CMSC 417

Computer Networks

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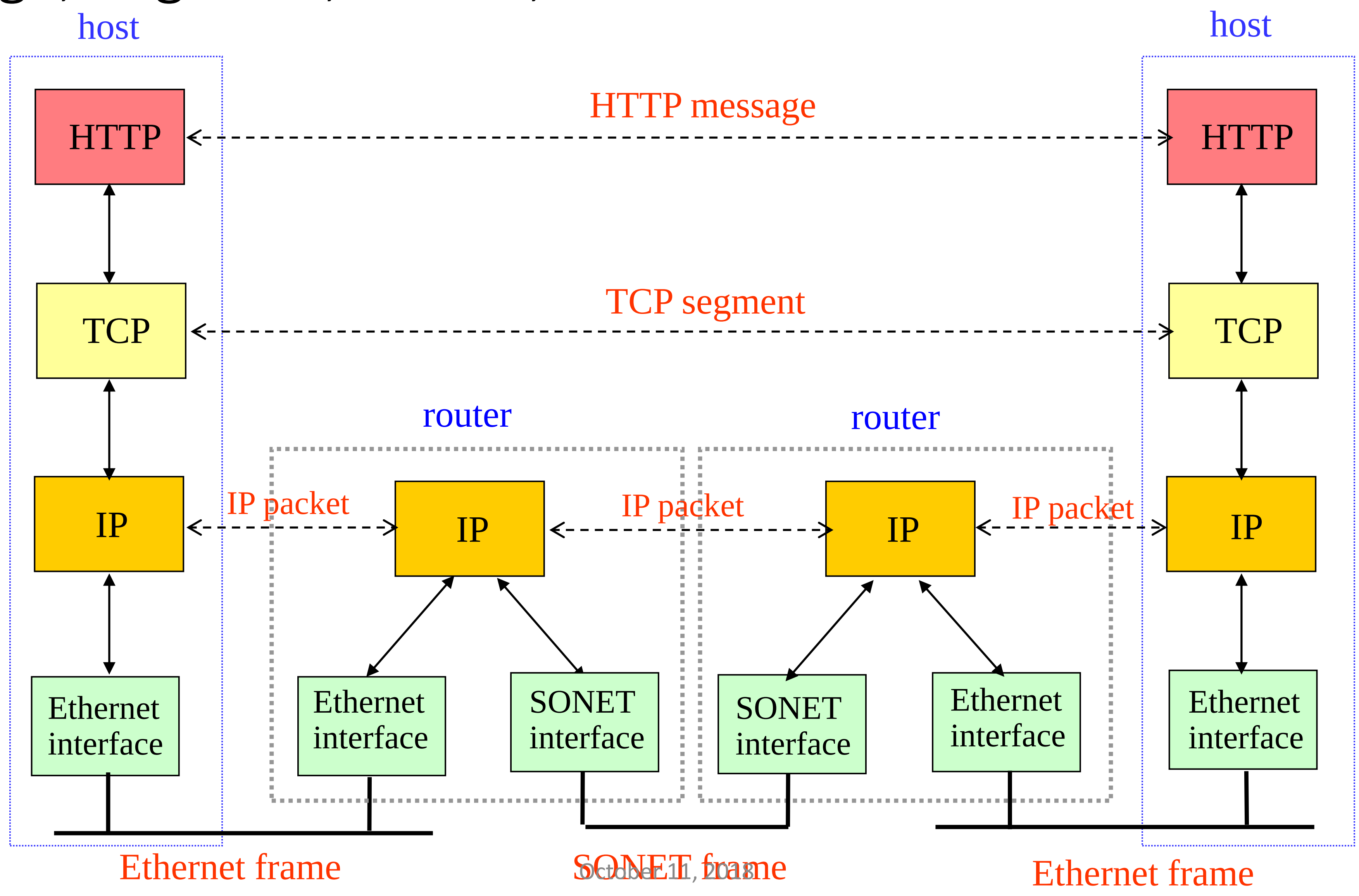
Message, Segment, Packet, and Frame



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

Acknowledgements & Timeouts

- An *acknowledgement* (ACK) is a packet sent by one host in response to a packet it has received
  - Making a packet an ACK is simply a matter of changing a field in the transport header
  - Data can be *piggybacked* in ACKs
- A *timeout* is a signal that an ACK to a packet that was sent has not yet been received within a specified timeframe
  - A timeout triggers a *retransmission* of the original packet from the sender
  - How are timers set?

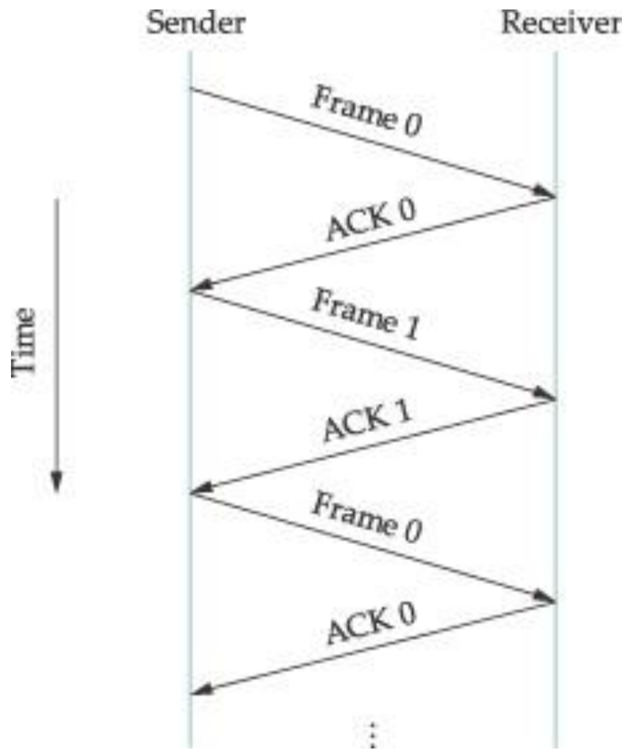
Propagation Delay

- Propagation delay is defined as the delay between transmission and receipt of packets between hosts
- Propagation delay can be used to estimate timeout period
- How can propagation delay be measured?
- What else must be considered in the measurement?

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 – 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 <u>stop-and-wait</u>
- 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)

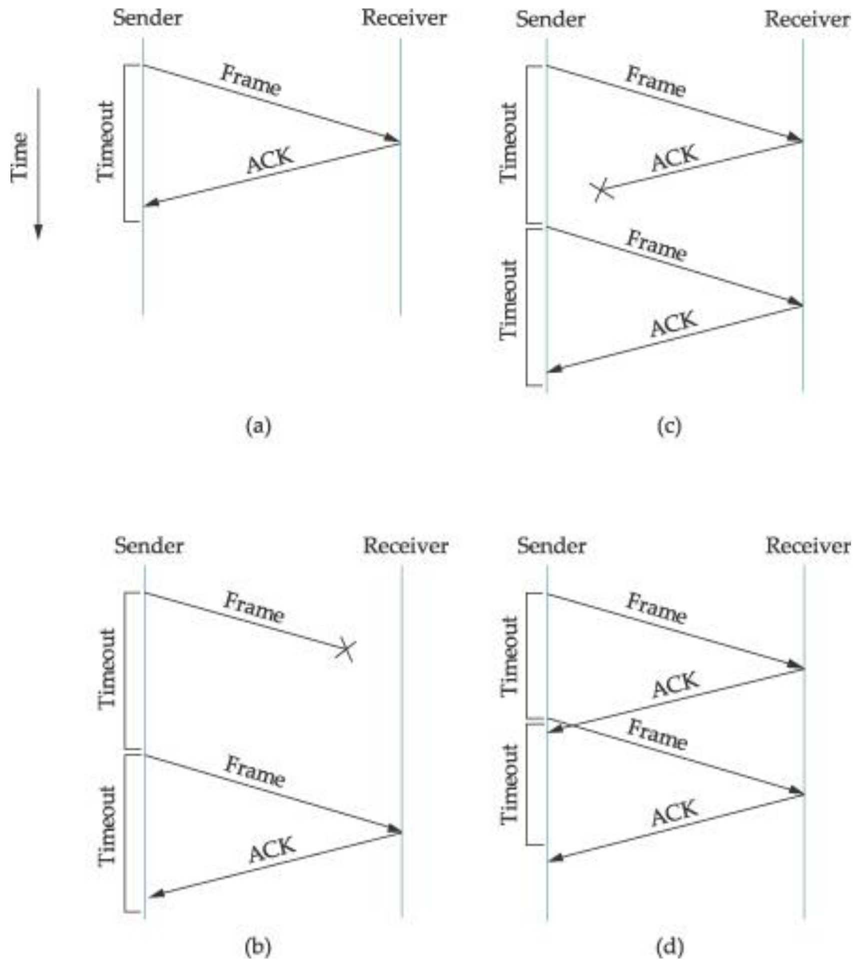
ARQ (Automatic Repeat reQuest) adds error control

- 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



Duplicate Frames

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

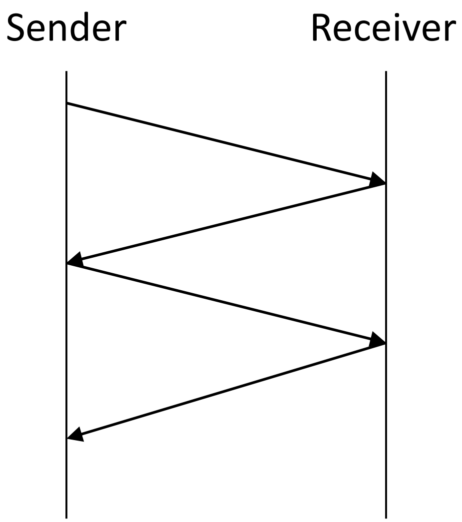
If it's new then take it and advance expected frame

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);
       }
  }
}
```

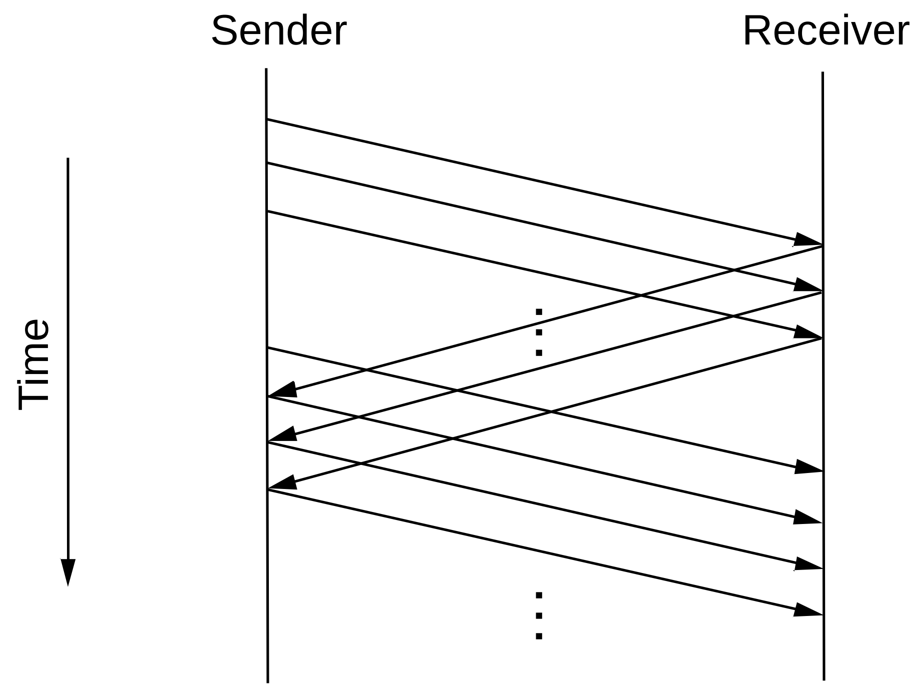
Stop-and-Wait Process



- Sender doesn’t send next packet until he’s sure receiver has received the last packet
- The packet/Ack sequence enables reliability
- Sequence numbers help avoid problem of duplicate packets
- Problem: keeping the pipe full
- Example
  - 1.5Mbps link x 45ms RTT = 67.5Kb (8KB) delay bandwidth product
  - 1KB frames implies 1/8th link utilization

Solution: Pipelining via Sliding Window

- Allow multiple outstanding (un-ACKed) frames
- Upper bound on un-ACKed frames, called *window*

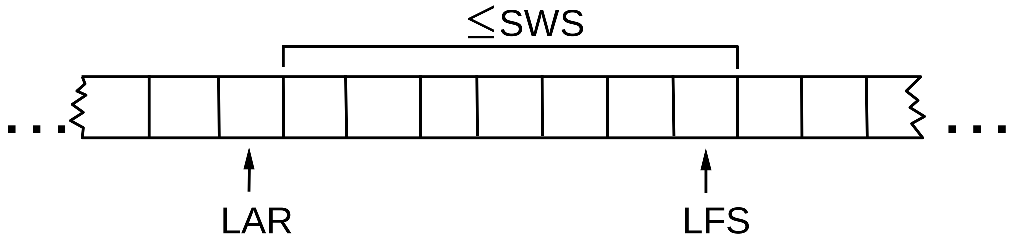


Buffering on Sender and Receiver

- Sender needs to buffer data so that if data is lost, it can be resent
- Receiver needs to buffer data so that if data is received out of order, it can be held until all packets are received
  - Flow control
- How can we prevent sender overflowing receiver's buffer?
  - Receiver tells sender its buffer size during connection setup
- How can we insure reliability in pipelined transmissions?
  - Go-Back-N
    - Send all N unACKed packets when a loss is signaled
    - Inefficient
  - Selective repeat
    - Only send specifically unACKed packets
    - A bit trickier to implement

Sliding Window: Sender

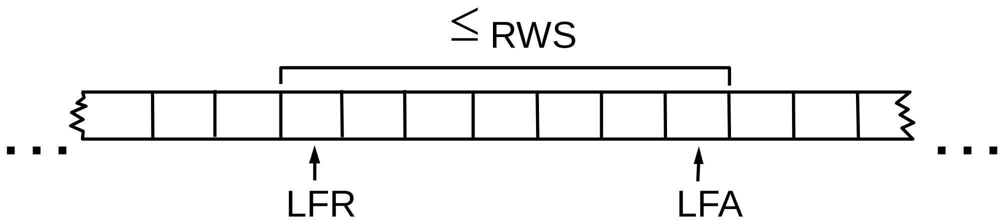
- Assign sequence number to each frame (**`SeqNum`**)
- Maintain three state variables:
  - send window size (**`SWS`**)
  - last acknowledgment received (**`LAR`**)
  - last frame sent (**`LFS`**)
- Maintain invariant: **`LFS`** - **`LAR`** <= **`SWS`**



- Advance **`LAR`** when ACK arrives
- Buffer up to **`SWS`** frames

Sliding Window: Receiver

- Maintain three state variables
  - receive window size (**`RWS`**)
  - largest frame acceptable (**`LFA`**)
  - last frame received (**`LFR`**)
- Maintain invariant: **`LFA`** - **`LFR`** <= **`RWS`**



- Frame **`SeqNum`** arrives:
  - if **`LFR`** < **`SeqNum`** < = **`LFA`** accept
  - if **`SeqNum`** < = **`LFR`** or **`SeqNum`** > **`LFA`** discarded
- Send *cumulative* ACKs – send ACK for largest frame such that all frames less than this have been received

Sequence Number Space

- **`SeqNum`** field is finite; sequence numbers wrap around
- Sequence number space must be larger then number of outstanding frames
- **`SWS <= MaxSeqNum-1`** is not sufficient
  - suppose 3-bit **`SeqNum`** field (0..7)
  - **`SWS=RWS=7`**
  - sender transmit frames 0..6
  - arrive successfully, but ACKs lost
  - sender retransmits 0..6
  - receiver expecting 7, 0..5, but receives the original incarnation of 0..5
- **`SWS < (MaxSeqNum+1)/2`** is correct rule
- Intuitively, **`SeqNum`** “slides” between two halves of sequence number space

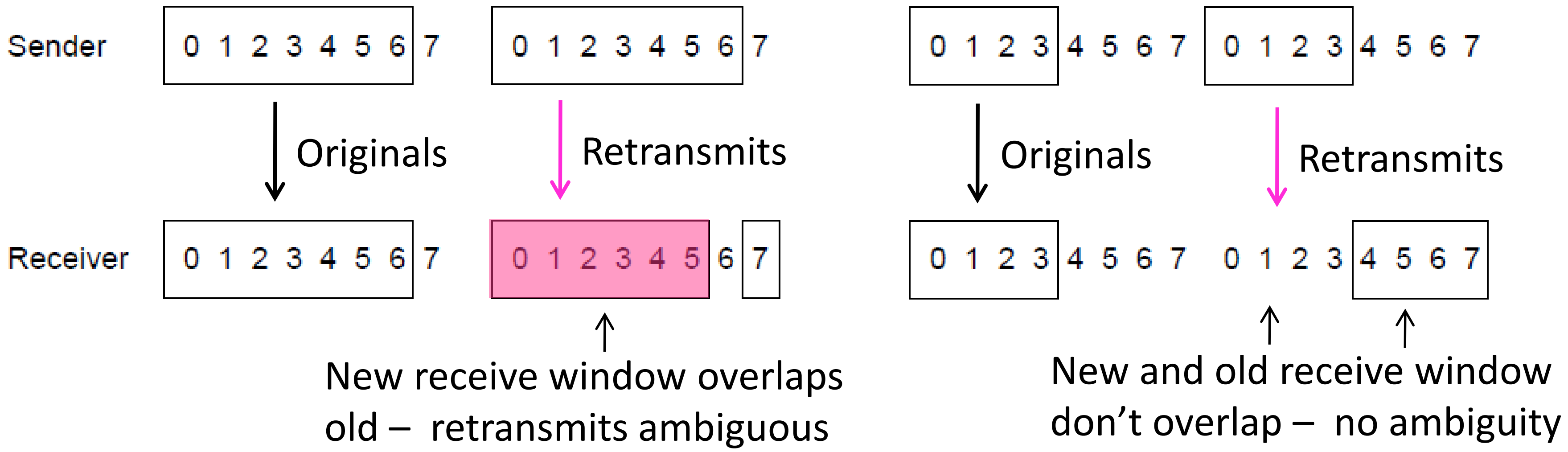
Sequence Number Space

For correctness, we require:

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

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

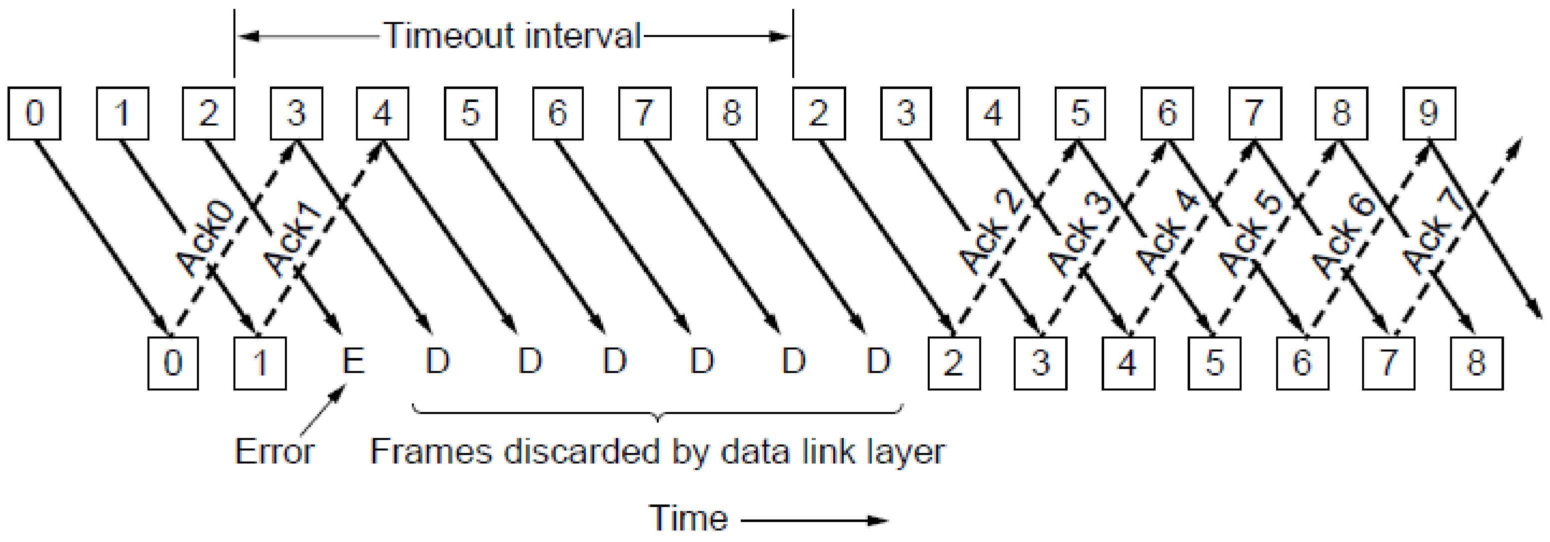
Correct (s=8, w=4) – enough sequence numbers



Go-Back-N

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

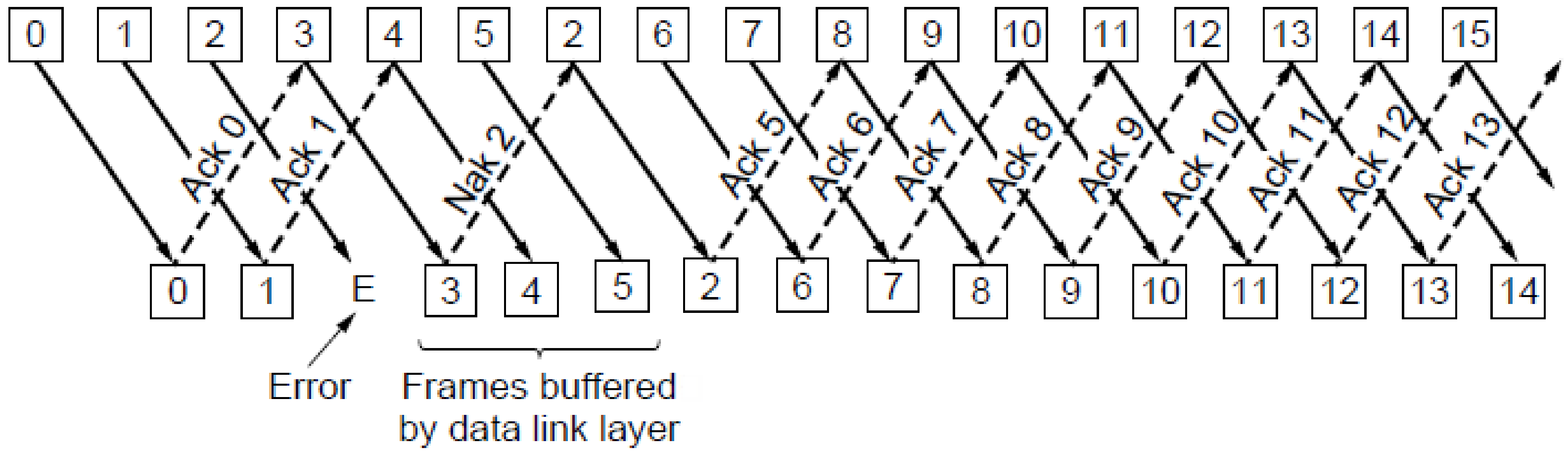
Tradeoff made for Go-Back-N:

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

Selective Repeat

Receiver accepts frames anywhere in receive window

- Cumulative ack indicates highest in-order frame
- NAK (negative ack) causes sender retransmission of a missing frame before a timeout resends window



Selective Repeat

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)

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

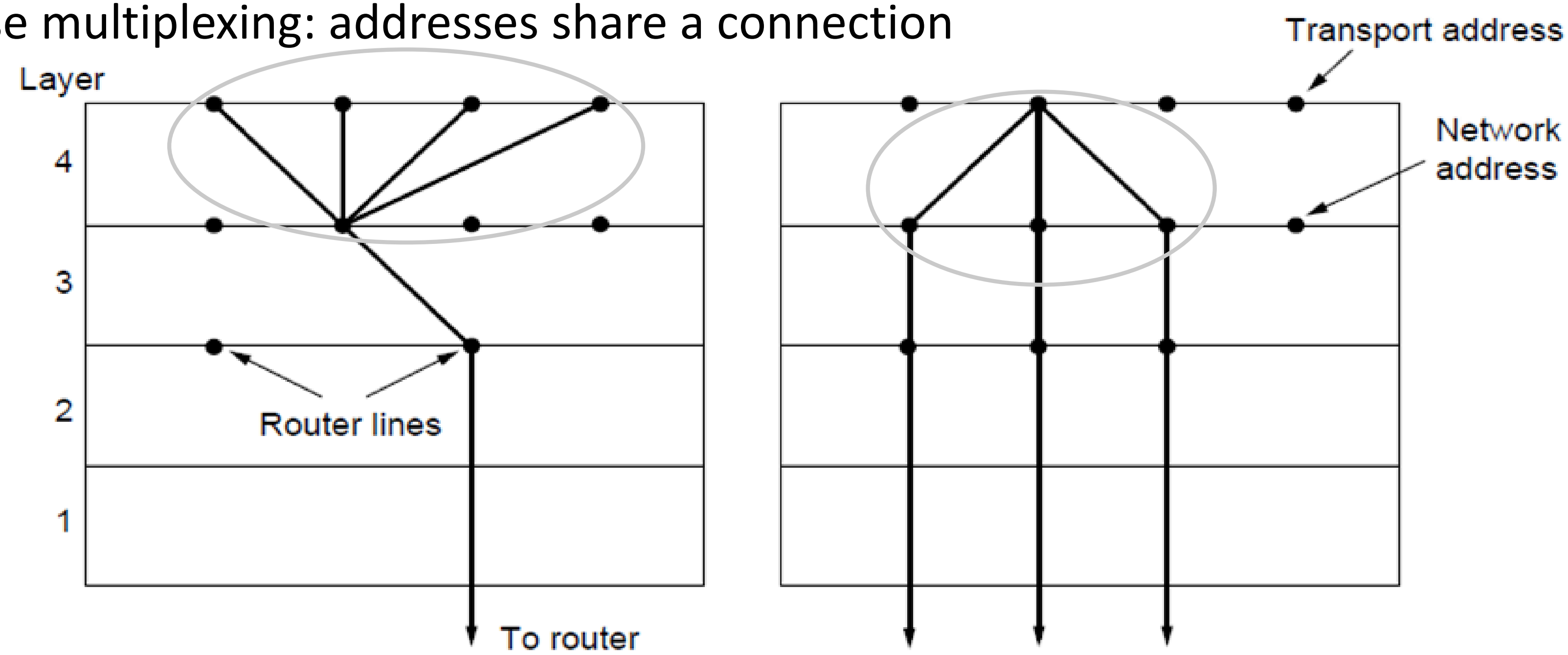
http://www2.rad.com/networks/2004/sliding_window/

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



Multiplexing

Inverse Multiplexing

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

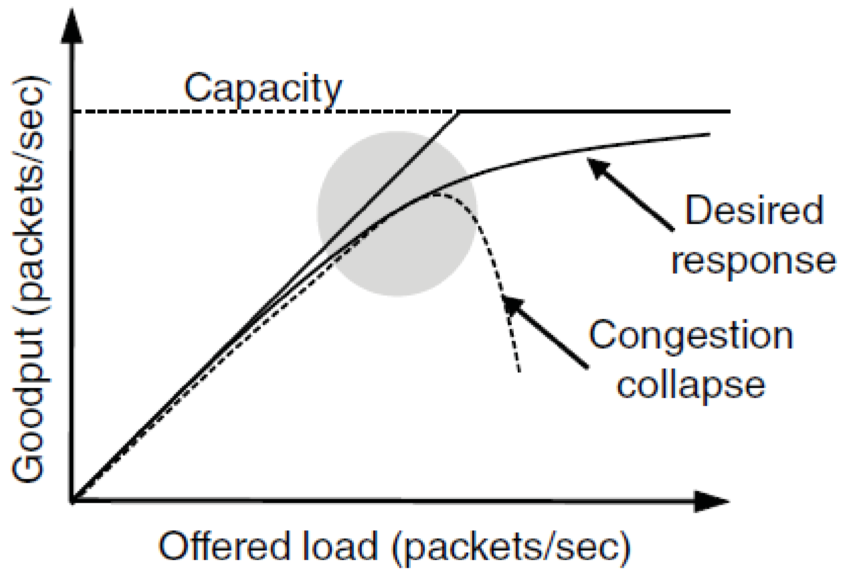
Two layers are responsible for congestion control:

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

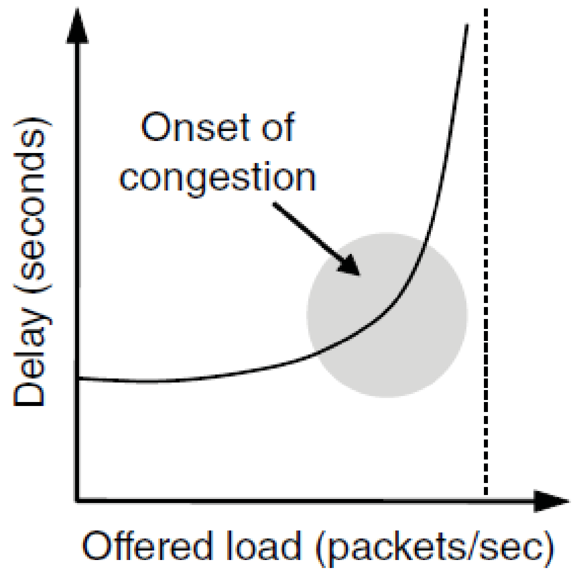
- Desirable bandwidth allocation »
- Regulating the sending rate »
- Wireless issues »

Desirable Bandwidth Allocation (1)

Efficient use of bandwidth gives high goodput, low delay



Goodput rises more slowly than load when congestion sets in

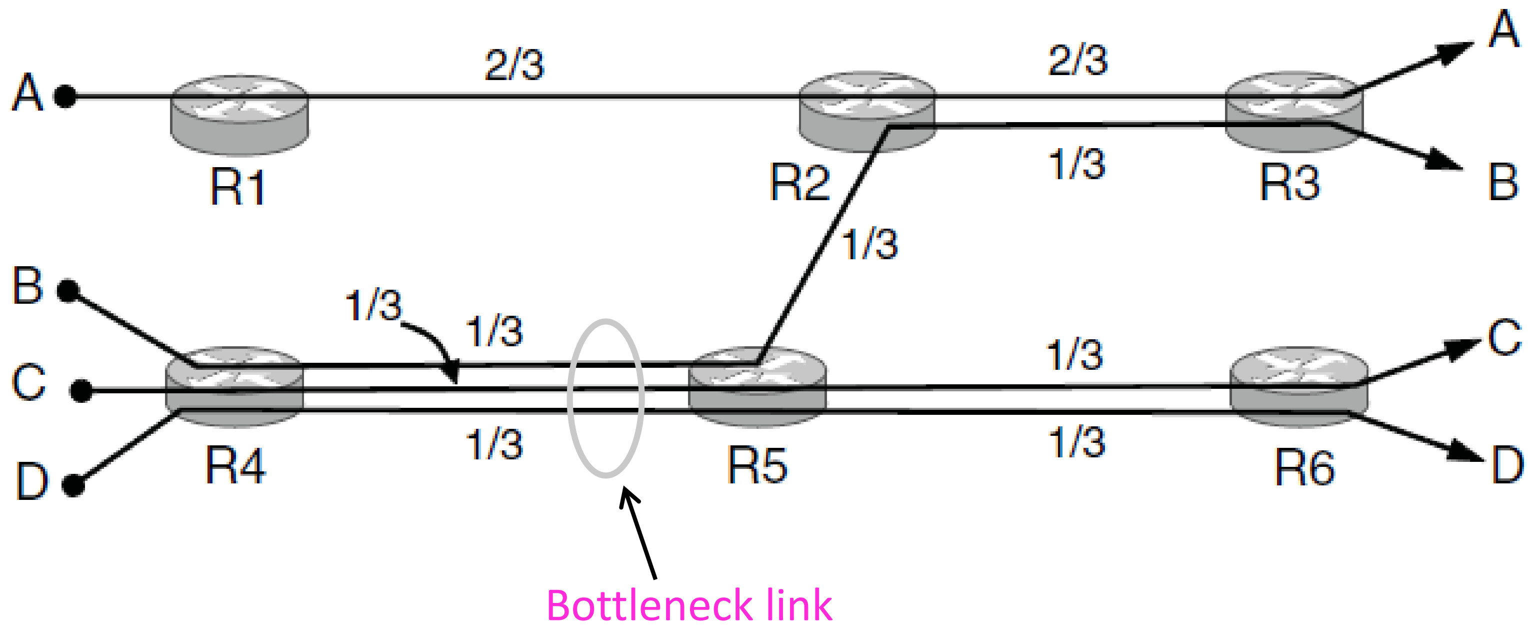


Delay begins to rise sharply when congestion sets in

Desirable Bandwidth Allocation (2)

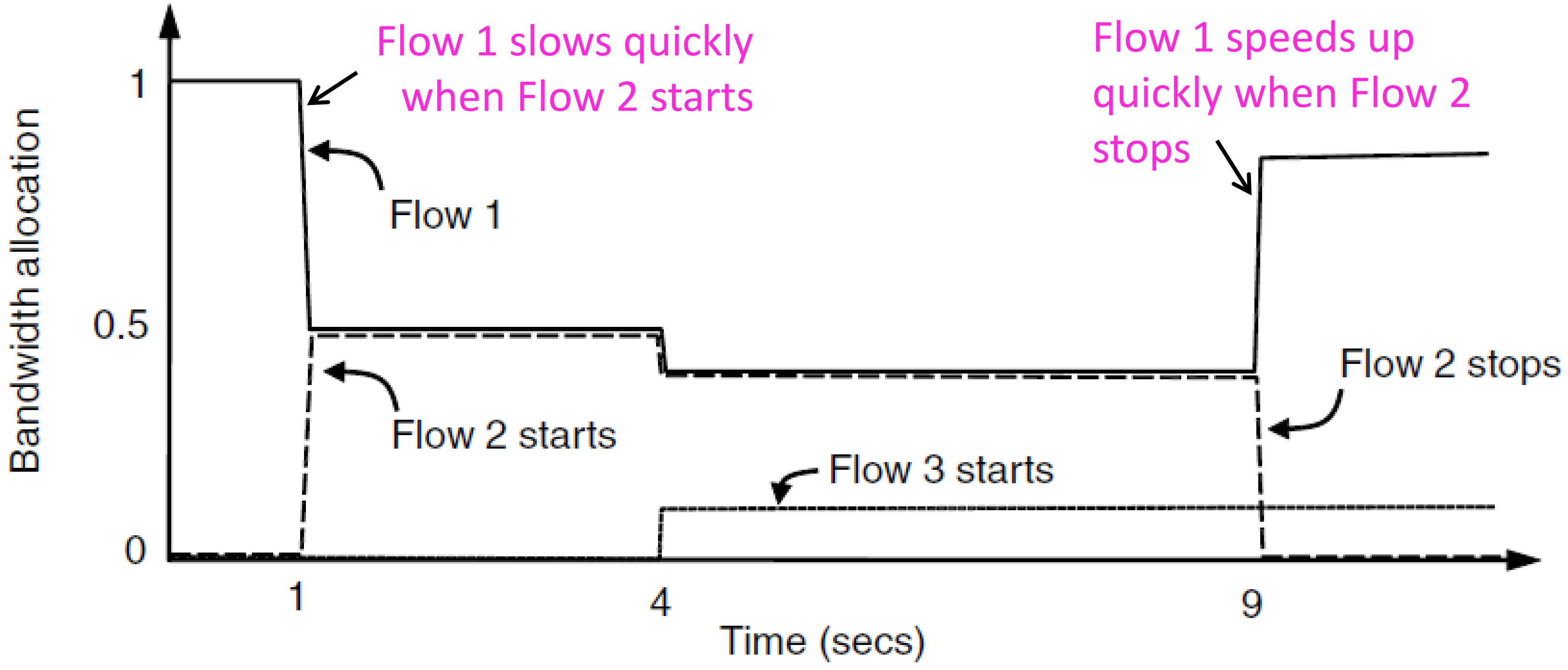
Fair use gives bandwidth to all flows (no starvation)

- Max-min fairness gives equal shares of bottleneck



Desirable Bandwidth Allocation (3)

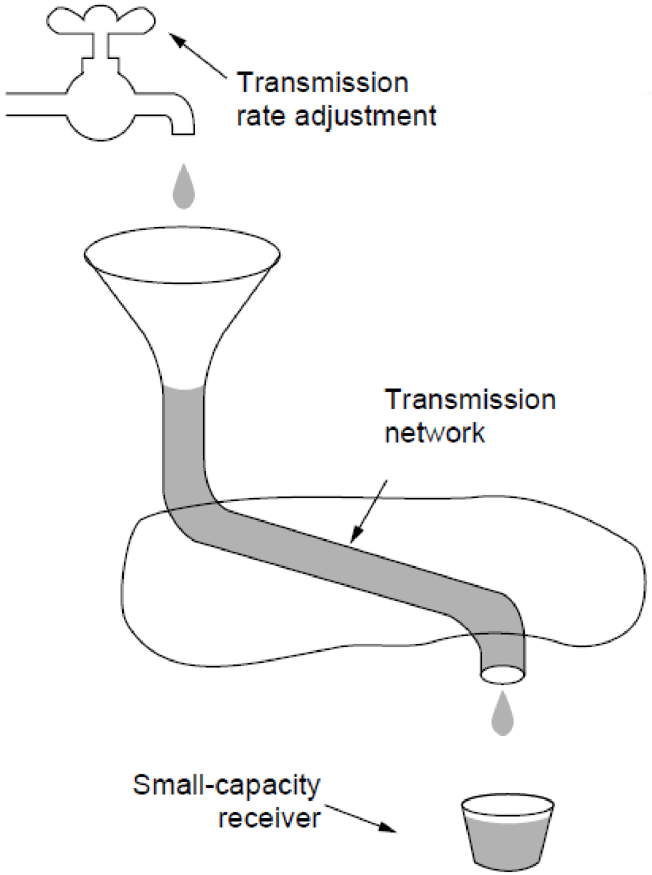
We want bandwidth levels to converge quickly when traffic patterns change



Regulating the Sending Rate (1)

Sender may need to slow down for different reasons:

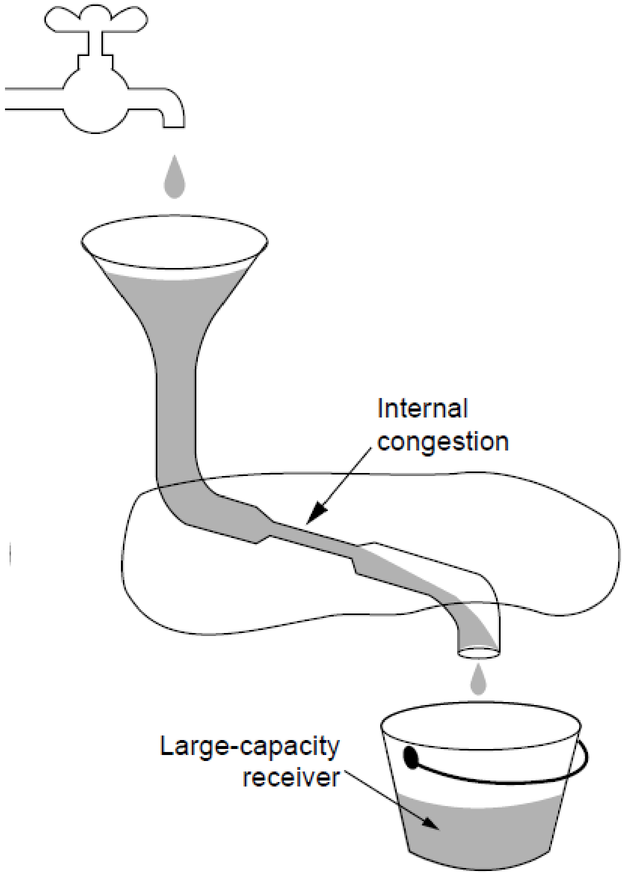
- Flow control, when the receiver is not fast enough [right]
- Congestion, when the network is not fast enough [over]



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

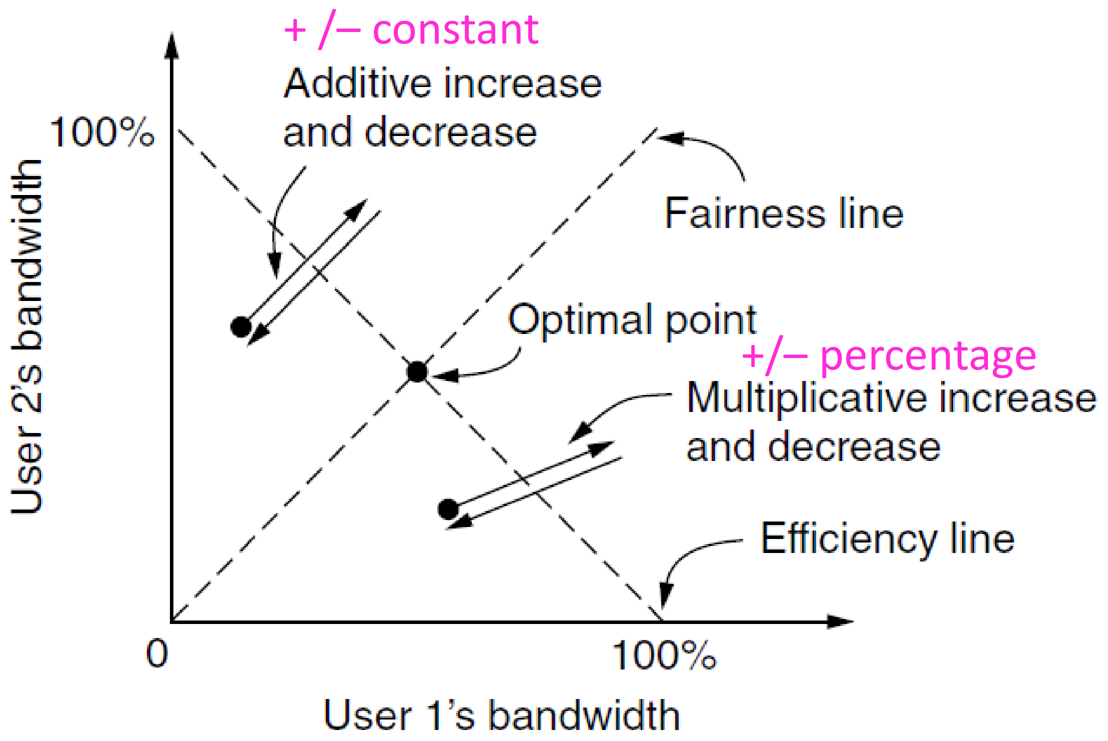
Regulating the Sending Rate (3)

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)

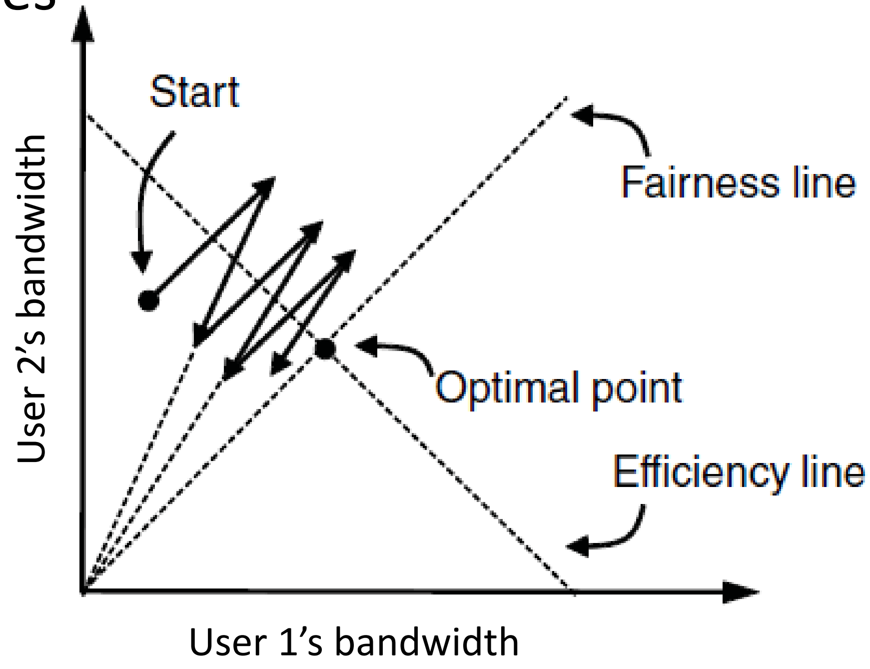
If two flows increase/decrease their bandwidth in the same way when the network signals free/busy they will not converge to a fair allocation



Regulating the Sending Rate (4)

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

- TCP uses



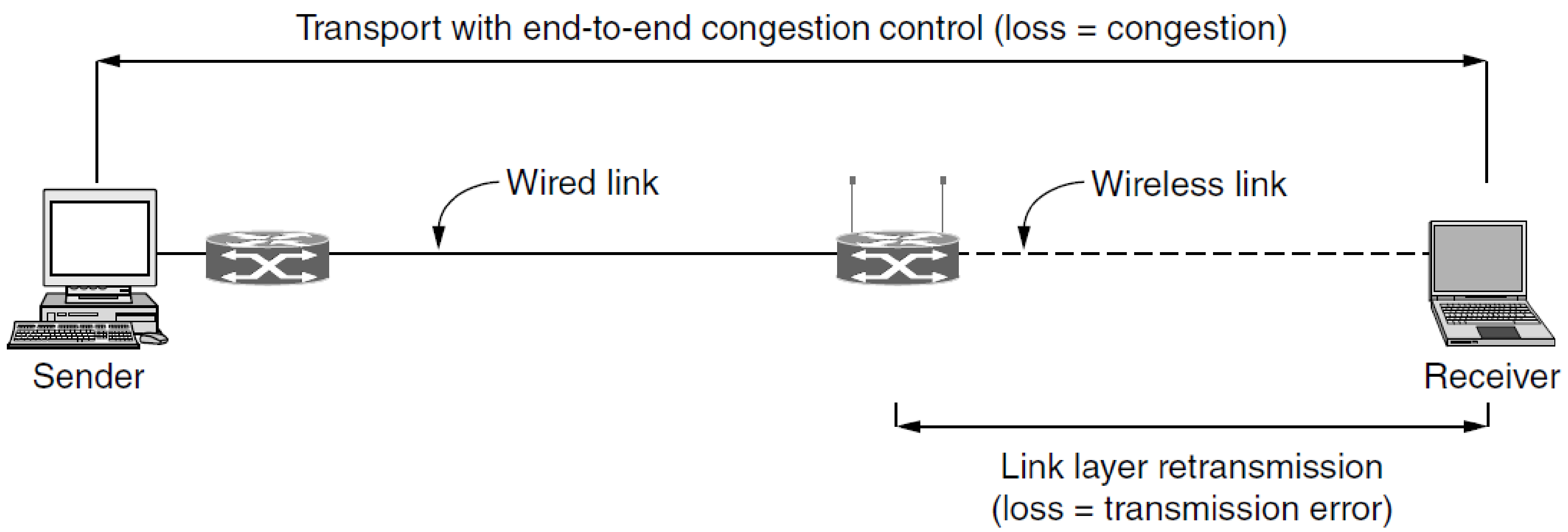
Wireless Issues

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.

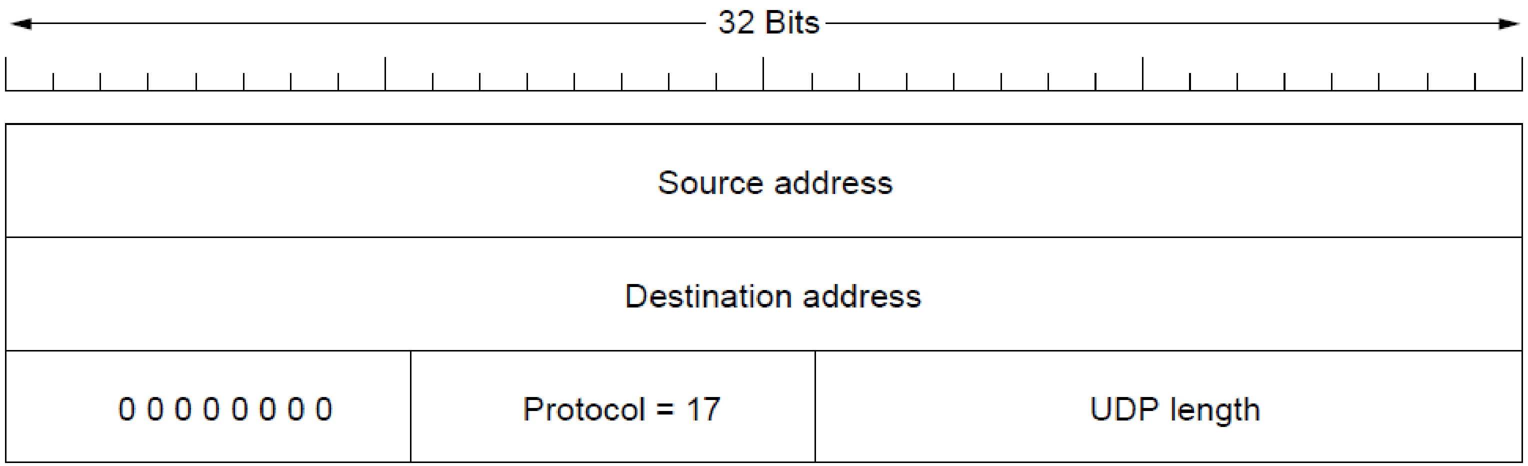
32 Bits

Source port	Destination port
UDP length	UDP checksum

Introduction to UDP (2)

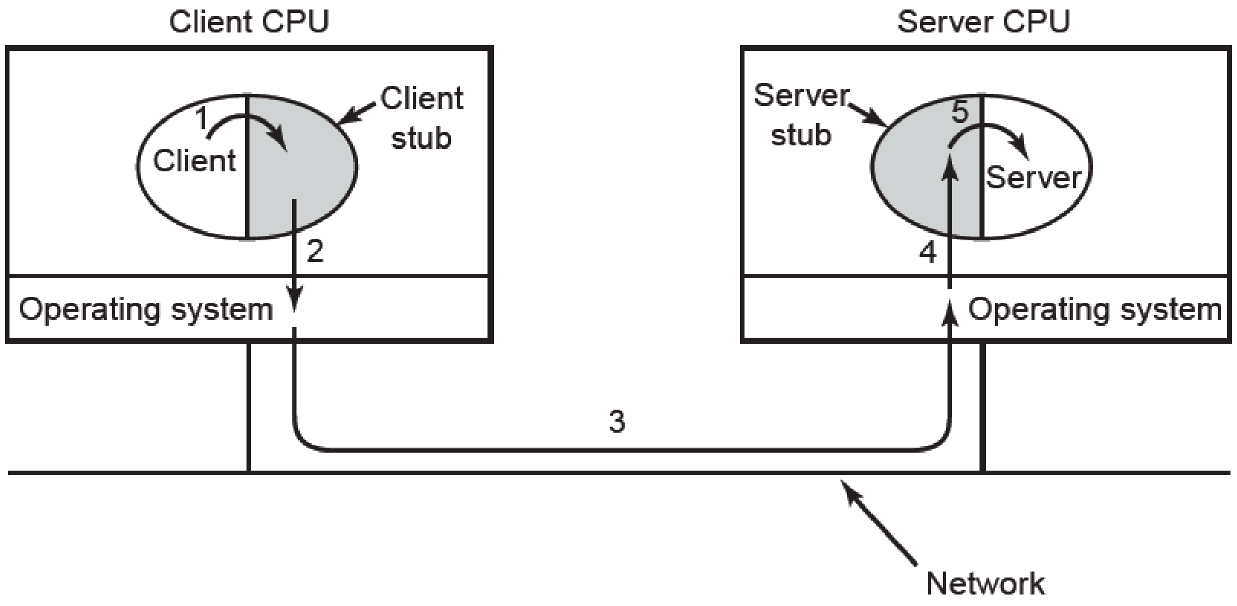
Checksum covers UDP segment and IP pseudoheader

- 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



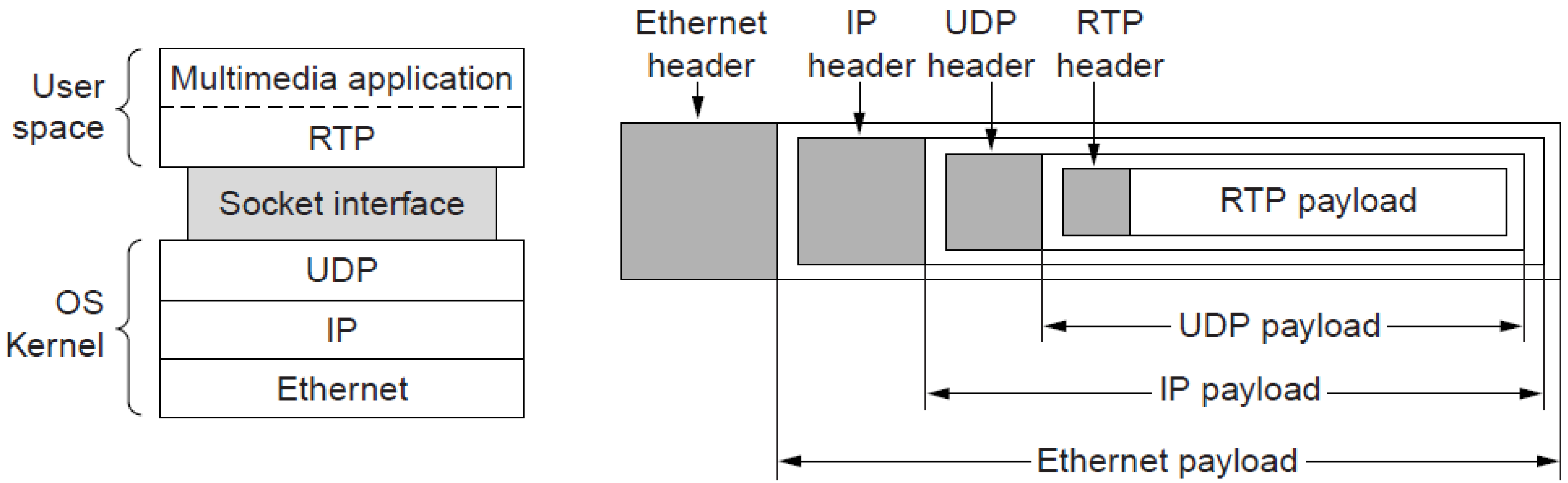
Limitations of RPC

- 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

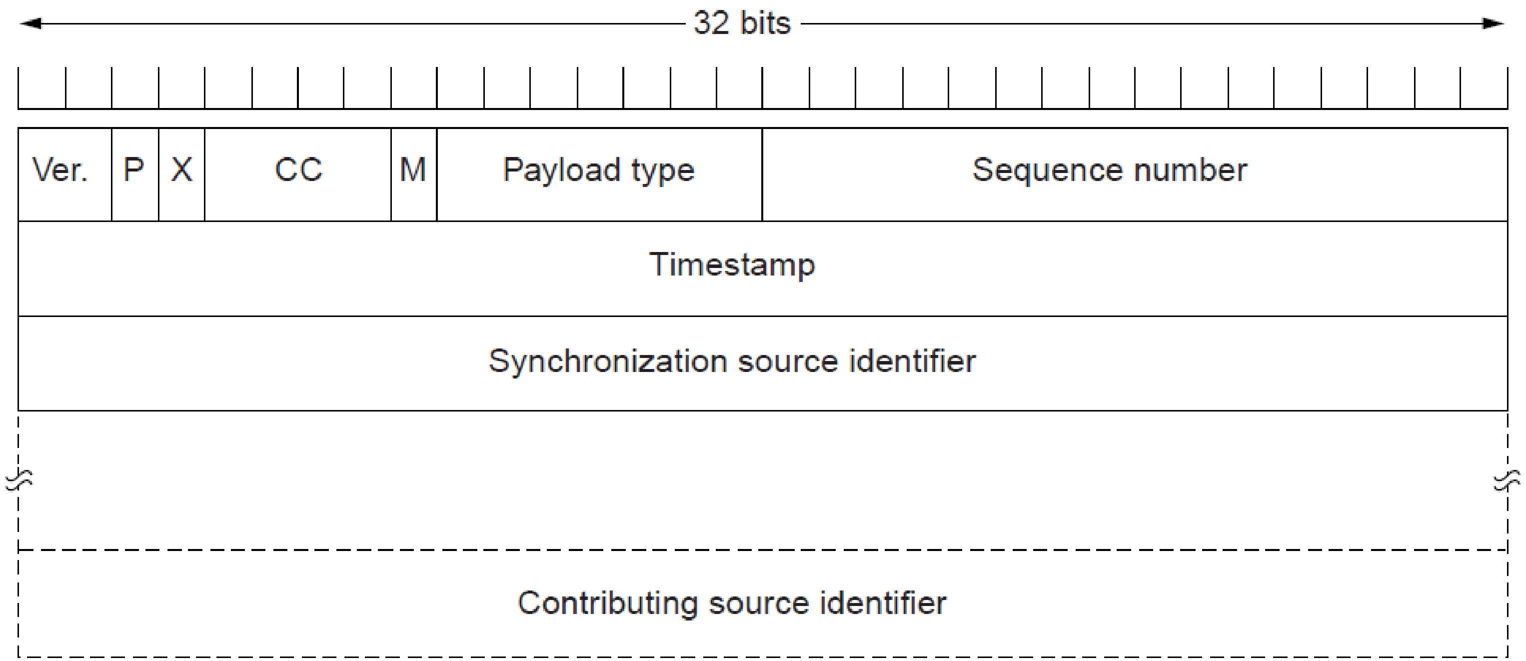
- Often implemented as part of the application



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

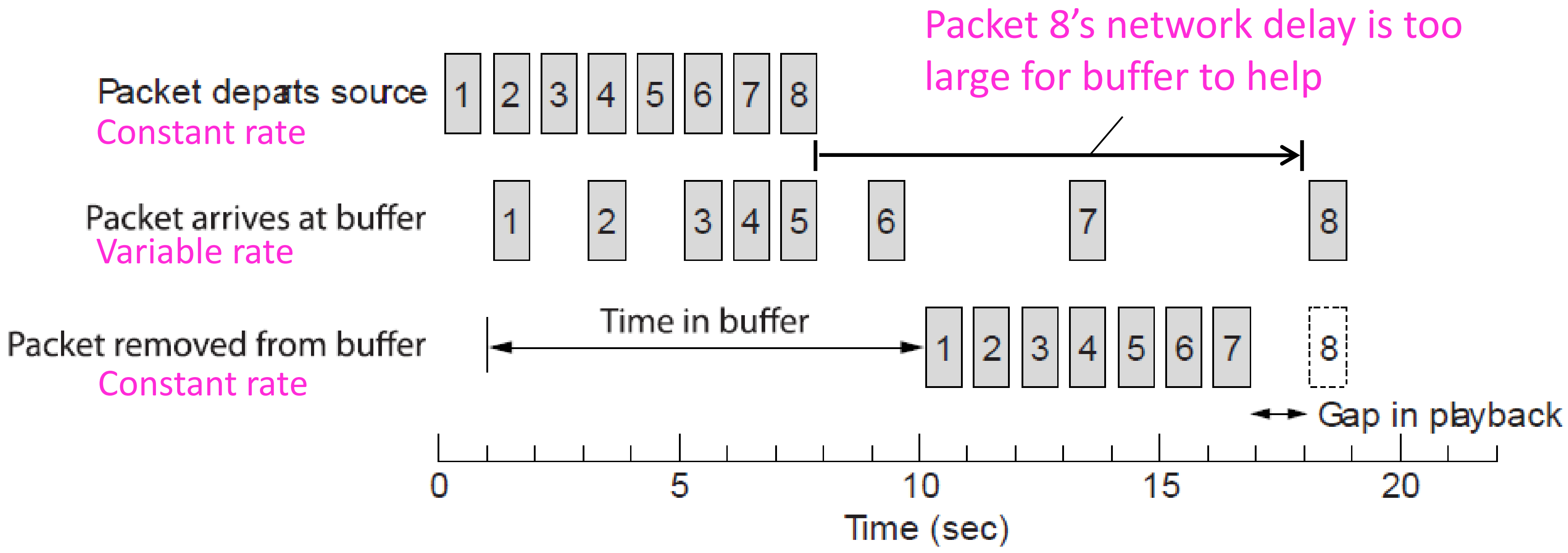
- RTP payloads may contain multiple samples coded in any way the application wants
- Profiles – to support interworking
  - Single Audio Stream
  - Multiple encoding formats may be supported
    - 8-bit pcm samples at 8KHz
    - Delta encoding
    - Predictive encoding
    - MP3
    - …

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