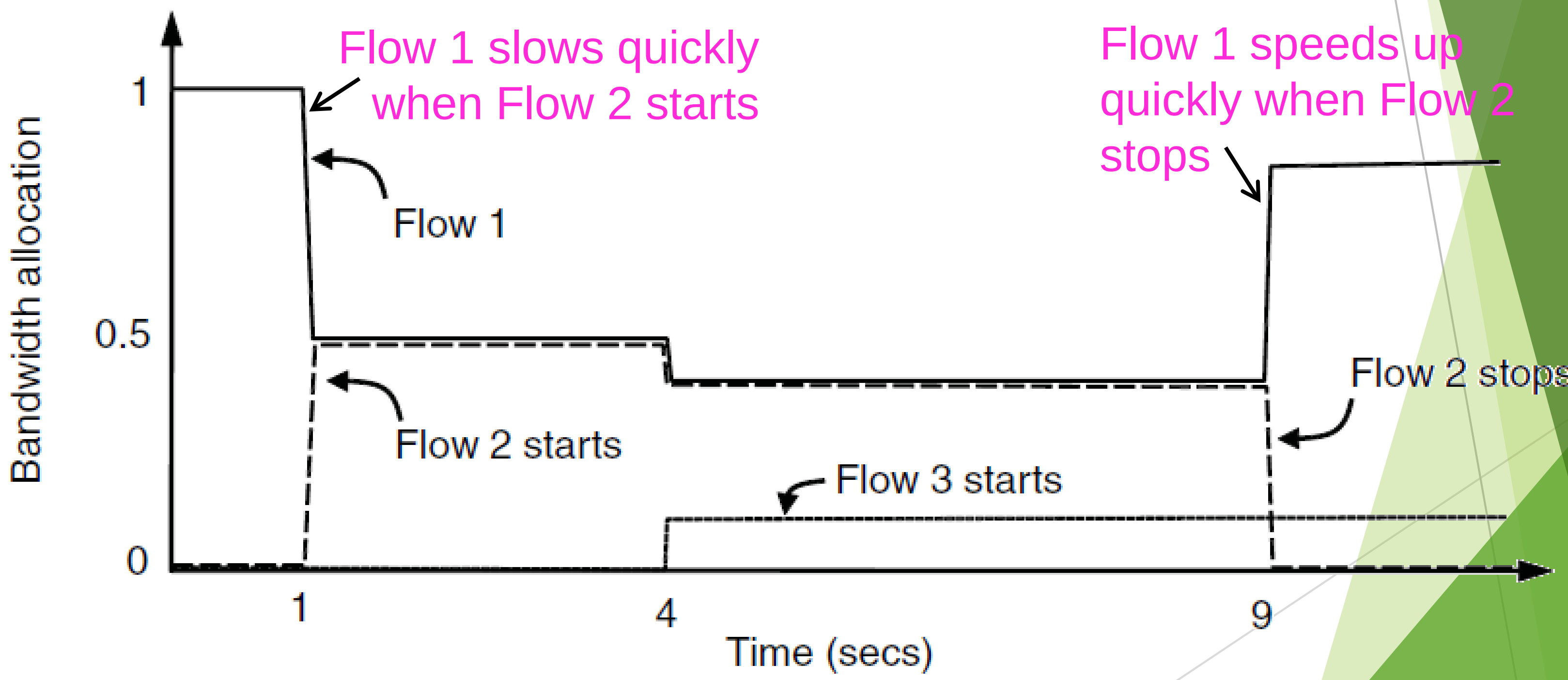
Fair use gives bandwidth to all flows (no starvation)

- Max-min fairness gives equal shares of bottleneck



# Desirable Bandwidth Allocation (3)

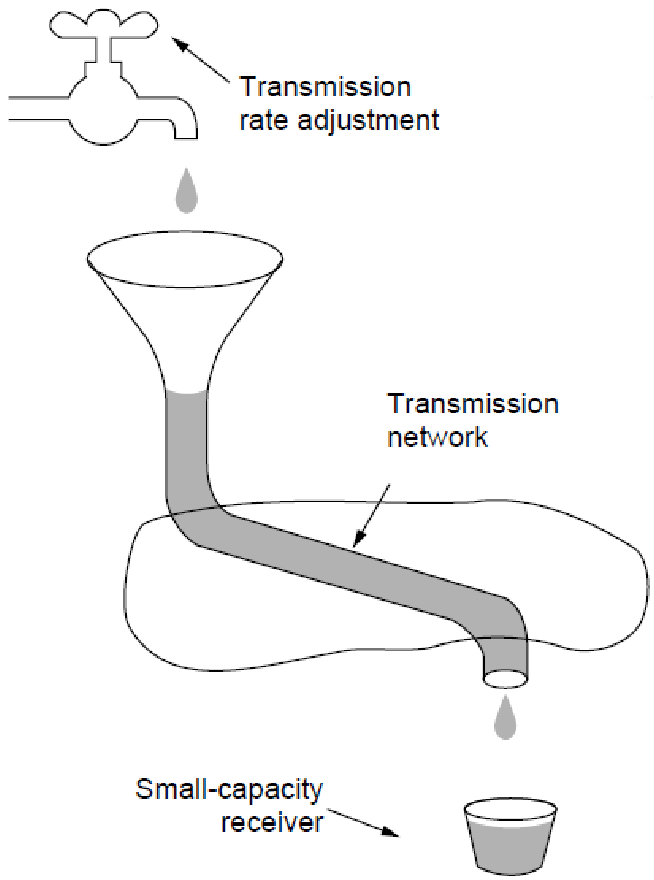
We want bandwidth levels to converge quickly when traffic patterns change



# Regulating the Sending Rate (1)

Sender may need to slow down for different reasons:

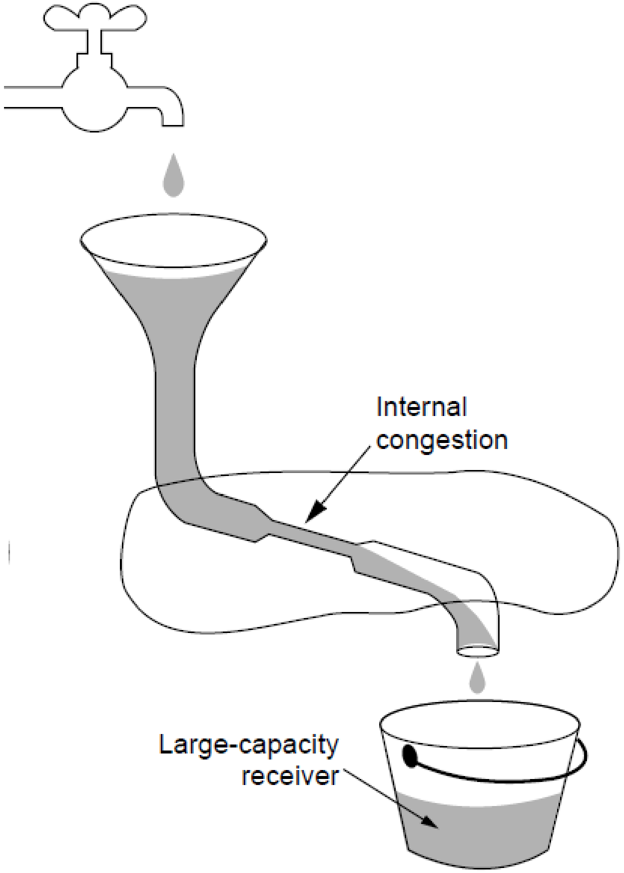
- Flow control, when the receiver is not fast enough [right]
- Congestion, when the network is not fast enough [over]




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

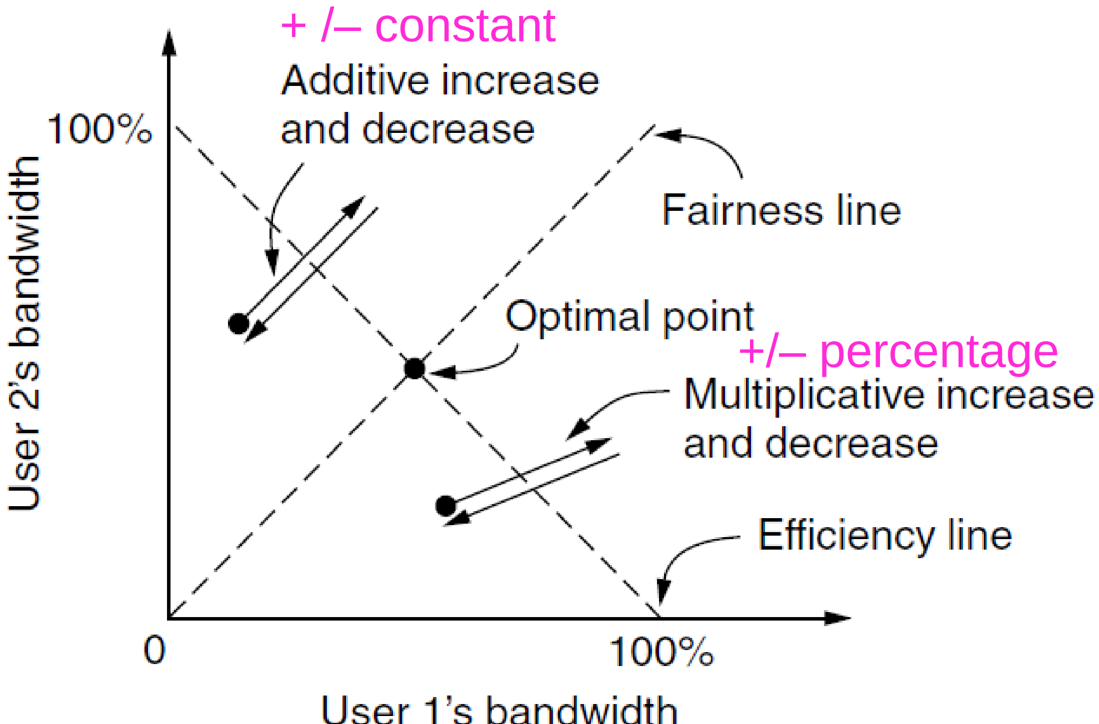
# Regulating the Sending Rate (3)

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)

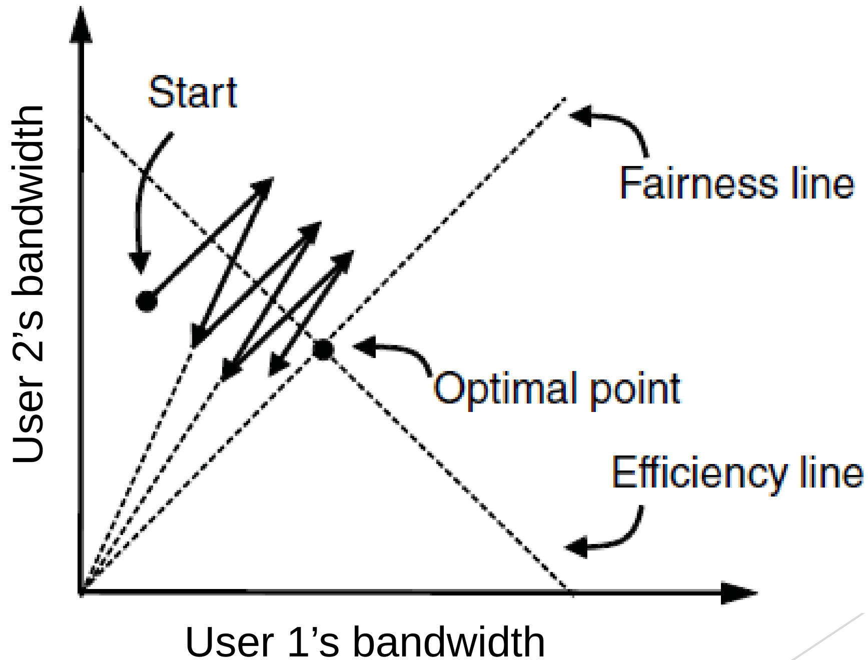
If two flows increase/decrease their bandwidth in the same way when the network signals free/busy they will not converge to a fair allocation



# Regulating the Sending Rate (4)

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

- TCP uses AIMD for this reason



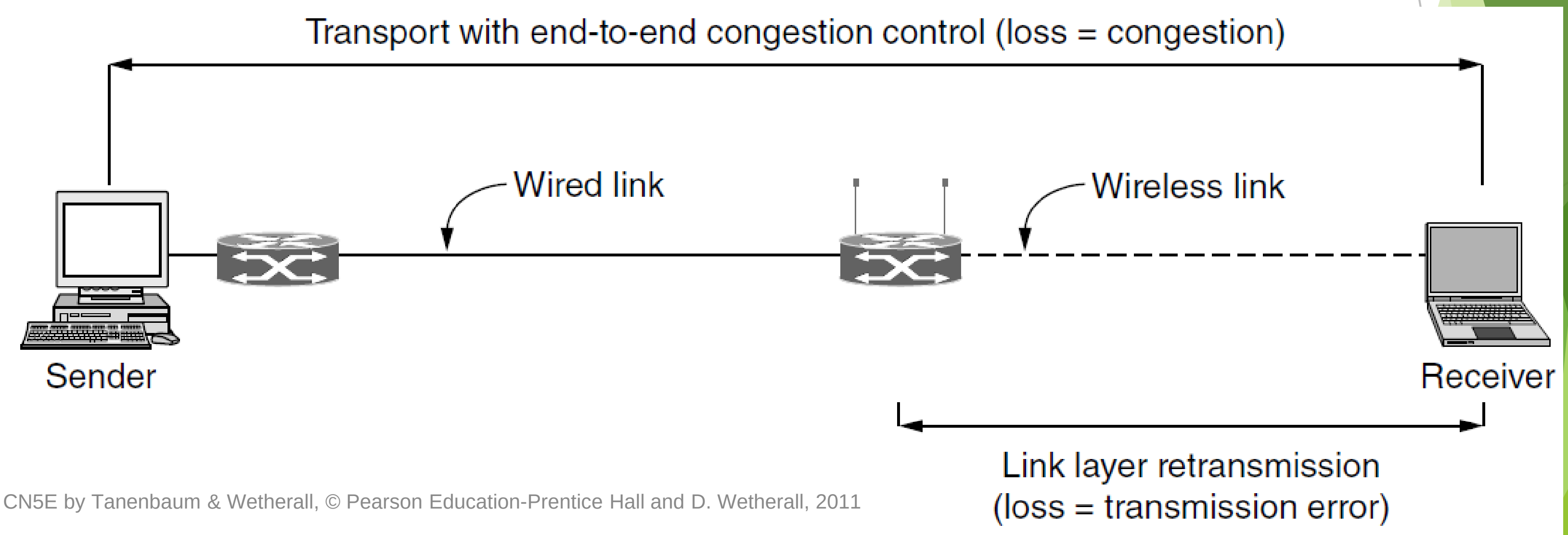
# Wireless Issues

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



# Internet Protocols – UDP

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

# Introduction to UDP (1)

UDP (User Datagram Protocol) is a shim over IP

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

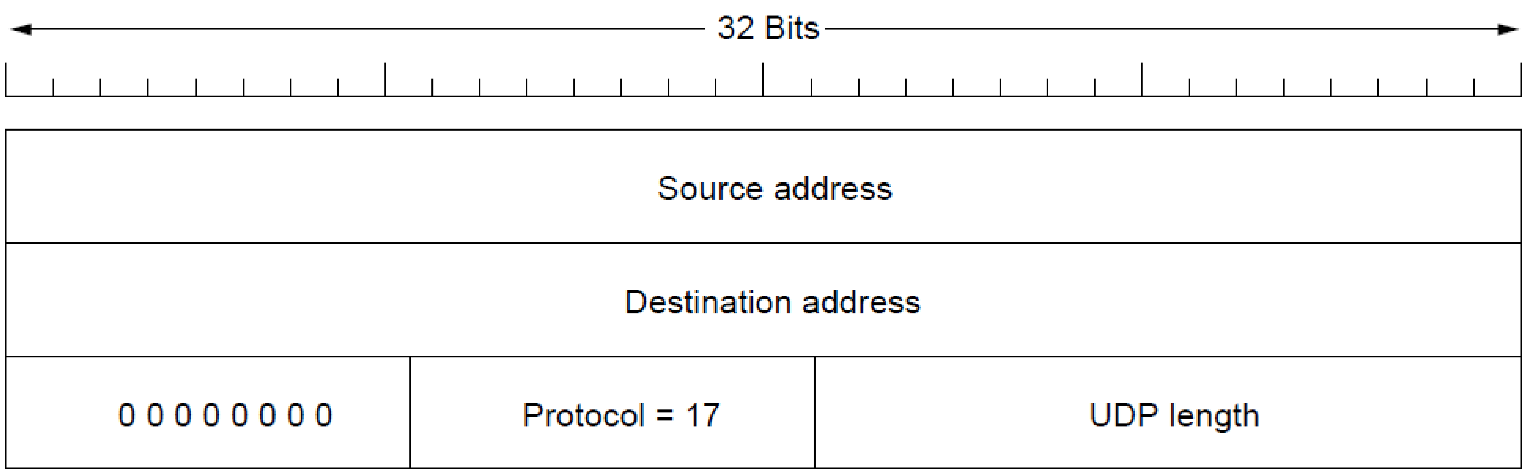
32 Bits

| | |
|---|---|
| Source port | Destination port |
| UDP length | UDP checksum |

# Introduction to UDP (2)

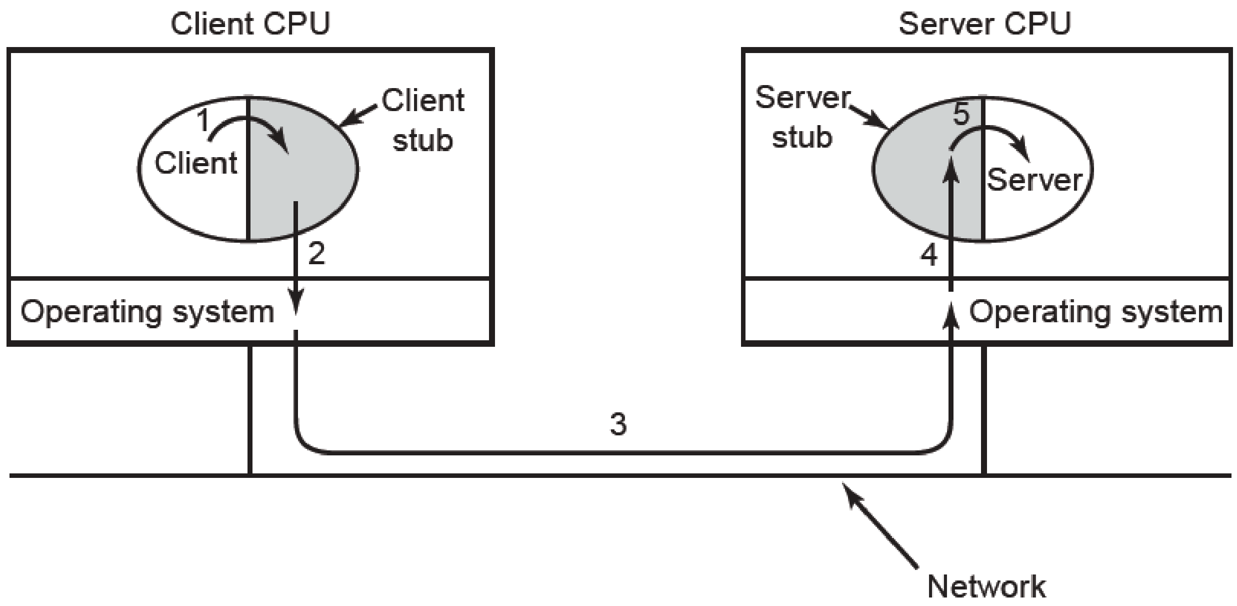
Checksum covers UDP segment and IP pseudoheader

- 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

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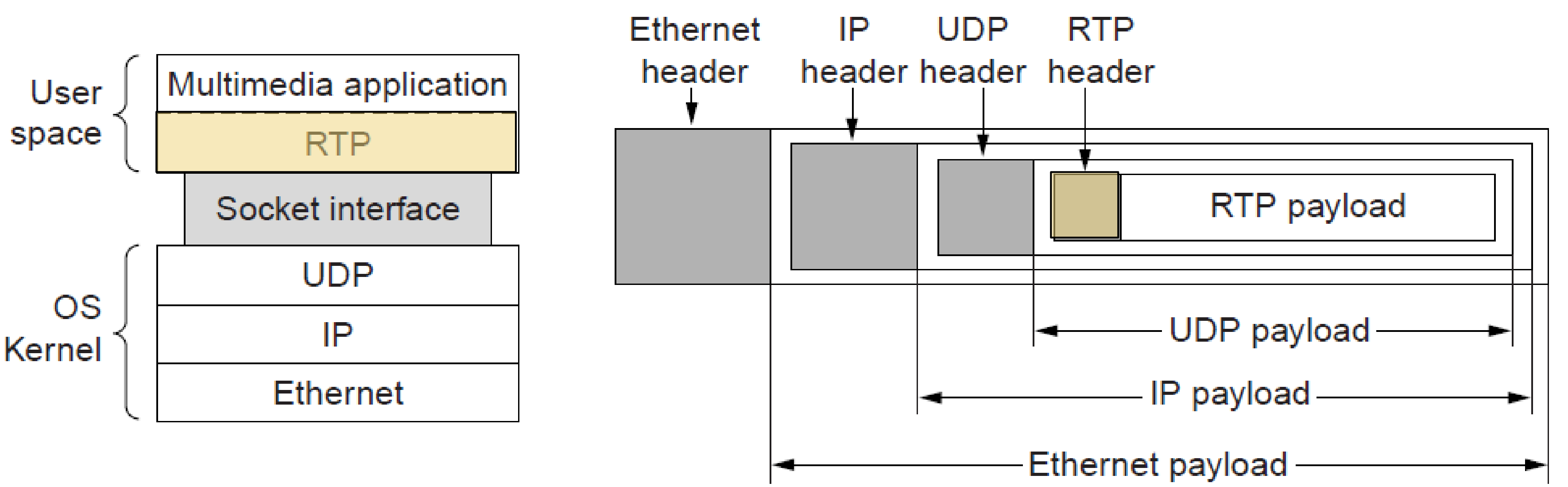
# Real-Time Transport (1)

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

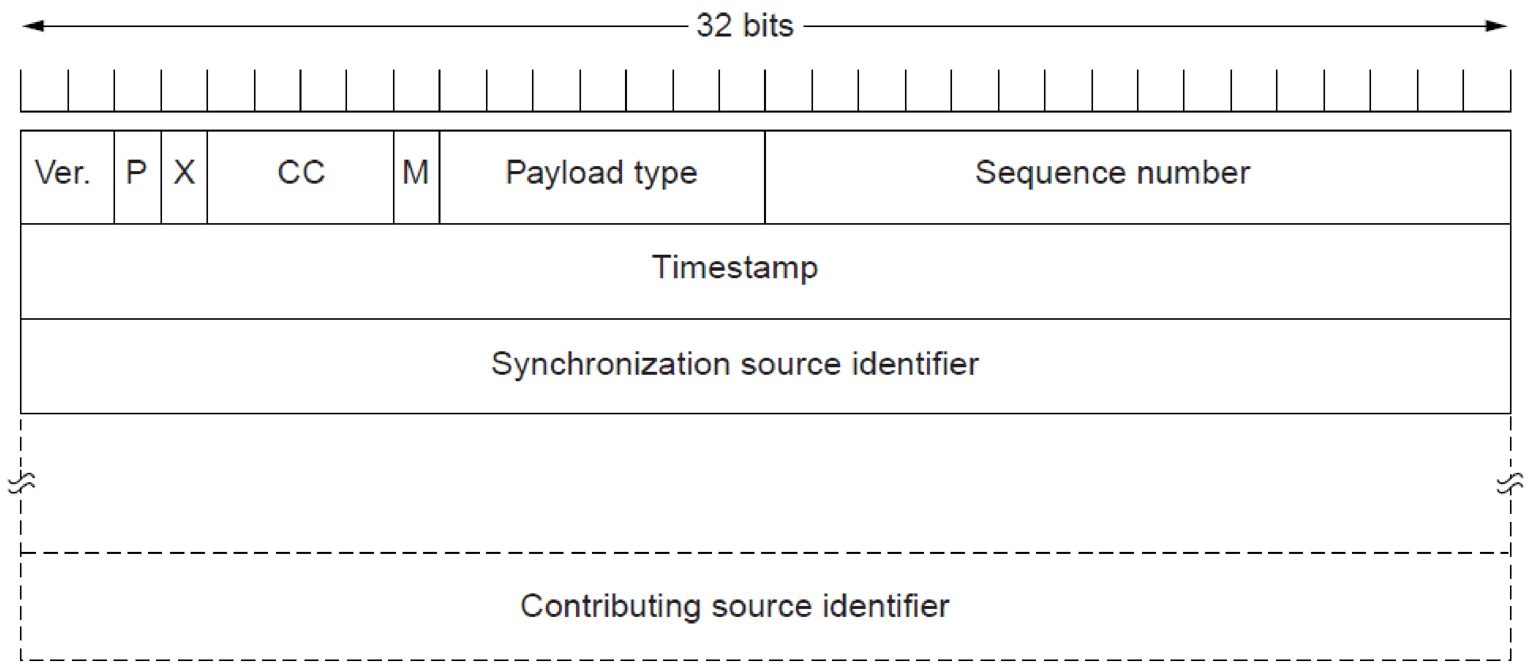
- Often implemented as part of the application



# Real-Time Transport (2)

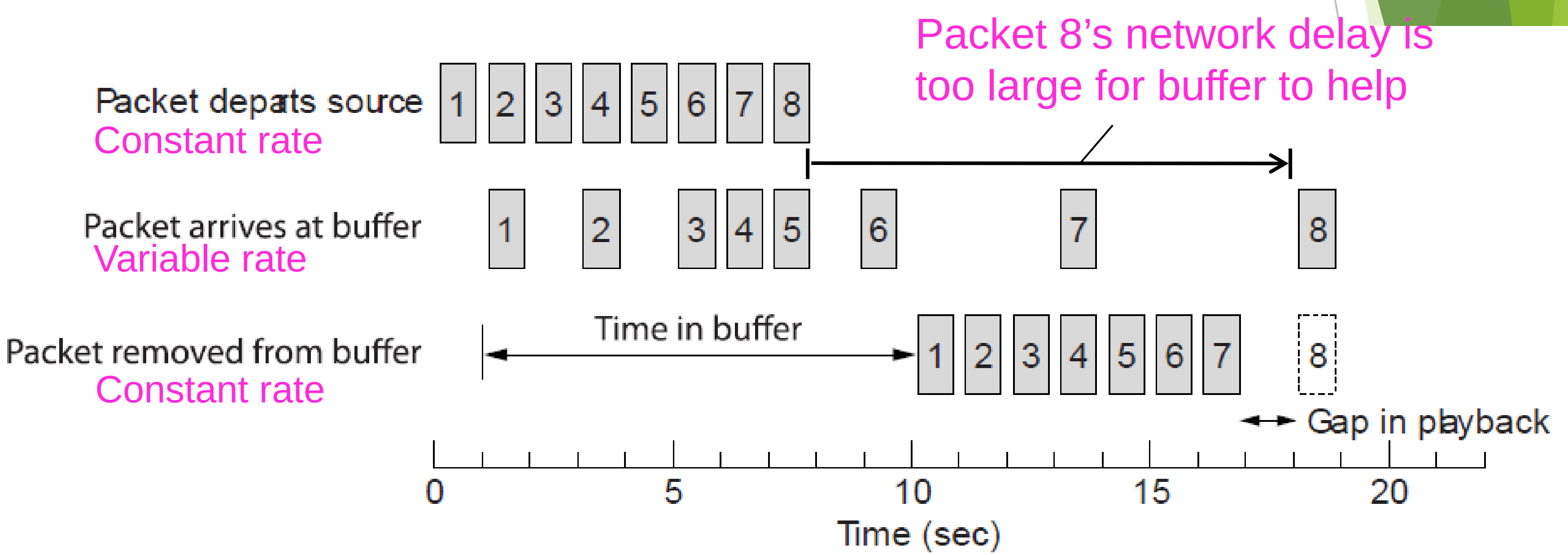
RTP header contains fields to describe the type of media and synchronize it across multiple streams

- RTCP sister protocol helps with management tasks



# Real-Time Transport (3)

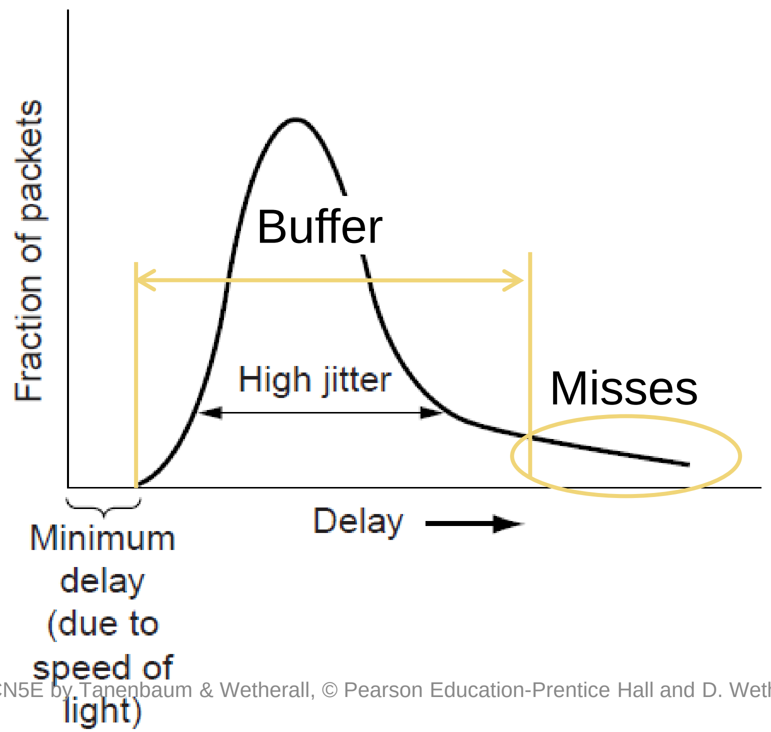
Buffer at receiver is used to delay packets and absorb jitter so that streaming media is played out smoothly

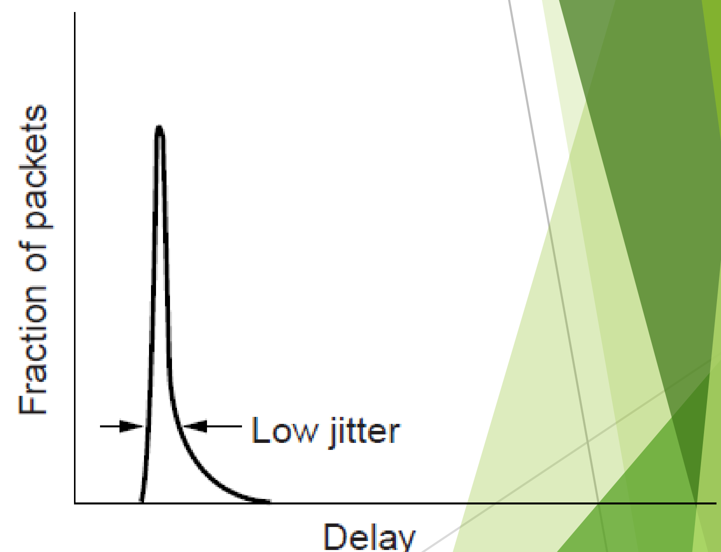


# Real-Time Transport (3)

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 »

# The TCP Service Model (1)

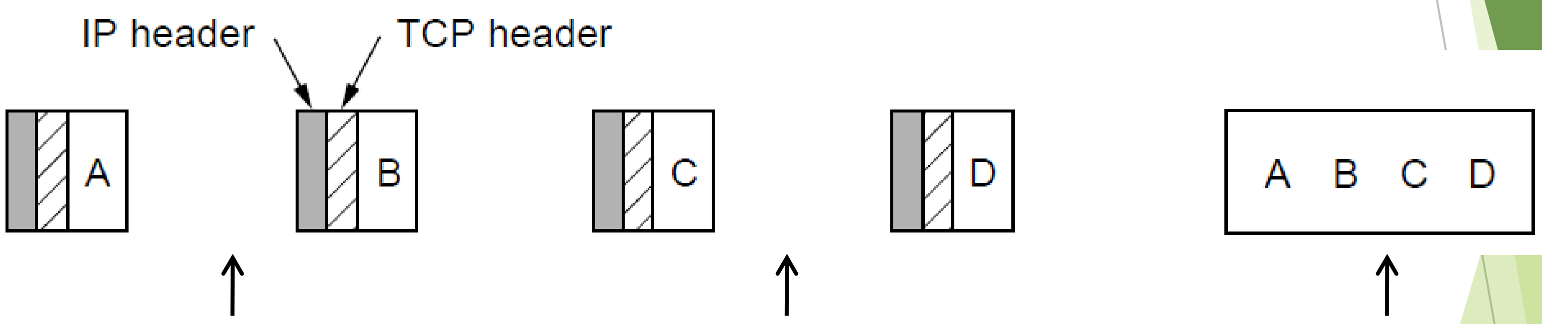
TCP provides applications with a reliable byte stream between processes; it is the workhorse of the Internet

- 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 |

# The TCP Service Model (2)

Applications using TCP see only the byte stream [right] and not the segments [left] sent as separate IP packets

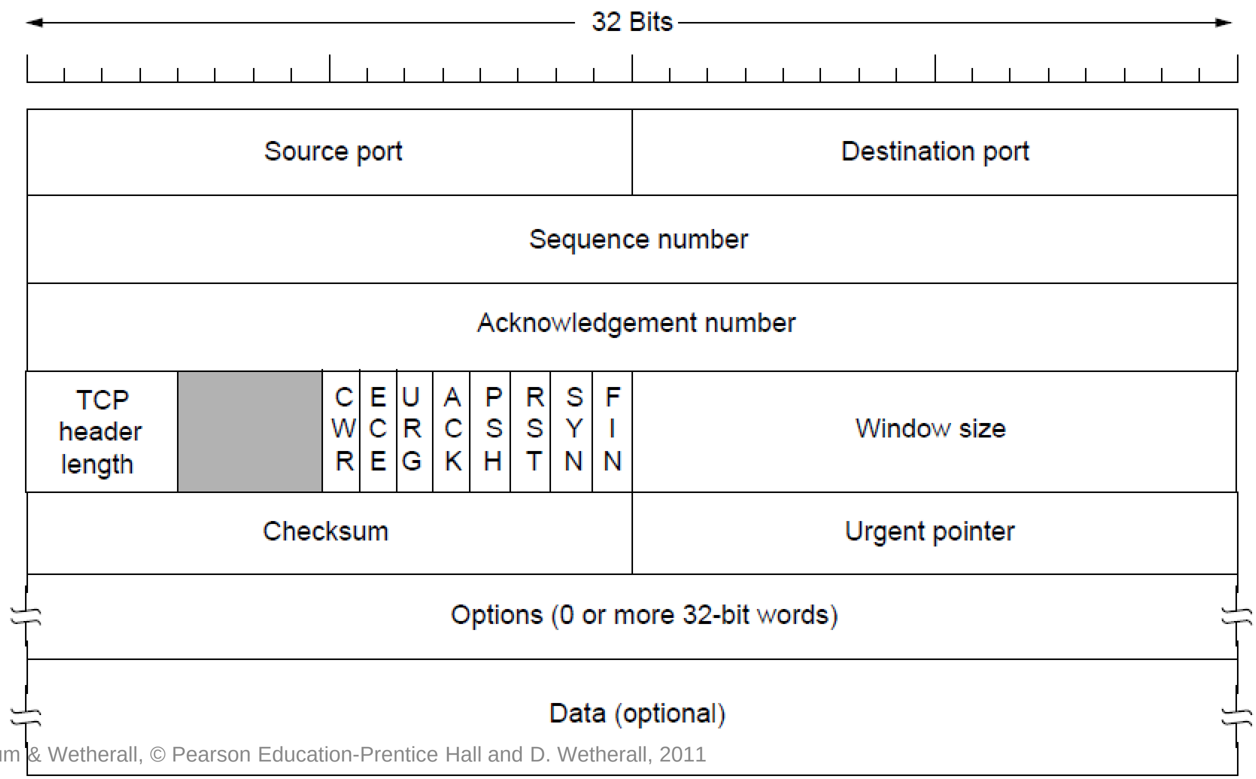



Four segments, each with 512 bytes of data and carried in an IP packet

2048 bytes of data delivered to application in a single READ call

# The TCP Segment Header

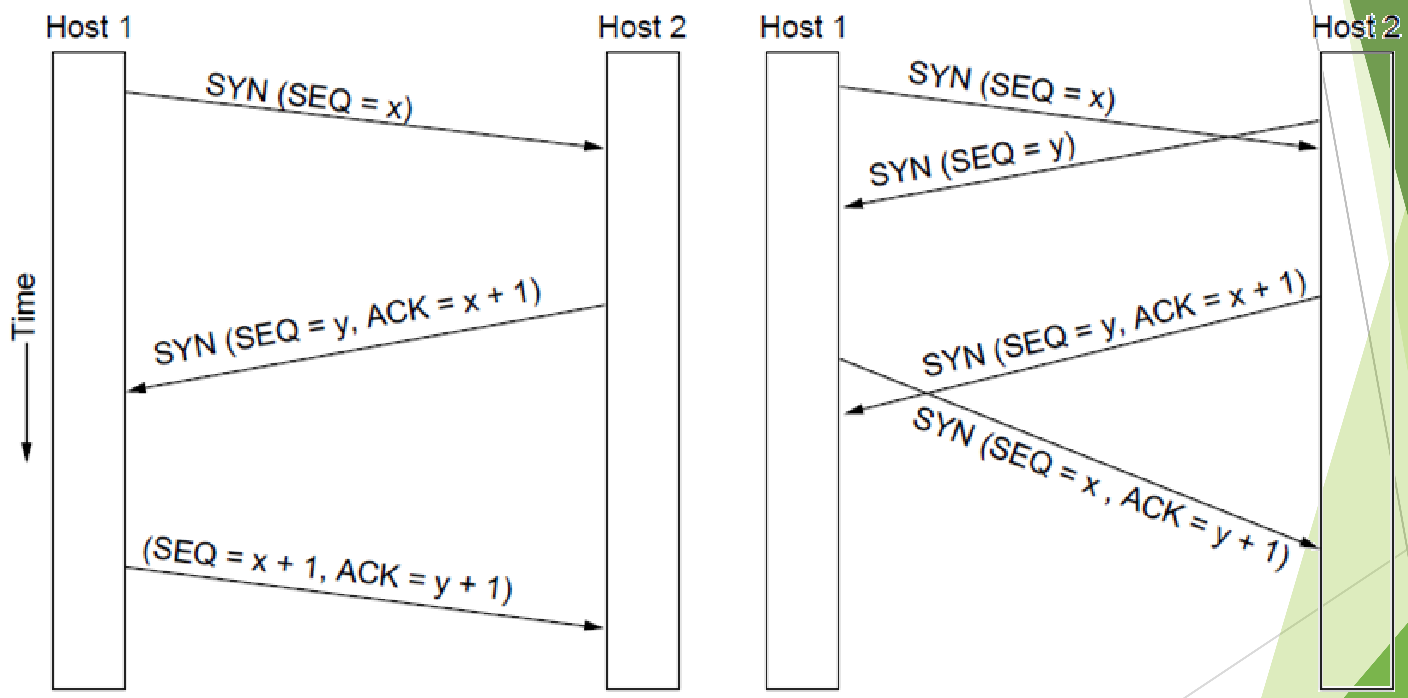
TCP header includes addressing (ports), sliding window (seq. / ack. number), flow control (window), error control (checksum) and more.



# 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 |

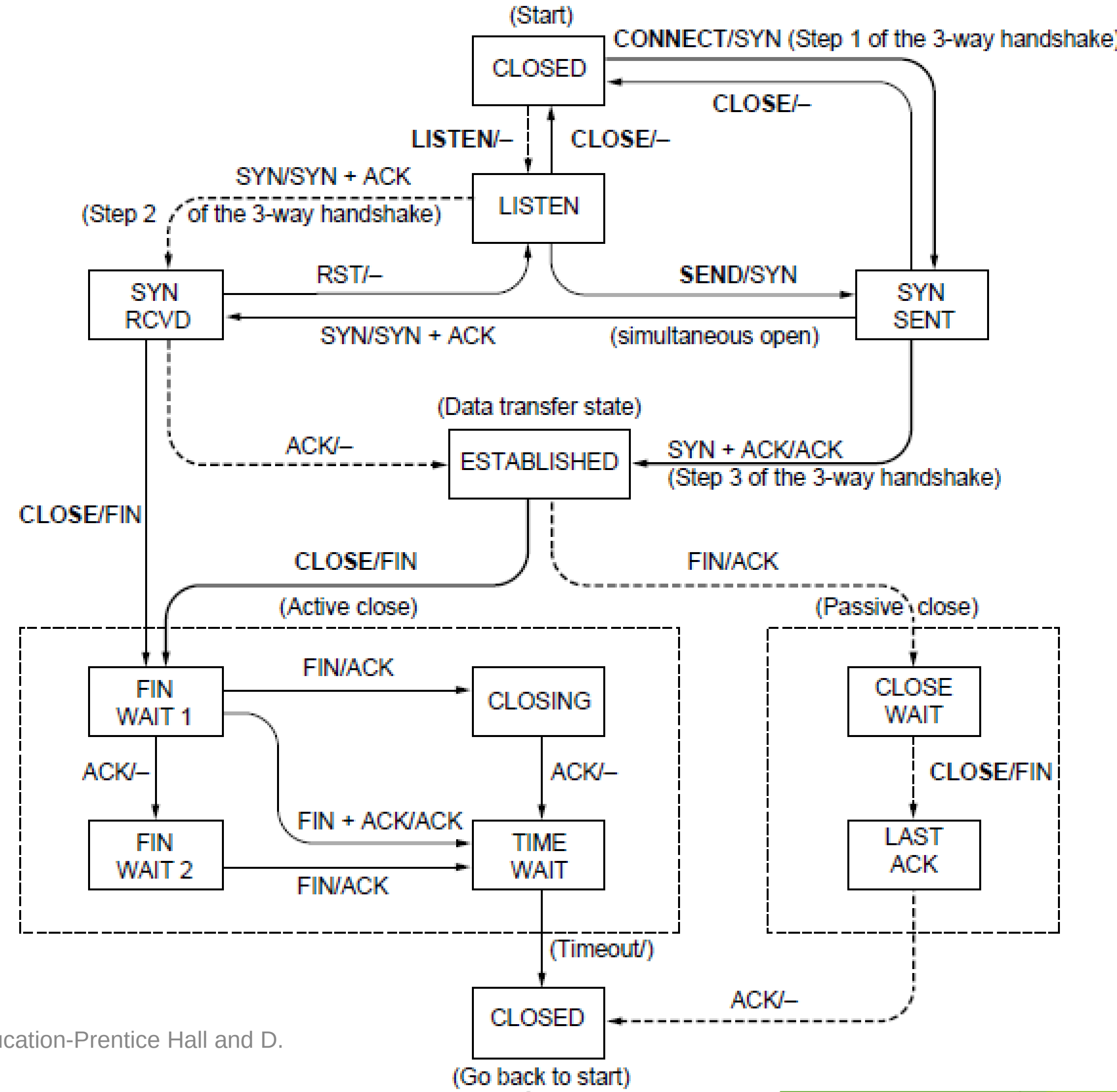
# TCP Connection State Modeling (2)

Solid line is the normal path for a client.

Dashed line is the normal path for a server.

Light lines are unusual events.

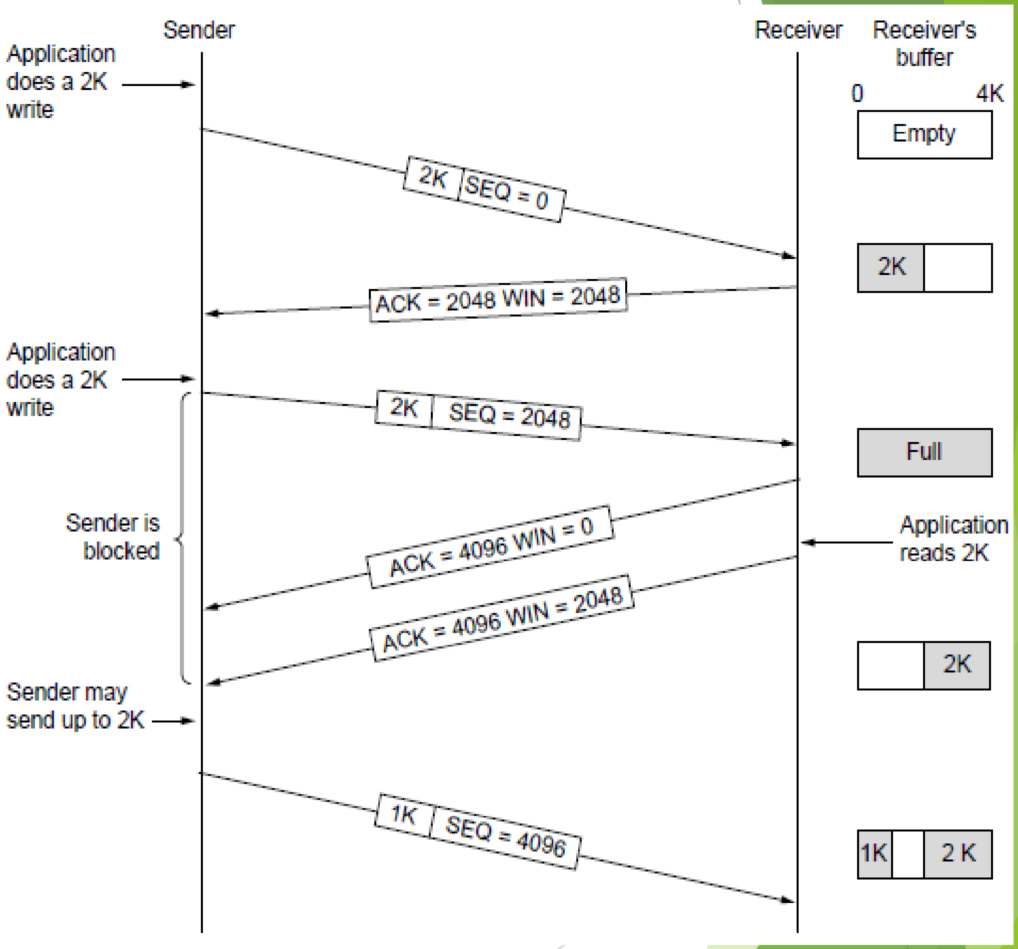
Transitions are labeled by the cause and action, separated by a slash.



# TCP Sliding Window (1)

TCP adds flow control to the sliding window as before

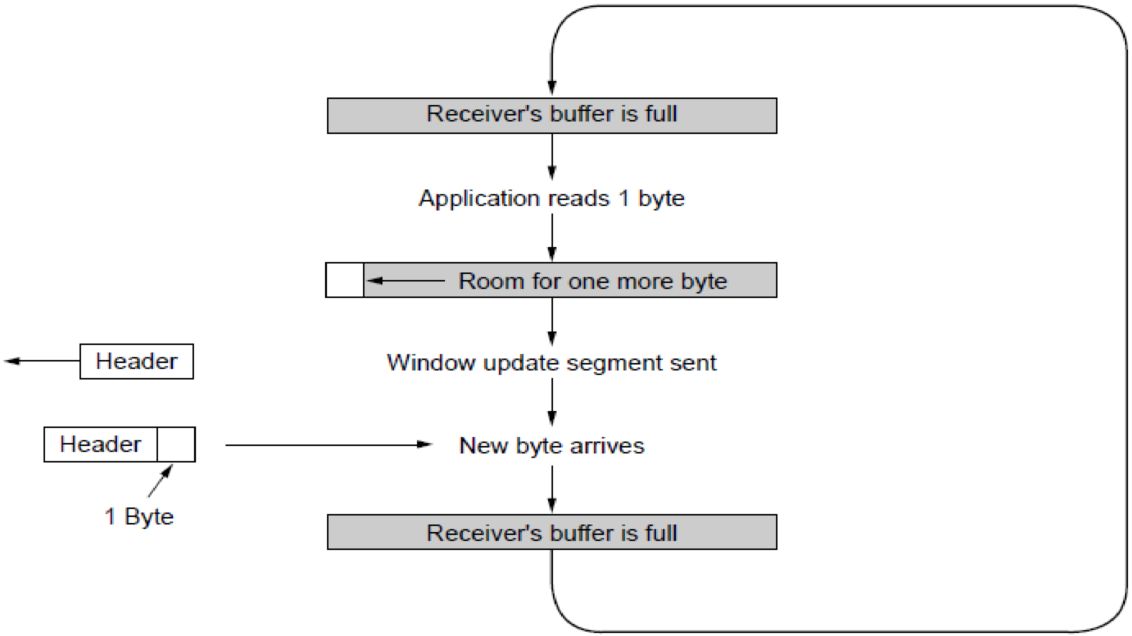
- ACK + WIN is the sender's limit



# TCP Sliding Window (2)

Need to add special cases to avoid unwanted behavior

- E.g., silly window syndrome [below]

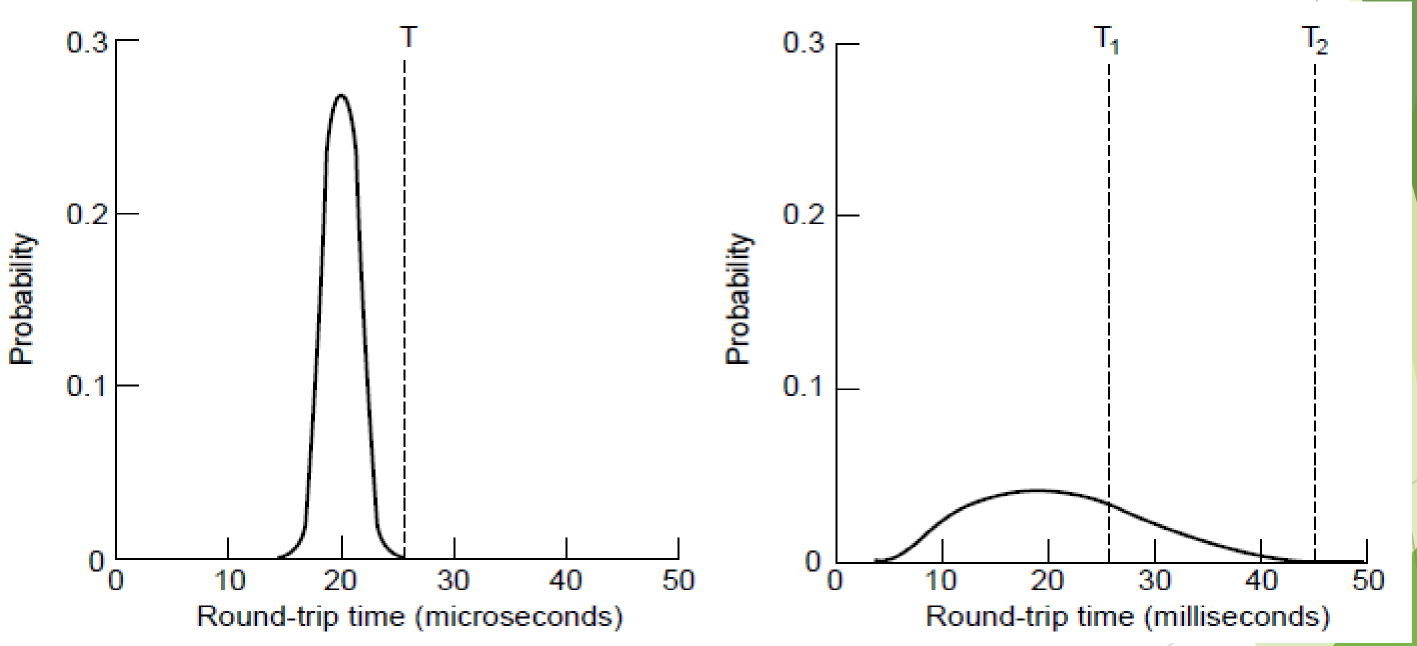



Receiver application reads single bytes, so sender always sends one byte segments

# 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

# TCP Congestion Control (1)

TCP uses AIMD with loss signal to control congestion

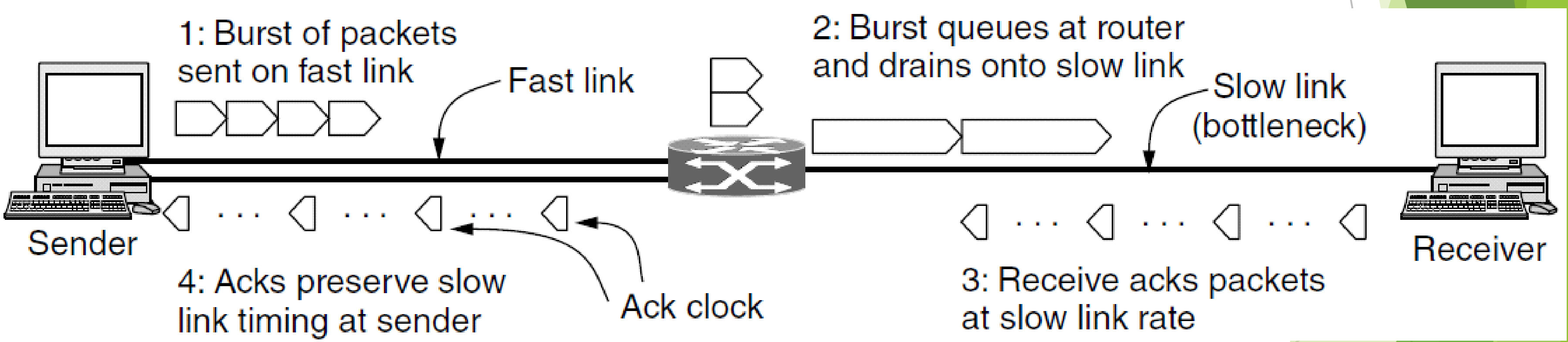
- Implemented as a congestion window (cwnd) for the number of segments that may be in the network
- Uses several mechanisms that work together

| 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



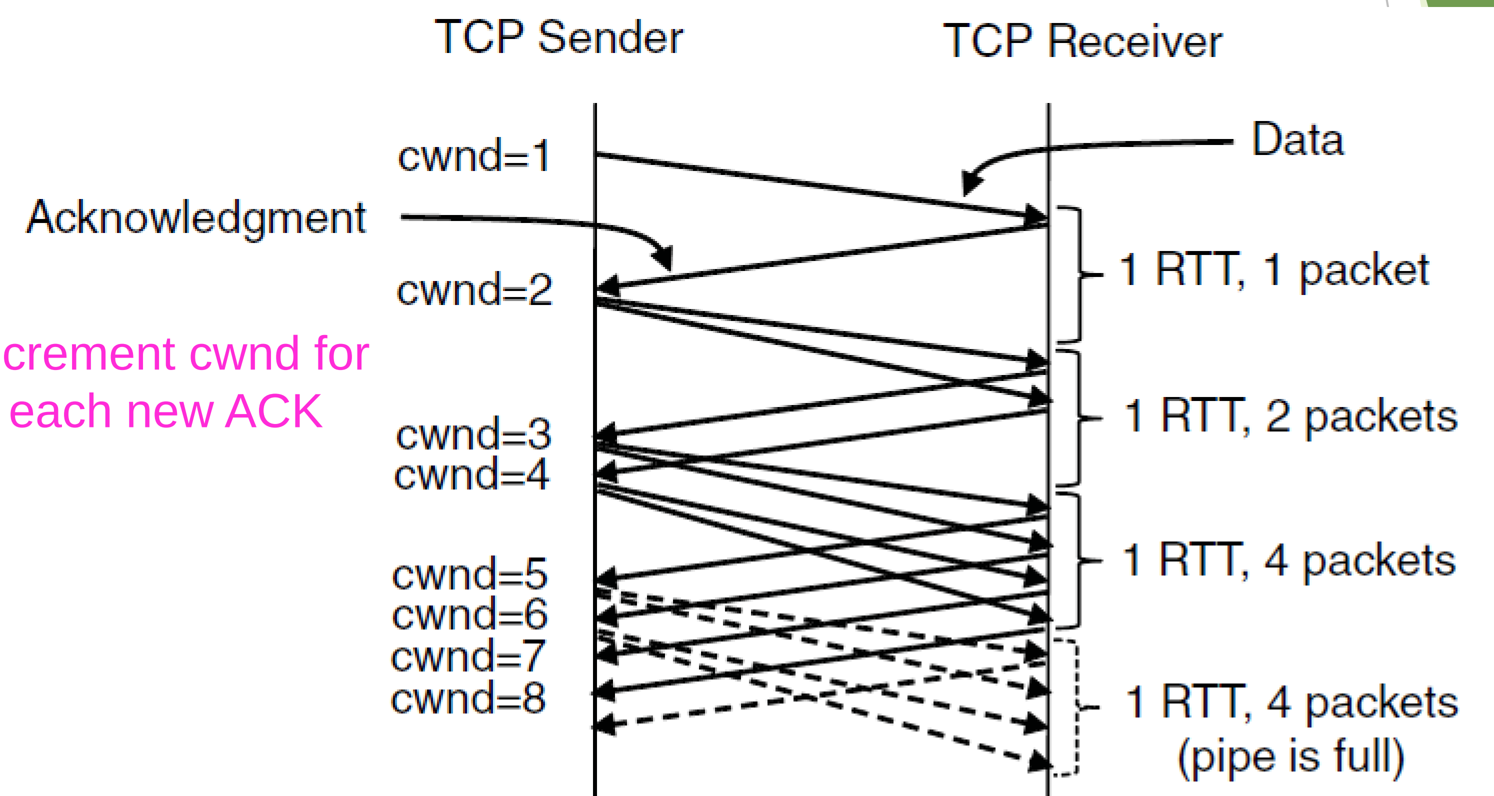

ACKs pace new segments into the network and smooth bursts

# TCP Congestion Control (3)

Slow start grows congestion window exponentially

- Doubles every RTT while keeping ACK clock going

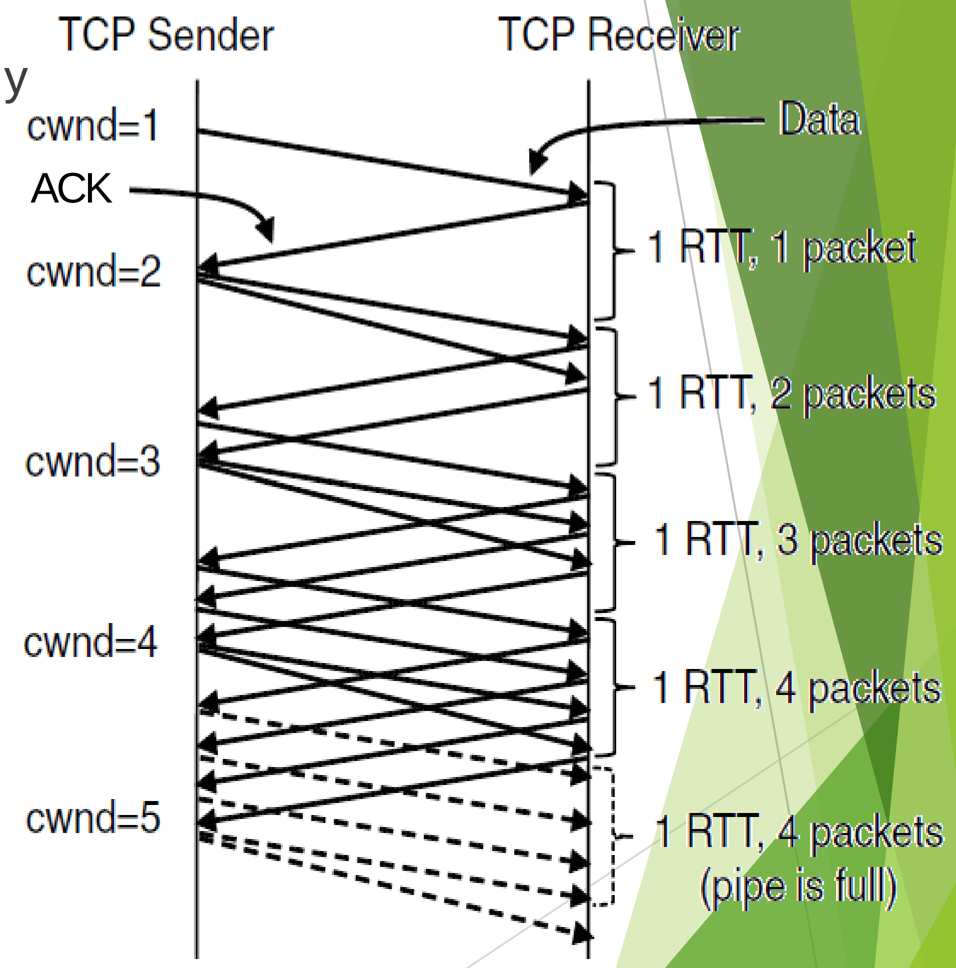
Increment cwnd for each new ACK



# TCP Congestion Control (4)

Additive increase grows cwnd slowly

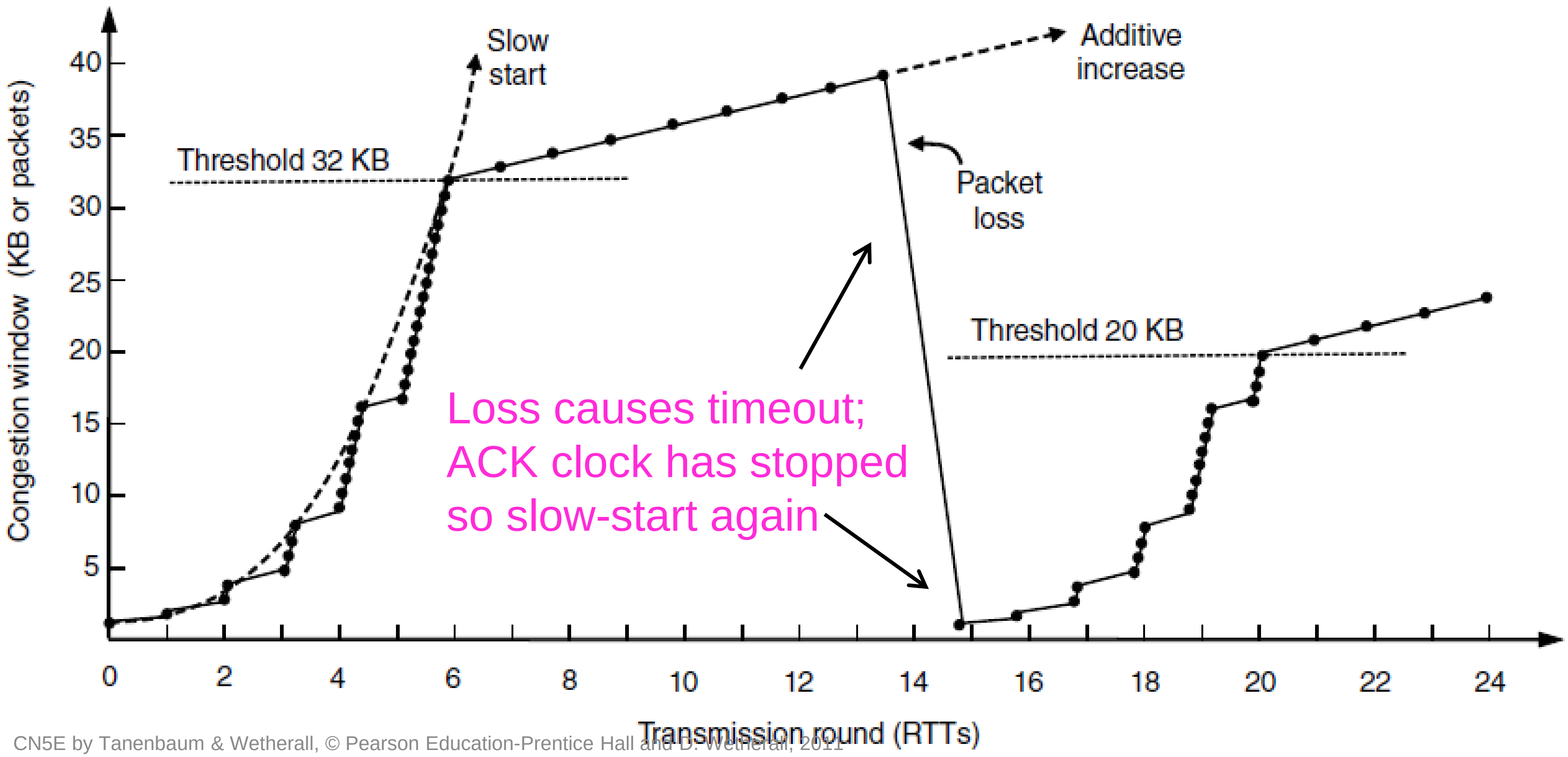
- Adds 1 every RTT
- Keeps ACK clock



# TCP Congestion Control (5)

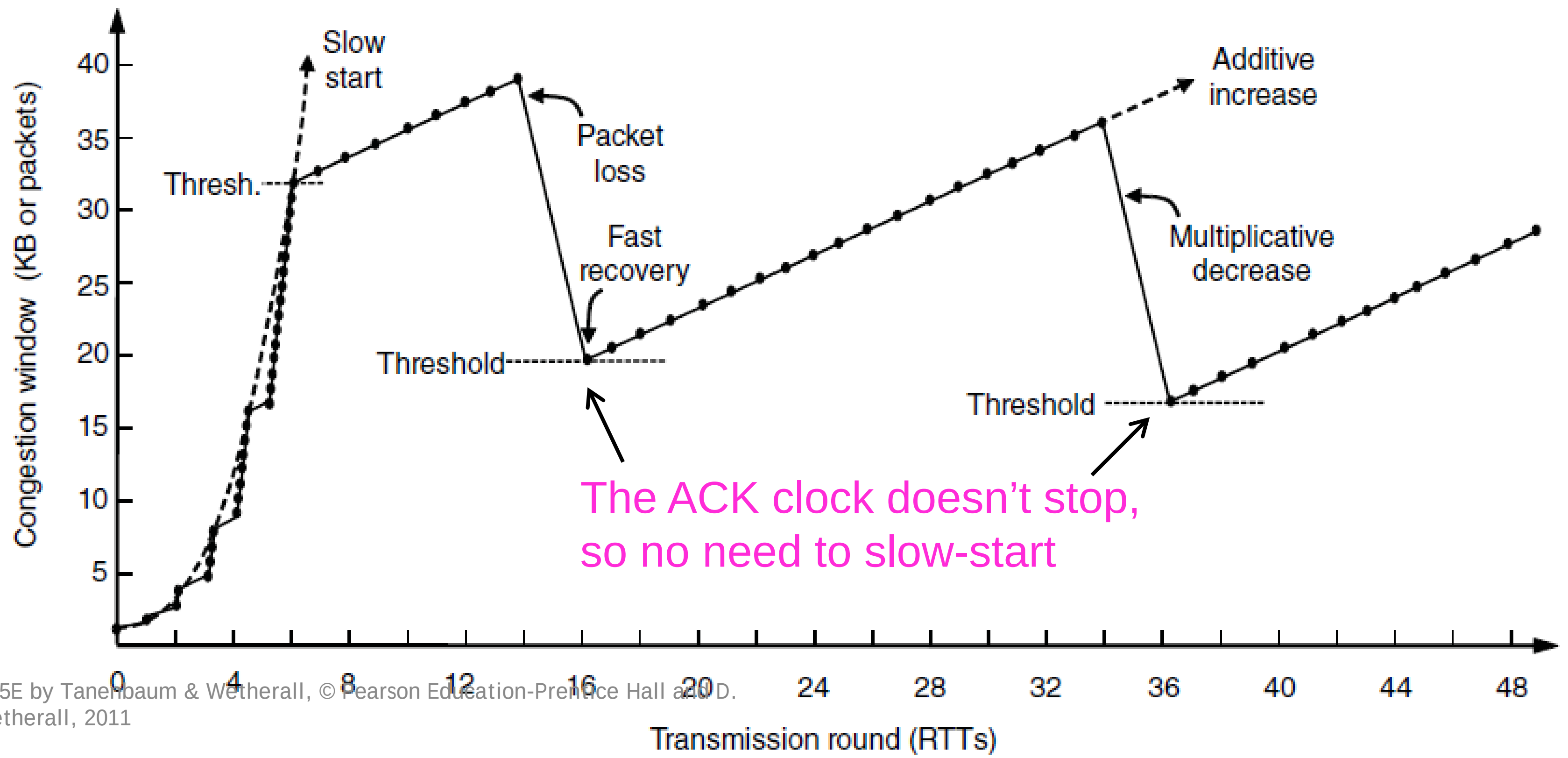
Slow start followed by additive increase (TCP Tahoe)

- 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
  - New packet for each dup. ACK until loss is repaired

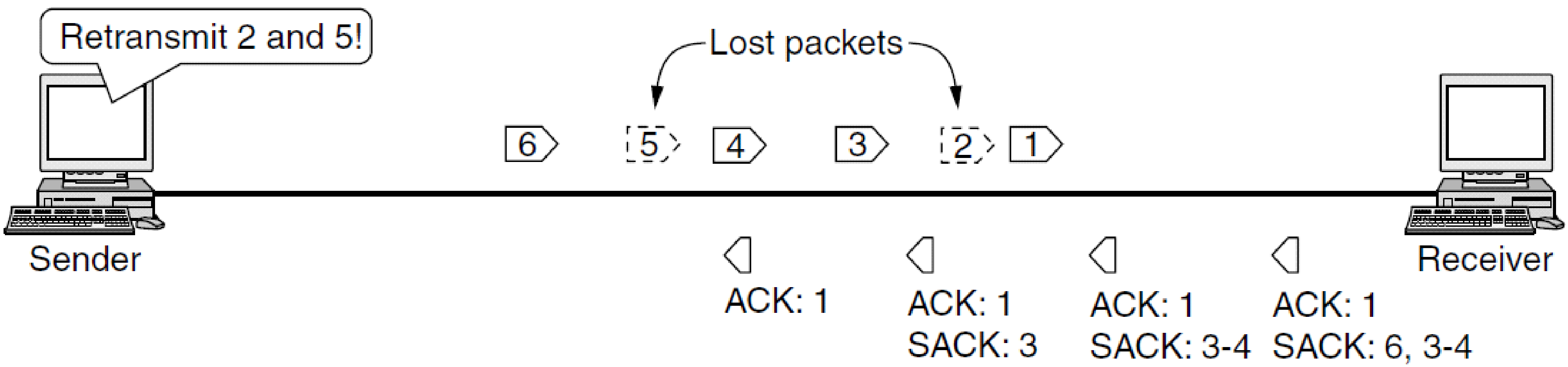



The ACK clock doesn't stop, so no need to slow-start

# 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

# 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

# Measuring Network Performance

Measurement is the key to understanding performance – but has its own pitfalls.

Example pitfalls:

- Caching: fetching Web pages will give surprisingly fast results if they are unexpectedly cached
- Timing: clocks may over/underestimate fast events
- Interference: there may be competing workloads

# Host Design for Fast Networks

Poor host software can greatly slow down networks.

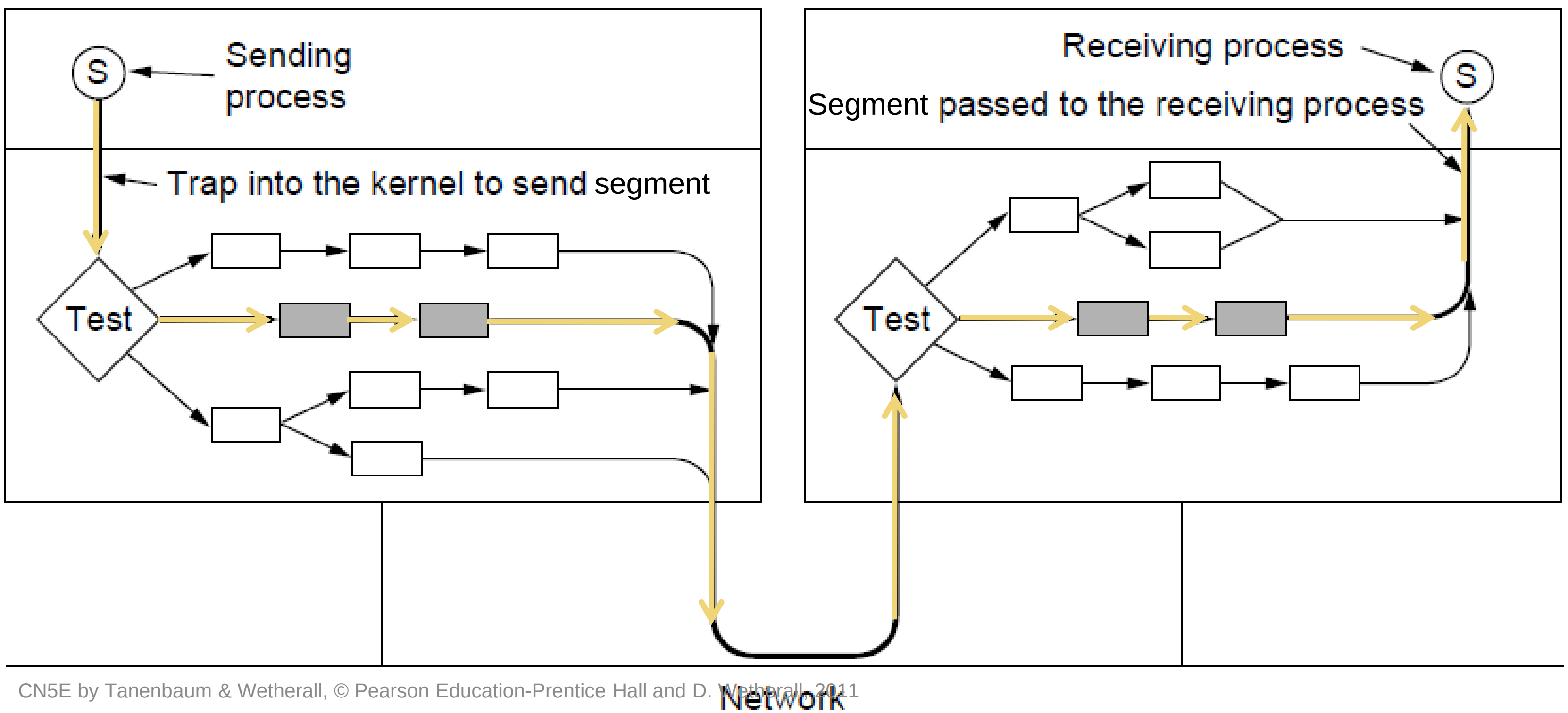
Rules of thumb for fast host software:

- 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)

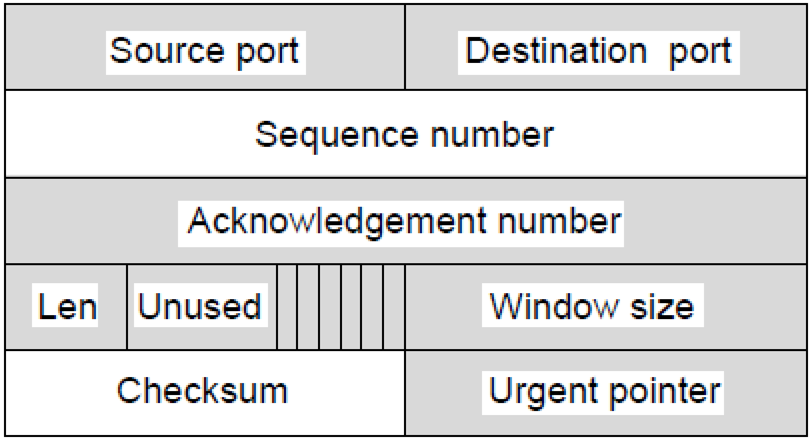
Speed up the common case with a fast path [pink]

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

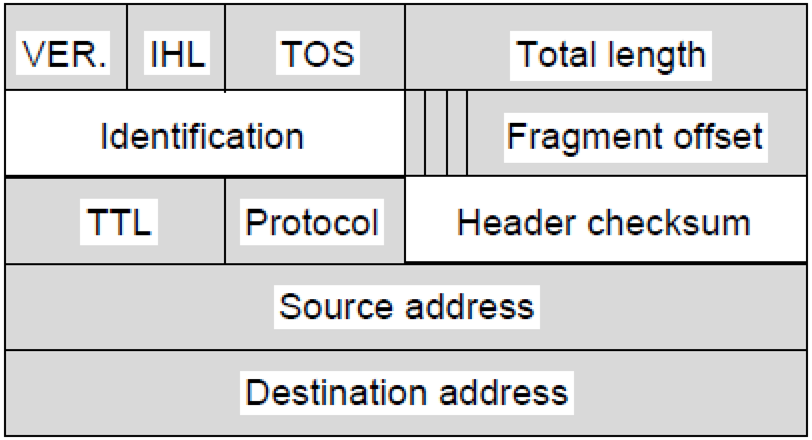


# Fast Segment Processing (2)

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)

# 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

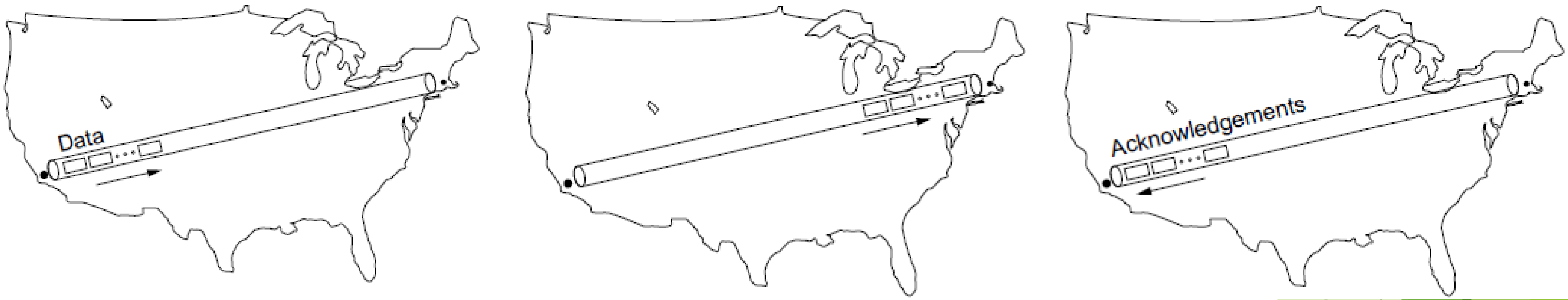
Header compression mitigates this problem

- 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

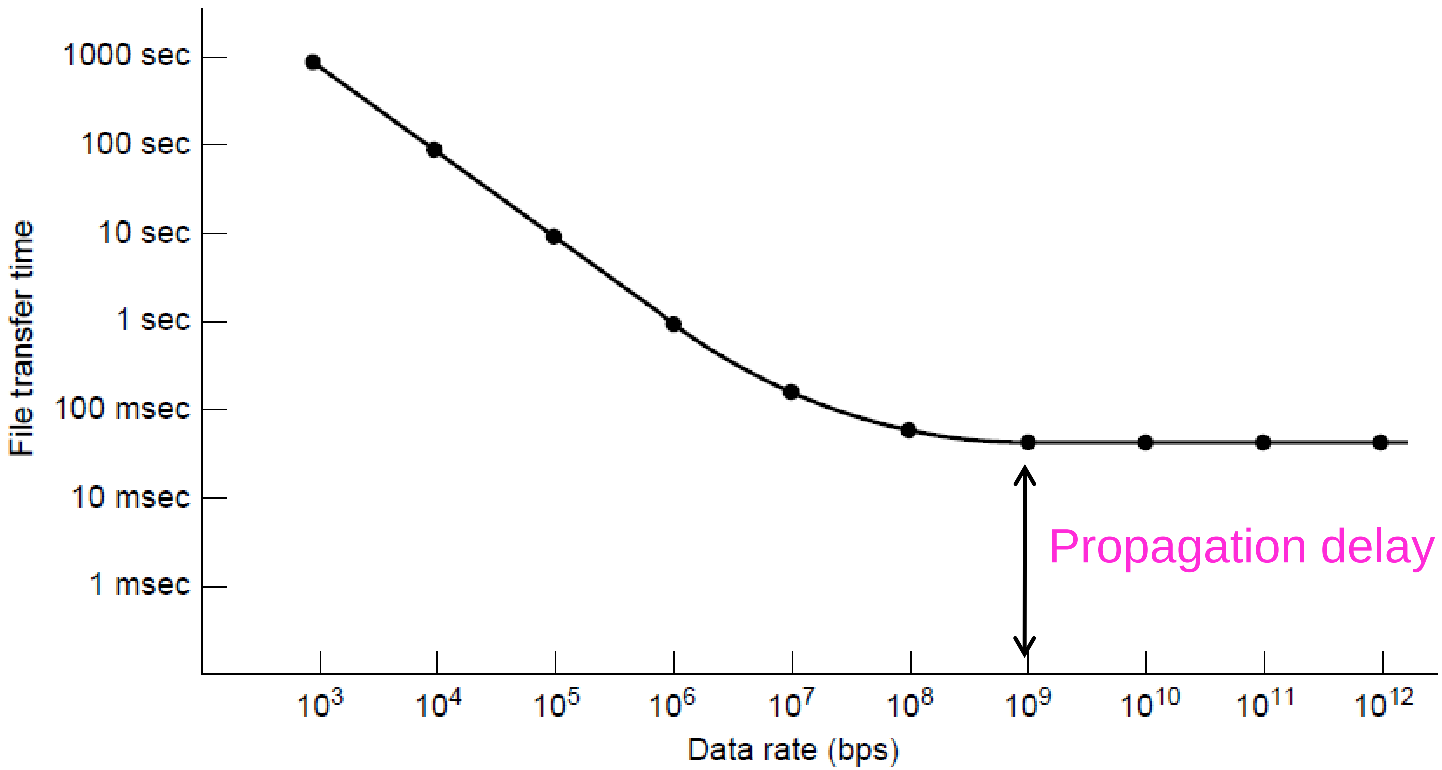
20ms after start

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

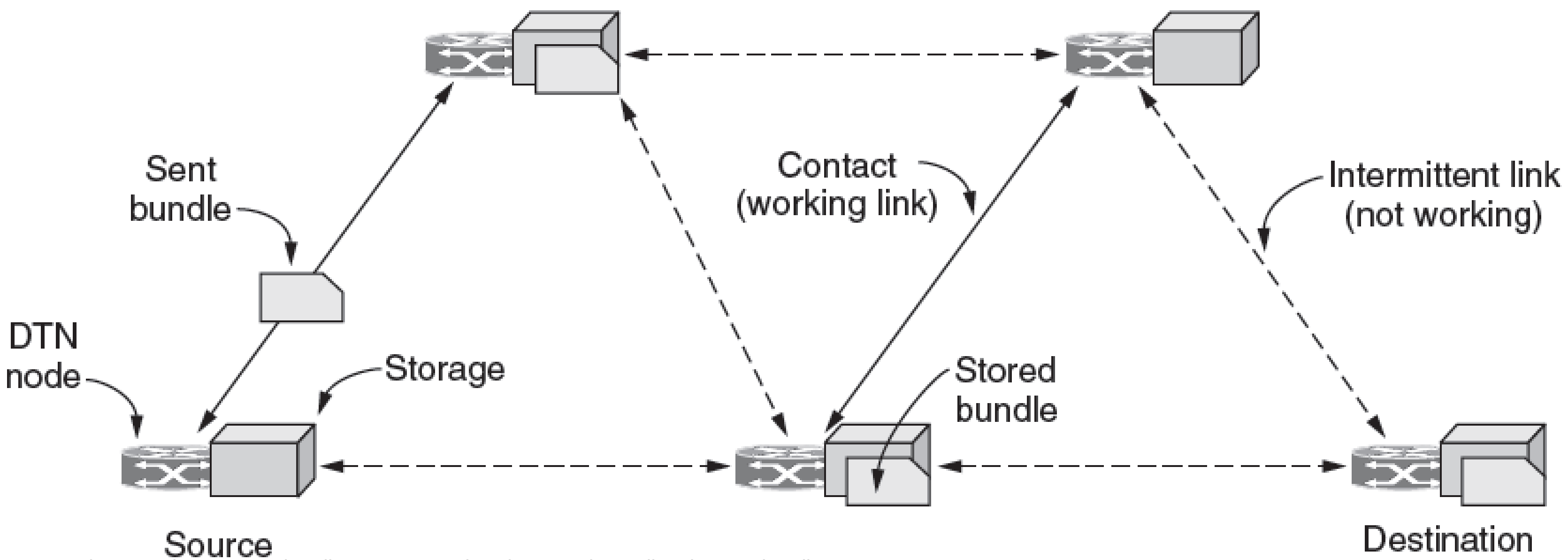
DTNs (Delay Tolerant Networks) store messages inside the network until they can be delivered

- DTN Architecture »
- Bundle Protocol »

# DTN Architecture (1)

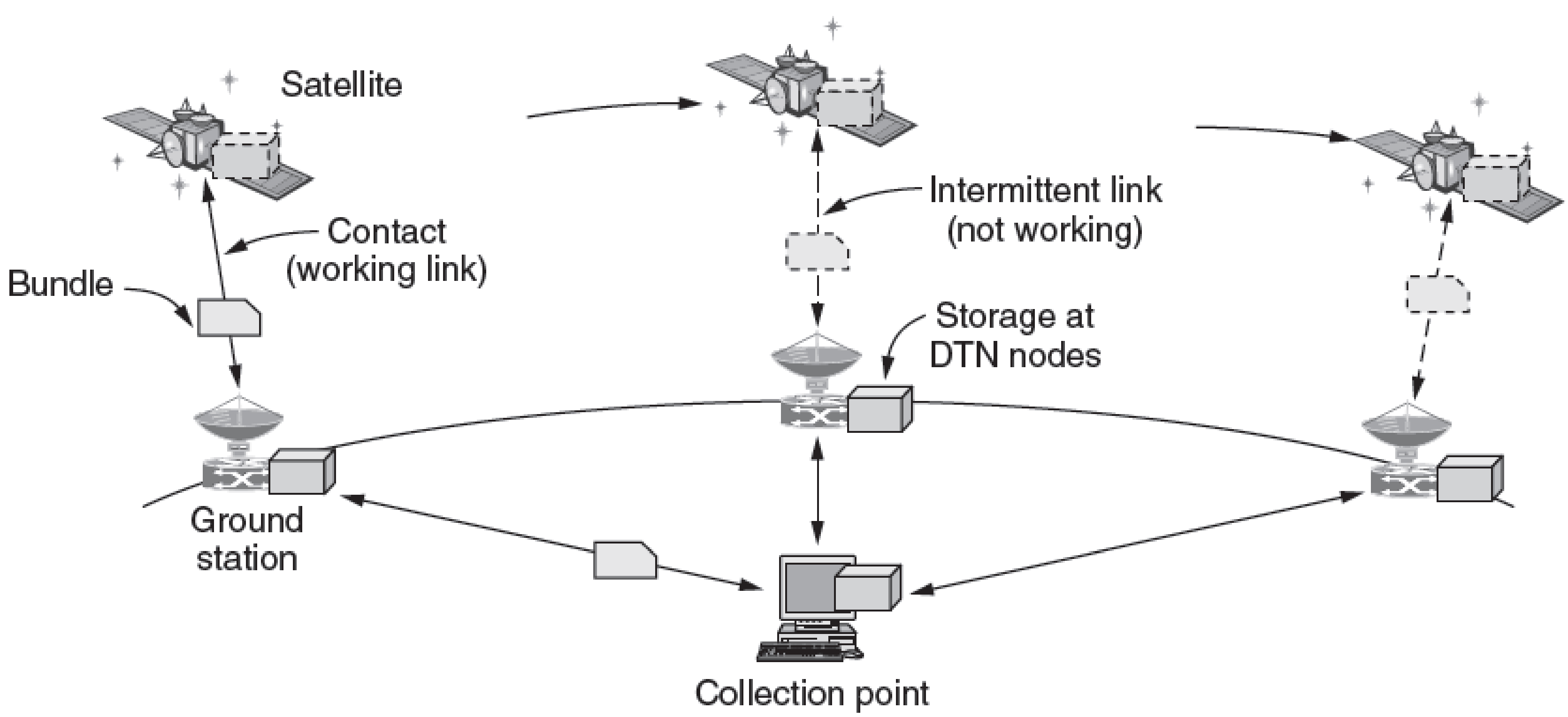
Messages called <u>bundles</u> are stored at DTN nodes while waiting for an intermittent link to become a contact

- Bundles might wait hours, not milliseconds in routers
- May be no working end-to-end path at any time



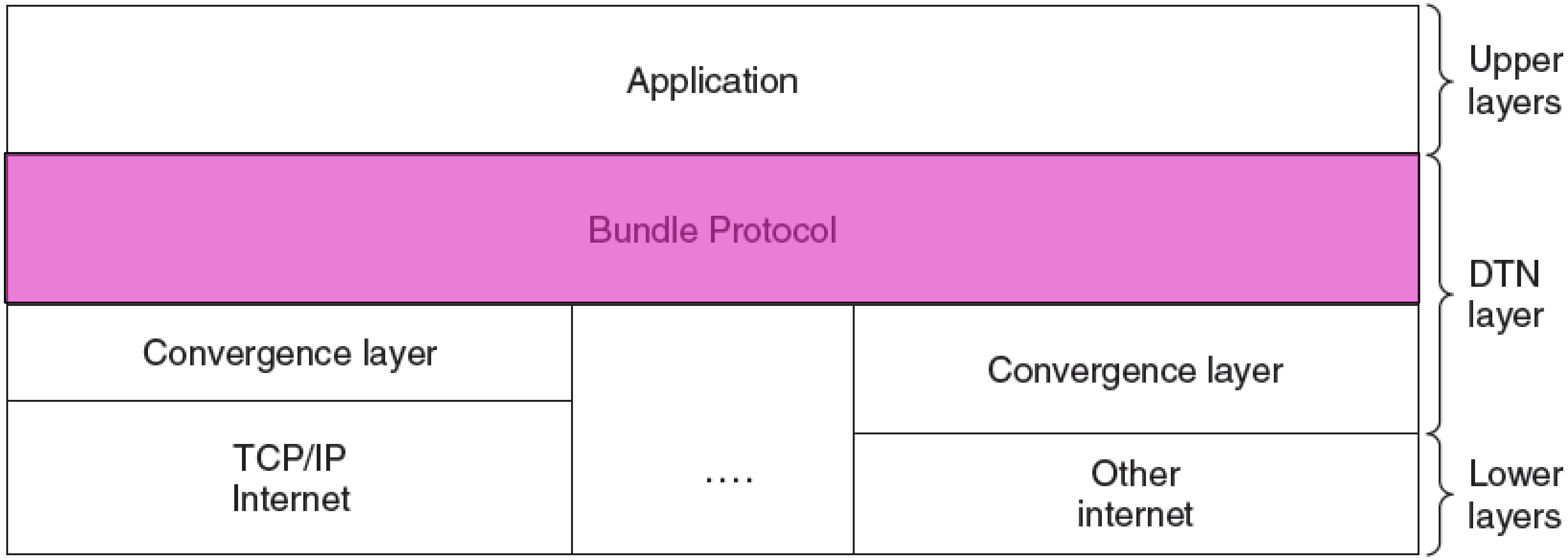
# DTN Architecture (2)

Example DTN connecting a satellite to a collection point



# Bundle Protocol (1)

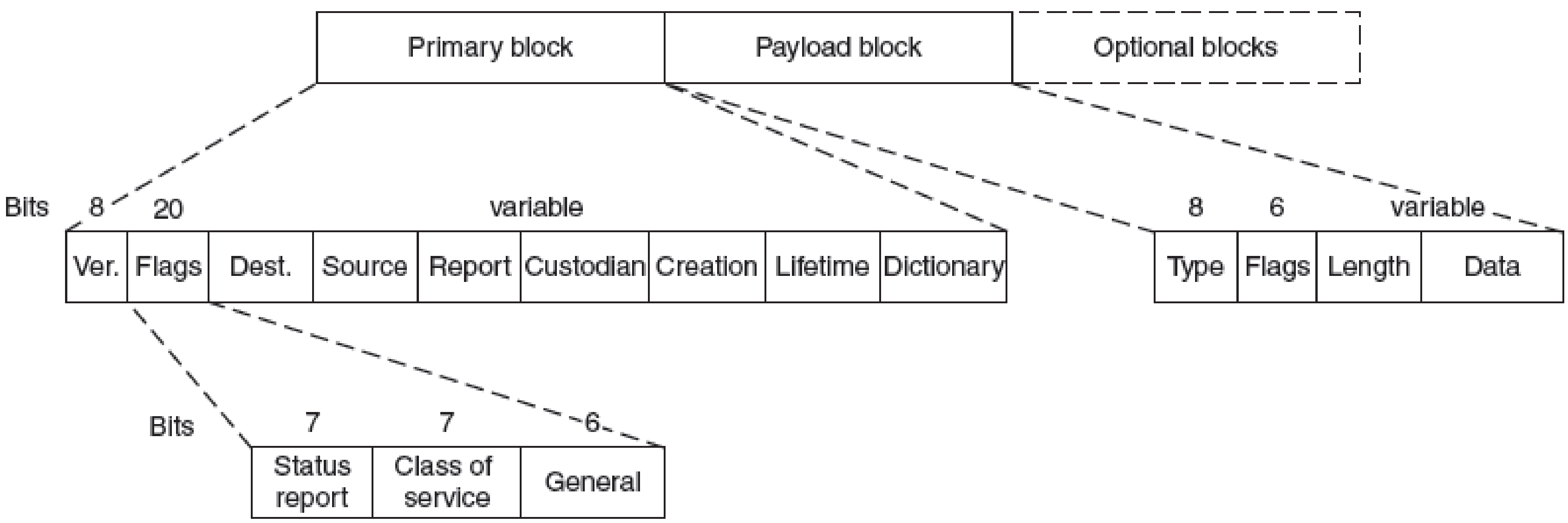
The Bundle protocol uses TCP or other transports and provides a DTN service to applications



# 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



# End

Chapter 6