Introduction to Computer Network

Chapter 4 : End-to-End Protocols
Transport Layer

Textbooks

- **Computer Networks: A Systems Approach, 5e , Larry L. Peterson and Bruce S. Davie :: Chapter 7**
- Computer Networking: A Top-Down Approach 6e, J.F. Kurose and K.W. Ross :: Chapter 3
- Computer Networks, Sixth Edition, Andrew S. Tanenbaum :: Chapter 5

Problem

- How to turn this host-to-host packet delivery service into a process-to-process communication channel

Chapter Outline

- Simple Demultiplexer (UDP)
- Reliable Byte Stream (TCP)

Chapter Goal

- Understanding the demultipexing service
- Discussing simple byte stream protocol

End-to-end Protocols

- Common properties that a transport protocol can be expected to provide
  - Guarantees message delivery
  - Delivers messages in the same order they were sent
  - Delivers at most one copy of each message
  - Supports arbitrarily large messages
  - Supports synchronization between the sender and the receiver
  - Allows the receiver to apply flow control to the sender
  - Supports multiple application processes on each host

End-to-end Protocols

- Typical limitations of the network on which transport protocol will operate
  - Drop messages
  - Reorder messages
  - Deliver duplicate copies of a given message
  - Limit messages to some finite size
  - Deliver messages after an arbitrarily long delay

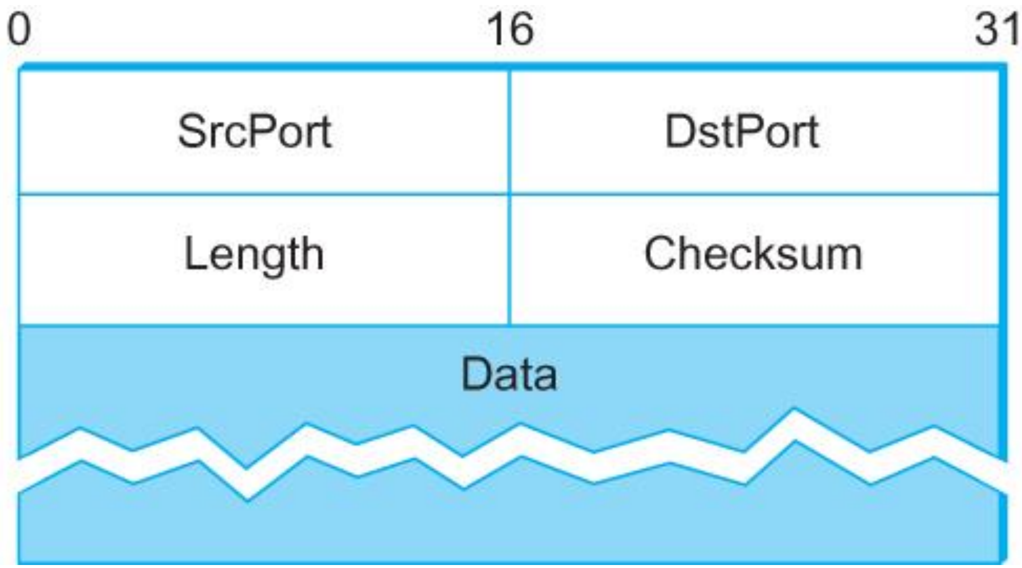
End-to-end Protocols

- Challenge for Transport Protocols
  - Develop algorithms that turn the less-than-desirable properties of the underlying network into the high level of service required by application programs

Simple Demultiplexer (UDP)

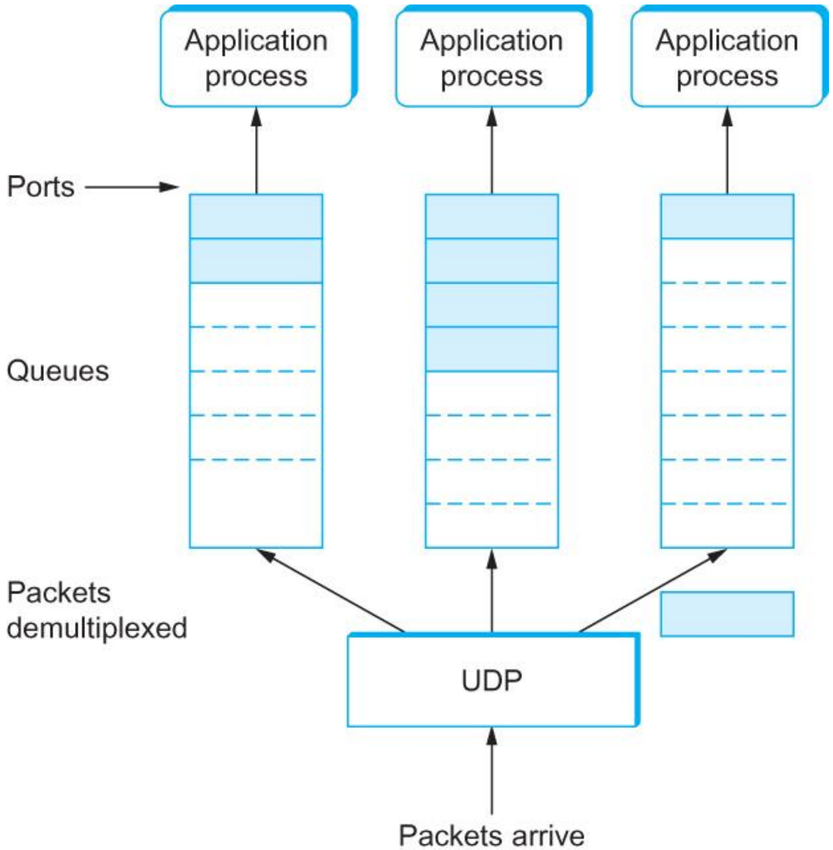
- Extends host-to-host delivery service of the underlying network into a process-to-process communication service
- Adds a level of demultiplexing which allows multiple application processes on each host to share the network

Simple Demultiplexer (UDP)



Format for UDP header (Note: length and checksum fields should be switched)

Simple Demultiplexer (UDP)



UDP Message Queue

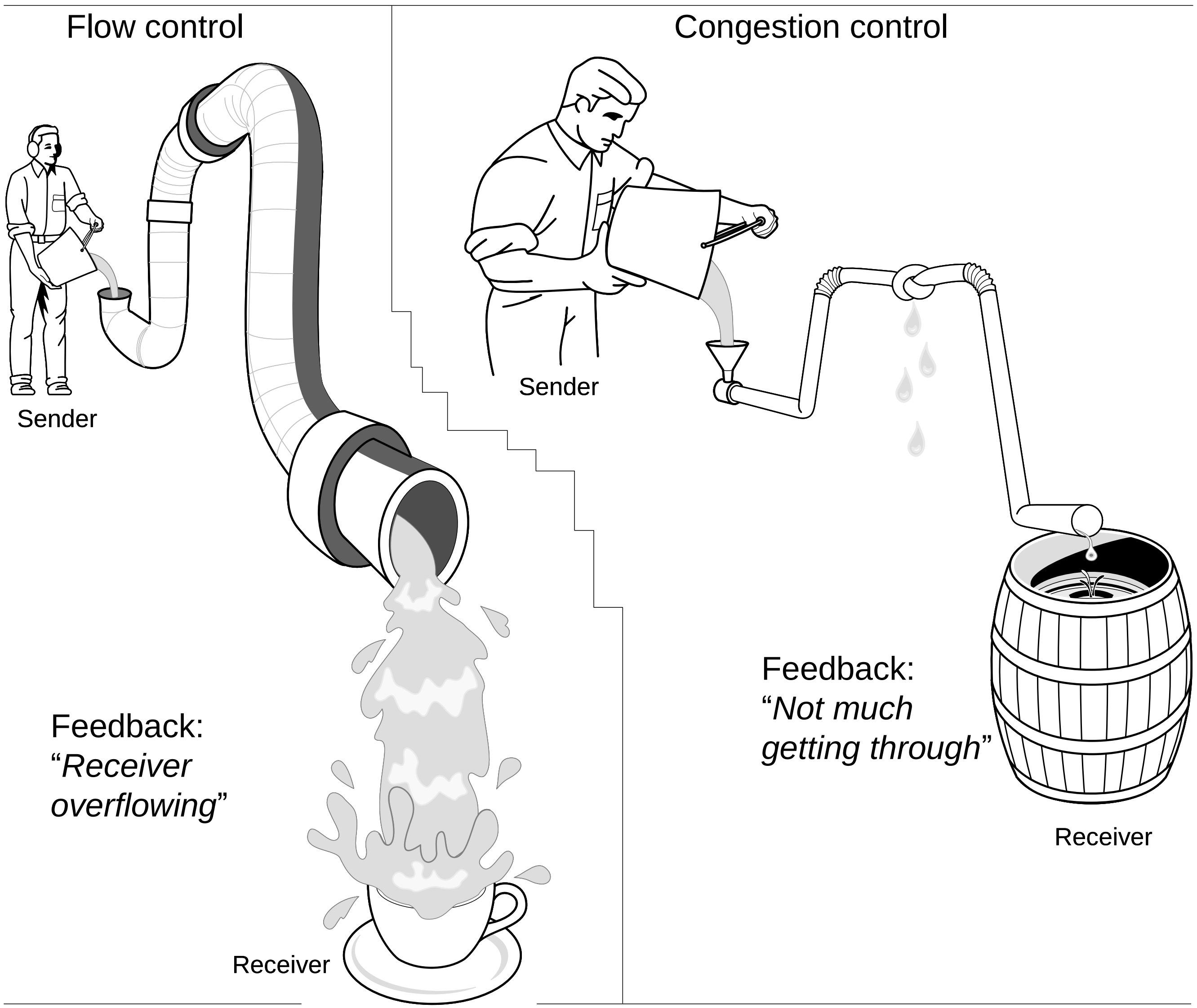
Reliable Byte Stream (TCP)

- In contrast to UDP, Transmission Control Protocol (TCP) offers the following services
  - Reliable
  - Connection oriented
  - Byte-stream service

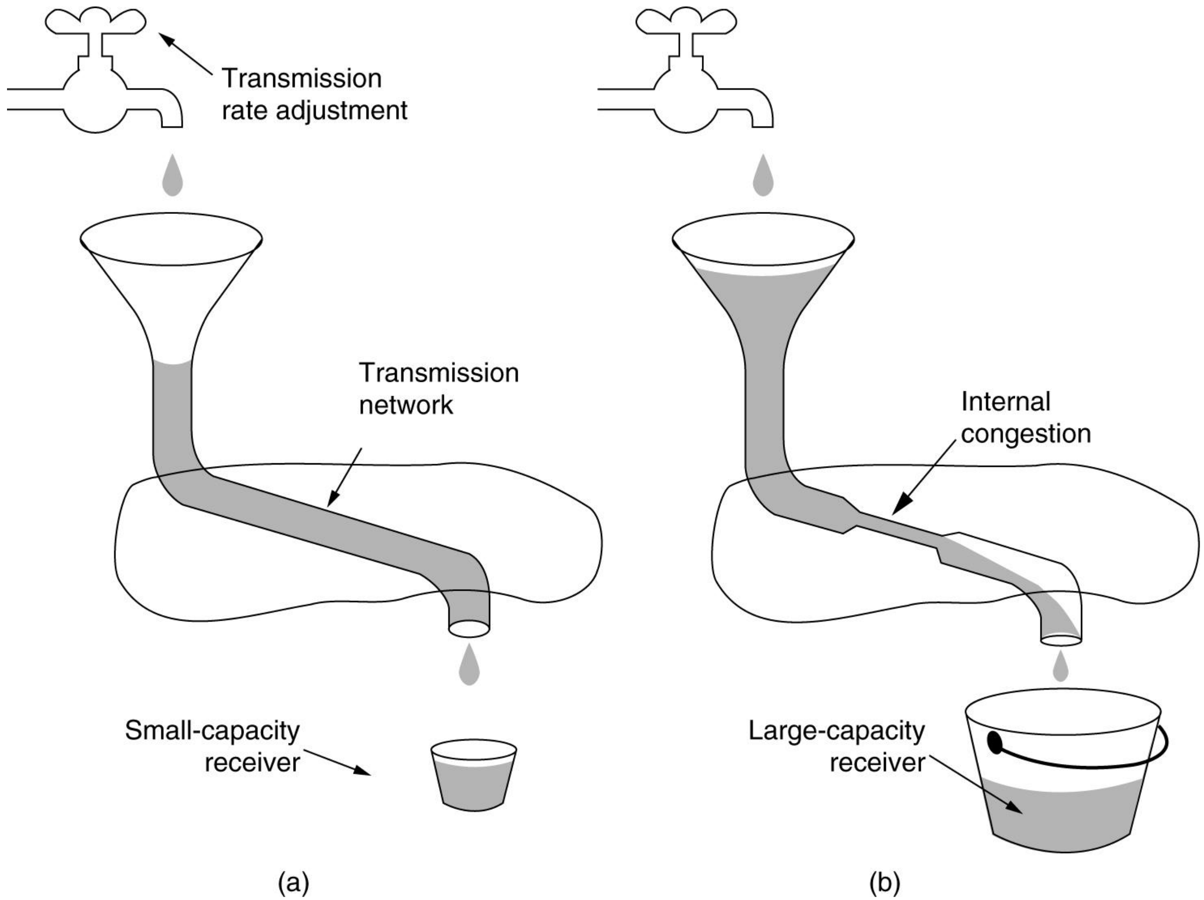
Flow vs. Congestion control

- Flow control involves preventing senders from overrunning the capacity of the receivers
- Congestion control involves preventing too much data from being injected into the network, thereby causing switches or links to become overloaded

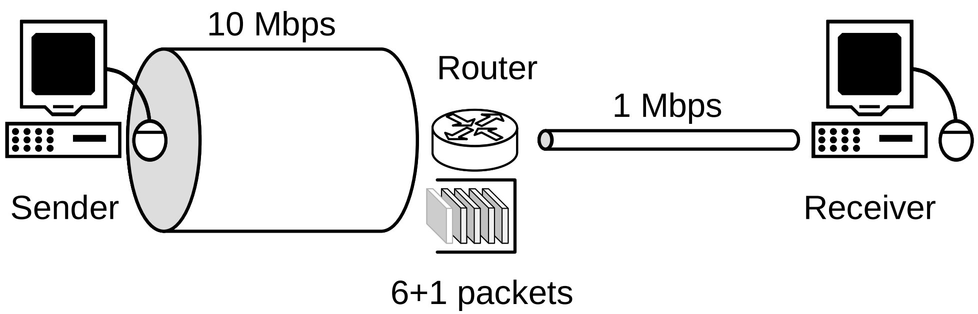
Flow vs. Congestion Control

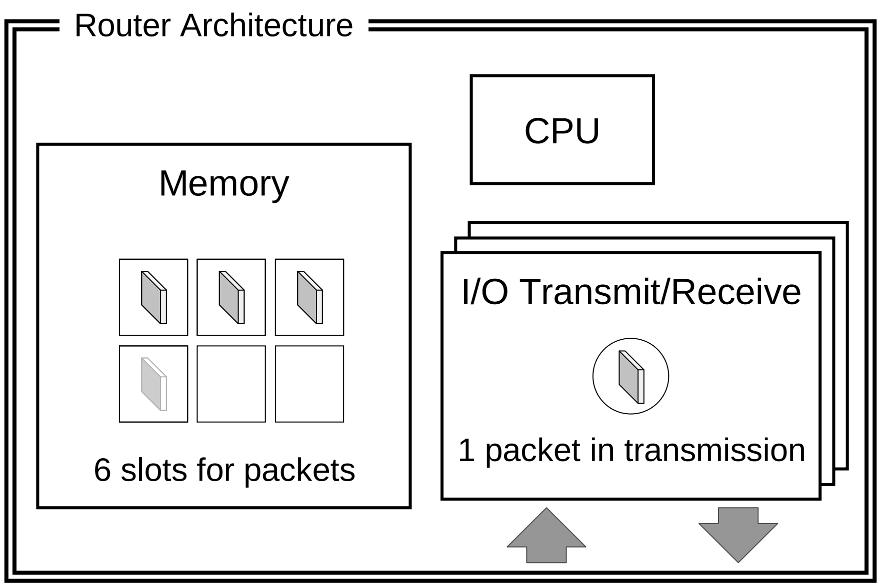


Flow vs. Congestion Control



TCP Bottleneck Link





End-to-end Issues

- At the heart of TCP is the sliding window algorithm
- As TCP runs over the Internet rather than a point-to-point link, the following issues need to be addressed by the sliding window algorithm
  - TCP supports logical connections between processes that are running on two different computers in the Internet
  - TCP connections are likely to have widely different RTT times
  - Packets may get reordered in the Internet

End-to-end Issues

- TCP needs a mechanism using which each side of a connection will learn what resources the other side is able to apply to the connection
- TCP needs a mechanism using which the sending side will learn the capacity of the network

TCP Segment

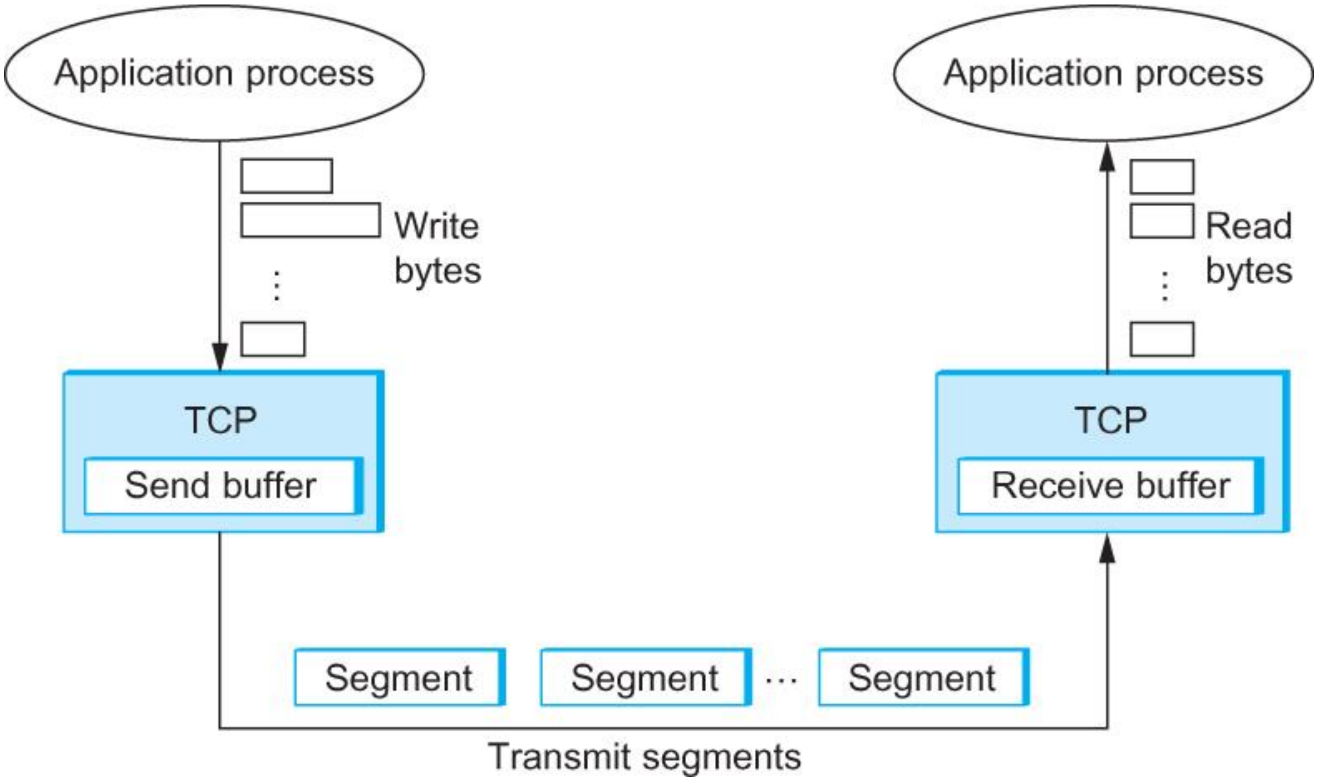
- TCP is a byte-oriented protocol, which means that the sender writes bytes into a TCP connection and the receiver reads bytes out of the TCP connection.

- Although “byte stream” describes the service TCP offers to application processes, TCP does not, itself, transmit individual bytes over the Internet.

TCP Segment

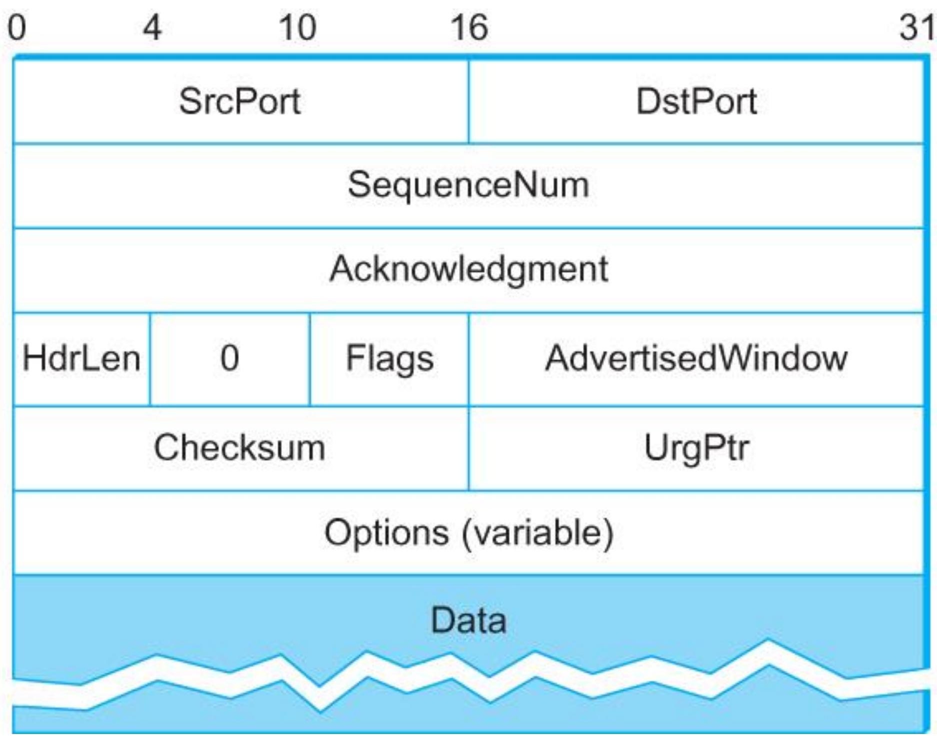
- TCP on the source host buffers enough bytes from the sending process to fill a reasonably sized packet and then sends this packet to its peer on the destination host.
- TCP on the destination host then empties the contents of the packet into a receive buffer, and the receiving process reads from this buffer at its leisure.
- The packets exchanged between TCP peers are called *segments*.

TCP Segment



How TCP manages a byte stream.

TCP Header



TCP Header Format

TCP Header

- The `SrcPort` and `DstPort` fields identify the source and destination ports, respectively.
- The `Acknowledgment`, `SequenceNum`, and `AdvertisedWindow` fields are all involved in TCP's sliding window algorithm.
- Because TCP is a byte-oriented protocol, each byte of data has a sequence number; the `SequenceNum` field contains the sequence number for the first byte of data carried in that segment.
- The `Acknowledgment` and `AdvertisedWindow` fields carry information about the flow of data going in the other direction.

TCP Header

- The 6-bit Flags field is used to relay control information between TCP peers.
- The possible flags include SYN, FIN, RESET, PUSH, URG, and ACK.
- The SYN and FIN flags are used when establishing and terminating a TCP connection, respectively.
- The ACK flag is set any time the Acknowledgment field is valid, implying that the receiver should pay attention to it.

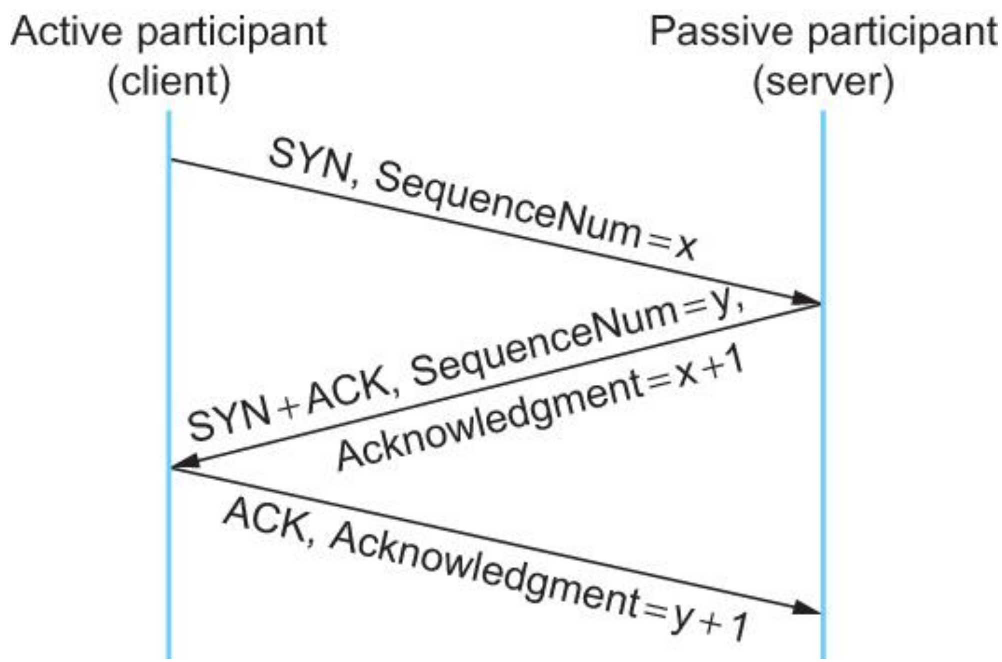
TCP Header

- The URG flag signifies that this segment contains urgent data. When this flag is set, the UrgPtr field indicates where the nonurgent data contained in this segment begins.
- The urgent data is contained at the front of the segment body, up to and including a value of UrgPtr bytes into the segment.
- The PUSH flag signifies that the sender invoked the push operation, which indicates to the receiving side of TCP that it should notify the receiving process of this fact.
- Finally, the RESET flag signifies that the receiver has become confused

TCP Header

- Finally, the RESET flag signifies that the receiver has become confused, it received a segment it did not expect to receive—and so wants to abort the connection.
- Finally, the Checksum field is used in exactly the same way as for UDP—it is computed over the TCP header, the TCP data, and the pseudoheader, which is made up of the source address, destination address, and length fields from the IP header.

Connection Establishment/Termination in TCP

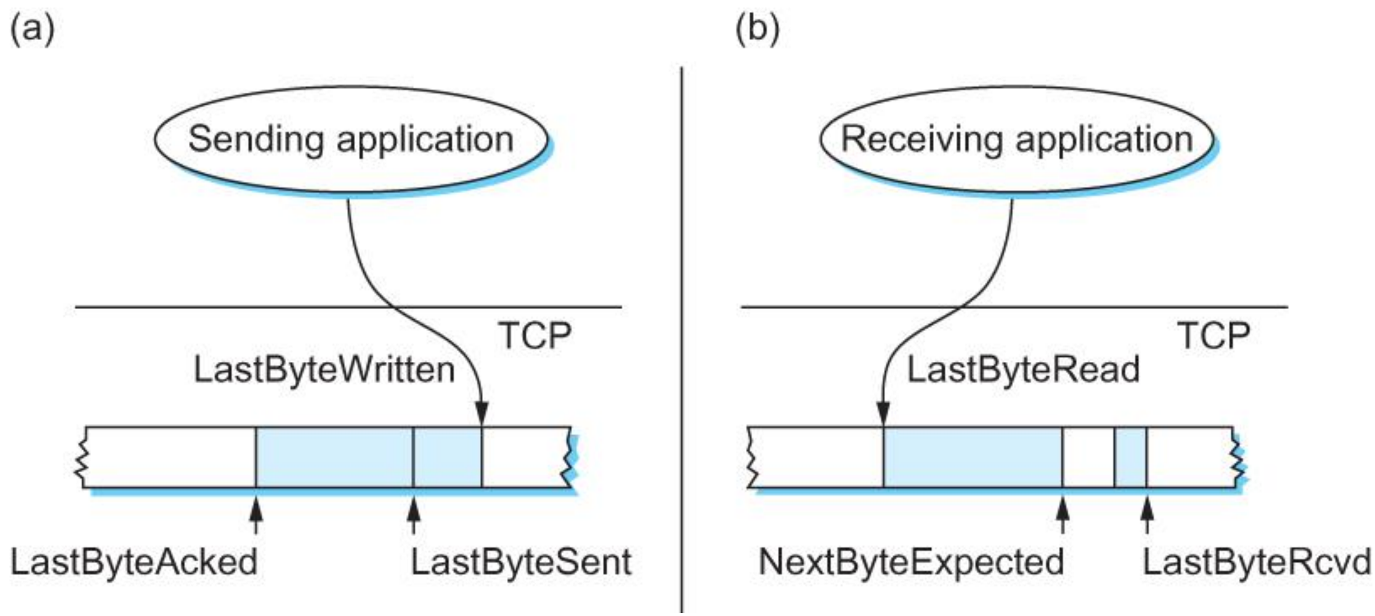


Timeline for three-way handshake algorithm

Sliding Window Revisited

- TCP’s variant of the sliding window algorithm, which serves several purposes:
  - (1) it guarantees the reliable delivery of data,
  - (2) it ensures that data is delivered in order, and
  - (3) it enforces flow control between the sender and the receiver.

Sliding Window Revisited



Relationship between TCP send buffer (a) and receive buffer (b).

TCP Sliding Window

- Sending Side
  - `LastByteAcked ≤ LastByteSent`
  - `LastByteSent ≤ LastByteWritten`
- Receiving Side
  - `LastByteRead < NextByteExpected`
  - `NextByteExpected ≤ LastByteRcvd + 1`

TCP Flow Control

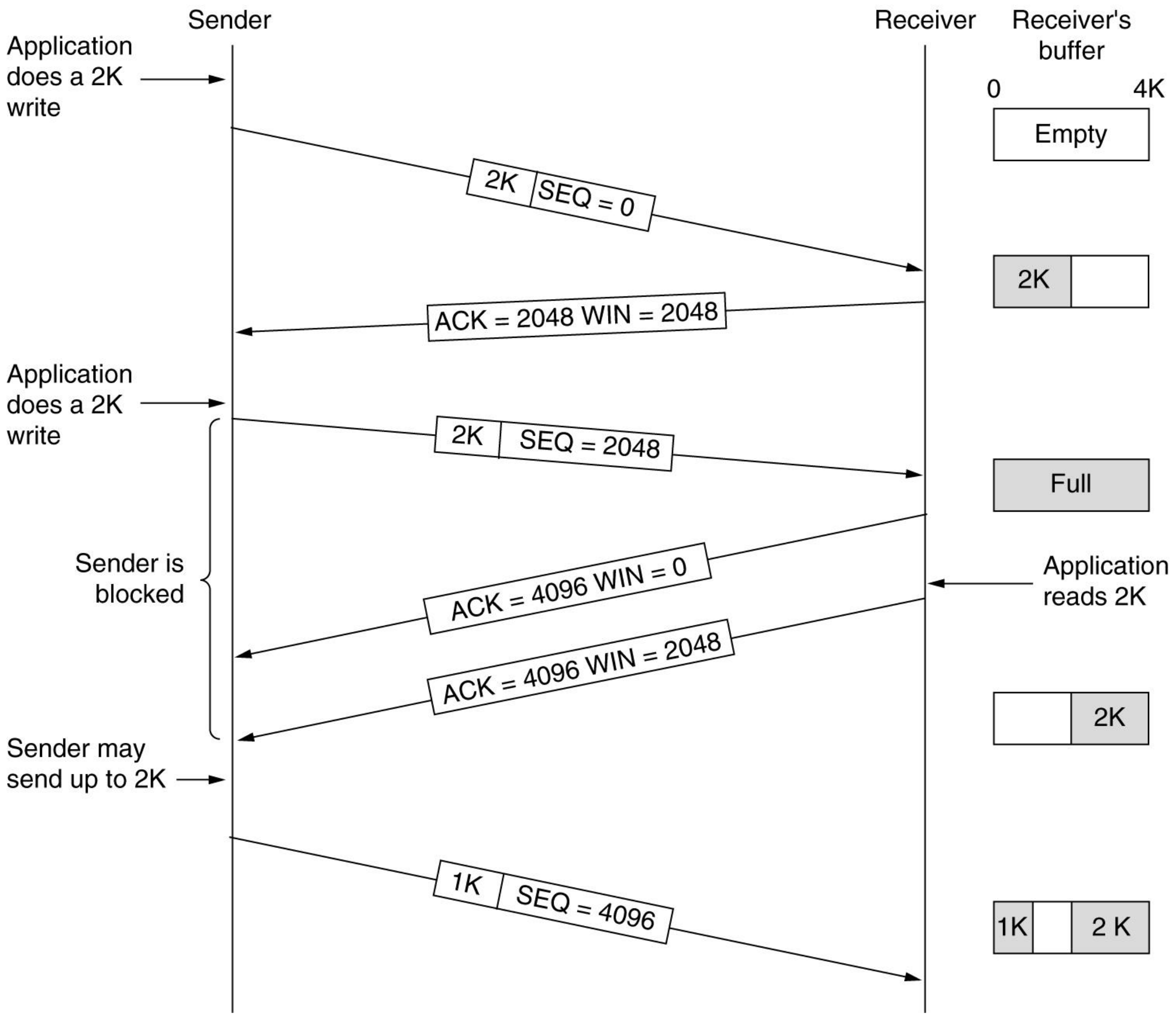
- LastByteRcvd – LastByteRead ≤ MaxRcvBuffer
- AdvertisedWindow = MaxRcvBuffer – ((NextByteExpected – 1) – LastByteRead)
- LastByteSent – LastByteAcked ≤ AdvertisedWindow
- EffectiveWindow = AdvertisedWindow – (LastByteSent – LastByteAcked)
- LastByteWritten – LastByteAcked ≤ MaxSendBuffer

If the sending process tries to write y bytes to TCP, but

- (LastByteWritten – LastByteAcked) + y > MaxSendBuffer

then TCP blocks the sending process and does not allow it to generate more data.

TCP Transmission Policy



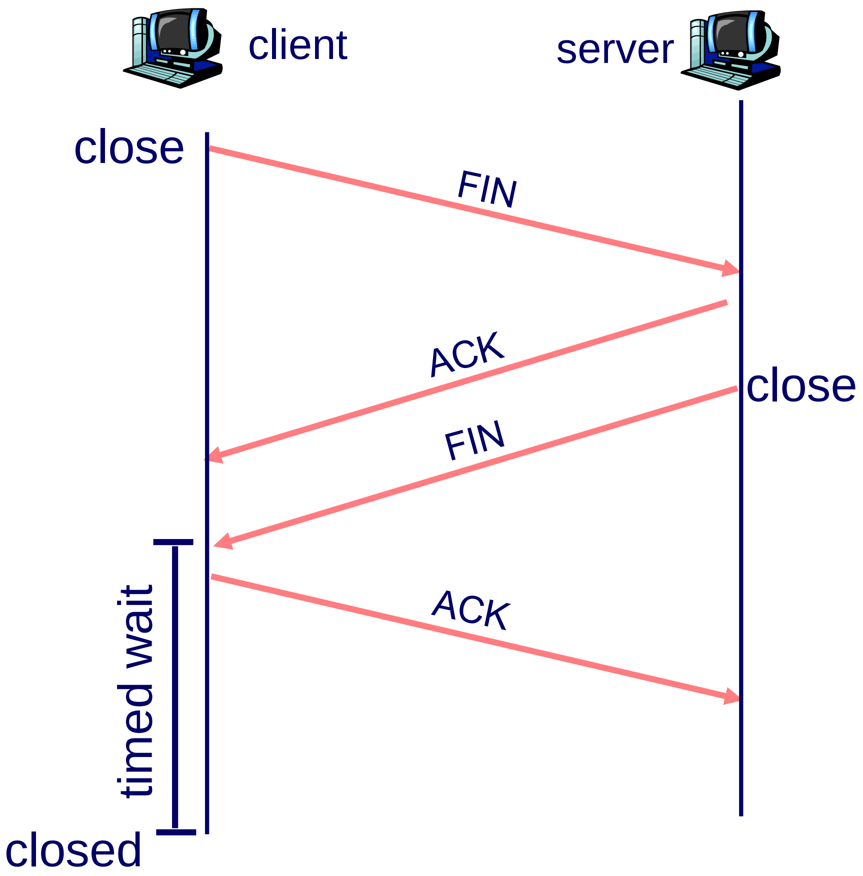
- Window management in TCP.

Closing a TCP Connection

client closes socket:
`clientSocket.close();`

Step 1: client end system sends TCP FIN control segment to server

Step 2: server receives FIN, replies with ACK. Closes connection, sends FIN.



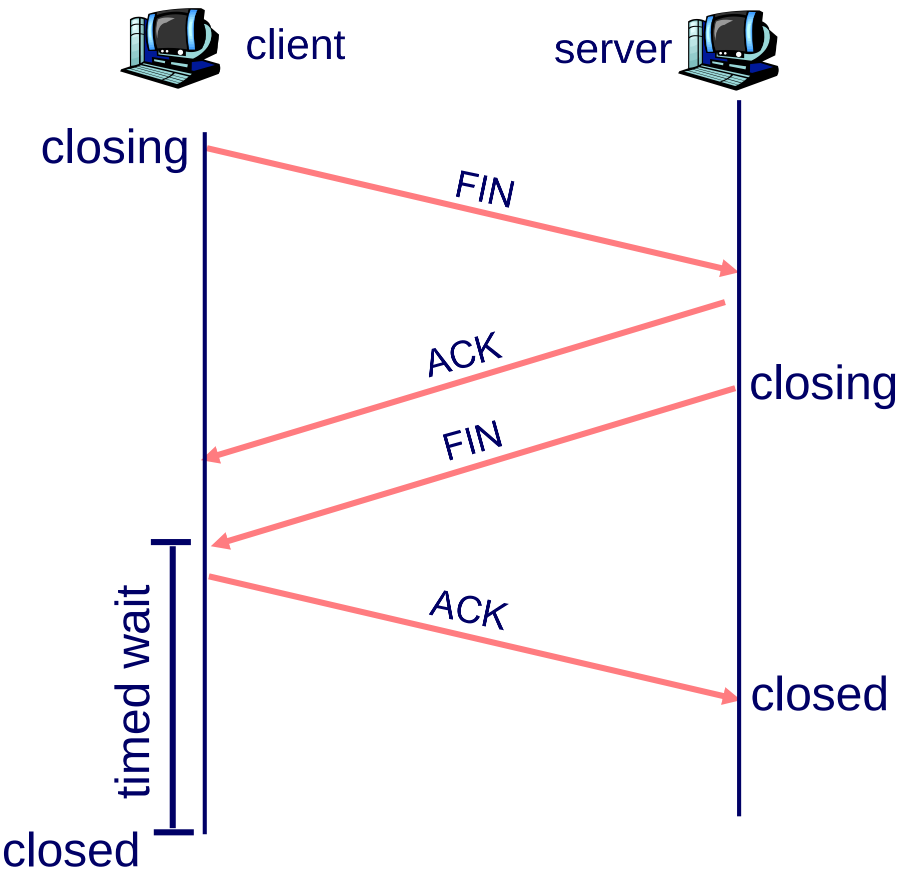
Closing a TCP Connection

Step 3: client receives FIN, replies with ACK.

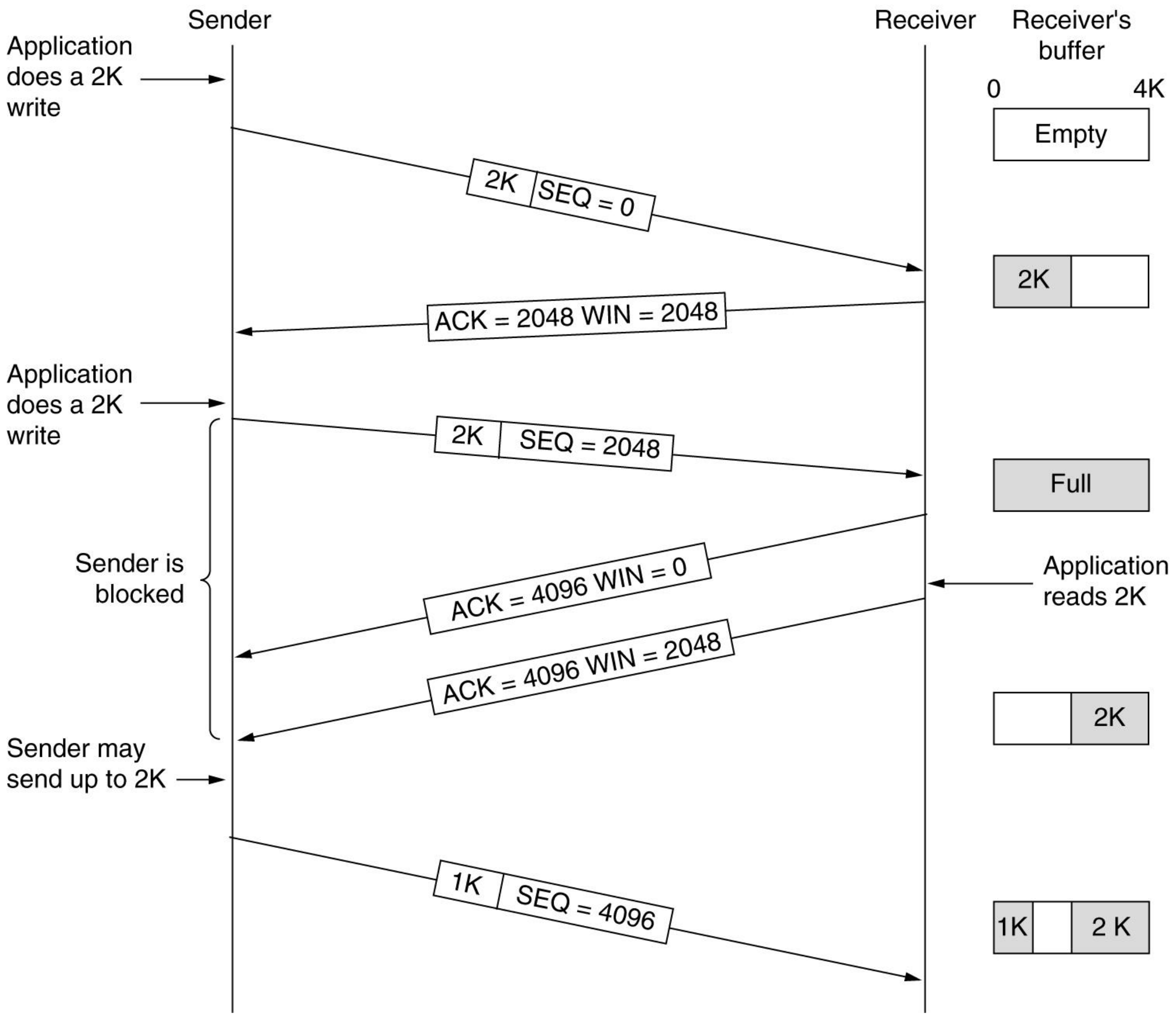
- Enters "timed wait" - will respond with ACK to received FINs

Step 4: server, receives ACK. Connection closed.

Note: with small modification, can handle simultaneous FINs.



Transmission Policy



Protecting against Wraparound

- SequenceNum: 32 bits longs
- AdvertisedWindow: 16 bits long
  - TCP has satisfied the requirement of the sliding
  - window algorithm that is the sequence number
  - space be twice as big as the window size
  - $2^{32} >> 2 \times 2^{16}$

Protecting against Wraparound

- Relevance of the 32-bit sequence number space
  - The sequence number used on a given connection might wraparound
  - A byte with sequence number *x* could be sent at one time, and then at a later time a second byte with the same sequence number *x* could be sent
  - Packets cannot survive in the Internet for longer than the **MSL**
  - **MSL** is set to 120 sec
  - We need to make sure that the sequence number does not wrap around within a 120-second period of time
  - Depends on how fast data can be transmitted over the Internet

Protecting against Wraparound

Bandwidth	Time until Wraparound
T1 (1.5 Mbps)	6.4 hours
Ethernet (10 Mbps)	57 minutes
T3 (45 Mbps)	13 minutes
Fast Ethernet (100 Mbps)	6 minutes
OC-3 (155 Mbps)	4 minutes
OC-12 (622 Mbps)	55 seconds
OC-48 (2.5 Gbps)	14 seconds

Time until 32-bit sequence number space wraps around.

Keeping the Pipe Full

- 16-bit AdvertisedWindow field must be big enough to allow the sender to keep the pipe full
- Clearly the receiver is free not to open the window as large as the AdvertisedWindow field allows
- If the receiver has enough buffer space
  - The window needs to be opened far enough to allow a full
  - delay × bandwidth product's worth of data
  - Assuming an RTT of 100 ms

Keeping the Pipe Full

Bandwidth	Delay × Bandwidth Product
T1 (1.5 Mbps)	18 KB
Ethernet (10 Mbps)	122 KB
T3 (45 Mbps)	549 KB
Fast Ethernet (100 Mbps)	1.2 MB
OC-3 (155 Mbps)	1.8 MB
OC-12 (622 Mbps)	7.4 MB
OC-48 (2.5 Gbps)	29.6 MB

Required window size for 100-ms RTT.

Triggering Transmission

- How does TCP decide to transmit a segment?
  - TCP supports a byte stream abstraction
  - Application programs write bytes into streams
  - It is up to TCP to decide that it has enough bytes to send a segment

Triggering Transmission

- What factors governs this decision
  - Ignore flow control: window is wide open, as would be the case when the connection starts
  - TCP has three mechanism to trigger the transmission of a segment
    - 1) TCP maintains a variable MSS and sends a segment as soon as it has collected MSS bytes from the sending process
      - MSS is usually set to the size of the largest segment TCP can send without causing local IP to fragment.
      - MSS: `MTU of directly connected network – (TCP header + and IP header)`
    - 2) Sending process has explicitly asked TCP to send it
      - TCP supports push operation
    - 3) When a timer fires
      - Resulting segment contains as many bytes as are currently buffered for transmission

Silly Window Syndrome

- If you think of a TCP stream as a conveyer belt with "full" containers (data segments) going in one direction and empty containers (ACKs) going in the reverse direction, then MSS-sized segments correspond to large containers and 1-byte segments correspond to very small containers.
- If the sender aggressively fills an empty container as soon as it arrives, then any small container introduced into the system remains in the system indefinitely.
- That is, it is immediately filled and emptied at each end, and never coalesced with adjacent containers to create larger containers.

Silly Window Syndrome

Silly Window Syndrome

Nagle's Algorithm

- If there is data to send but the window is open less than MSS, then we may want to wait some amount of time before sending the available data
- But how long?
- If we wait too long, then we hurt interactive applications like Telnet
- If we don't wait long enough, then we risk sending a bunch of tiny packets and falling into the *silly window* syndrome
  - The solution is to introduce a timer and to transmit when the timer expires

Nagle's Algorithm

- We could use a clock-based timer, for example one that fires every 100 ms
- Nagle introduced an elegant self-clocking solution
- Key Idea
  - As long as TCP has any data in flight, the sender will eventually receive an ACK
  - This ACK can be treated like a timer firing, triggering the transmission of more data

Nagle's Algorithm

```
When the application produces data to send
    if both the available data and the window ≥ MSS
        send a full segment
    else
        if there is unACKed data in flight
                buffer the new data until an ACK arrives
        else
                send all the new data now
```

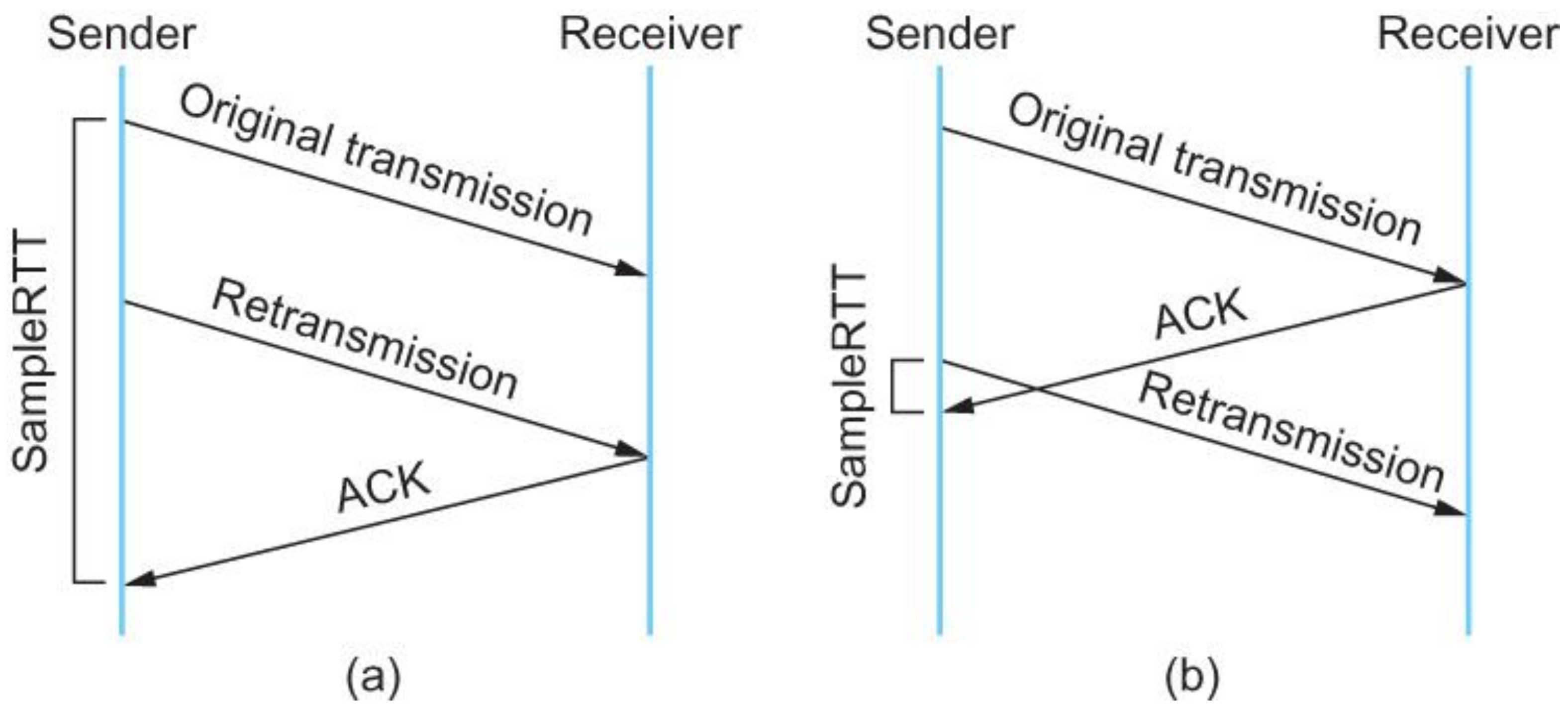
Adaptive Retransmission

- Original Algorithm
  - Measure **`SampleRTT`** for each segment/ ACK pair
  - Compute weighted average of RTT
    - **`EstRTT`** = α x **`EstRTT`** + (1 - α)x **`SampleRTT`**
      - α between 0.8 and 0.9
  - Set timeout based on **`EstRTT`**
    - **`TimeOut`** = 2 x **`EstRTT`**

Original Algorithm

- Problem
  - ACK does not really acknowledge a transmission
    - It actually acknowledges the receipt of data
  - When a segment is retransmitted and then an ACK arrives at the sender
    - It is impossible to decide if this ACK should be associated with the first or the second transmission for calculating RTTs

Karn/Partridge Algorithm



Associating the ACK with (a) original transmission versus (b) retransmission

Karn/Partridge Algorithm

- Do not sample RTT when retransmitting
- Double timeout after each retransmission

Karn/Partridge Algorithm

- Karn-Partridge algorithm was an improvement over the original approach, but it does not eliminate congestion

- We need to understand how timeout is related to congestion
  - If you timeout too soon, you may unnecessarily retransmit a segment which adds load to the network

Karn/Partridge Algorithm

- Main problem with the original computation is that it does not take variance of Sample RTTs into consideration.
- If the variance among Sample RTTs is small
  - Then the Estimated RTT can be better trusted
  - There is no need to multiply this by 2 to compute the timeout

Karn/Partridge Algorithm

- On the other hand, a large variance in the samples suggest that timeout value should not be tightly coupled to the Estimated RTT
- Jacobson/Karels proposed a new scheme for TCP retransmission

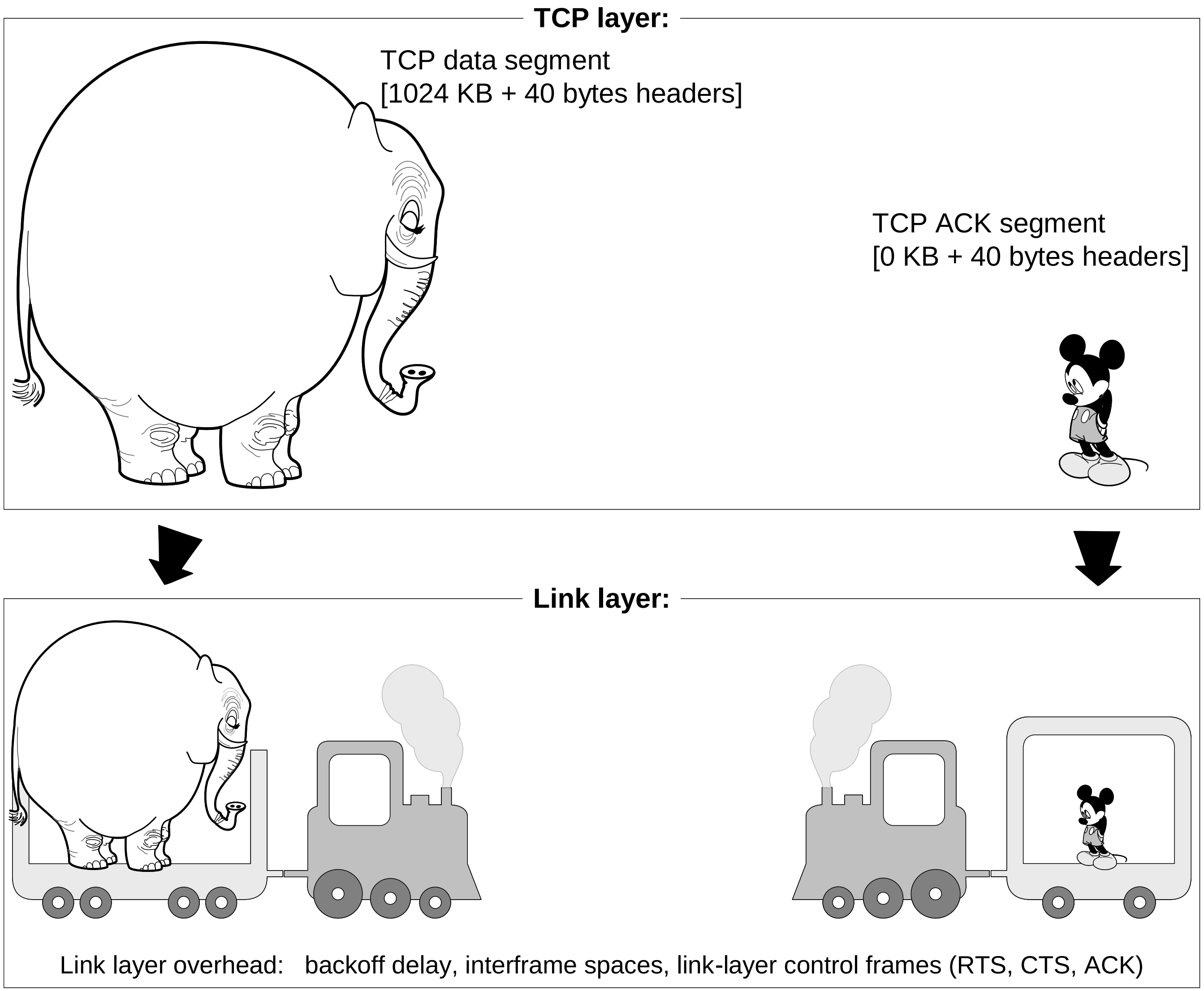
Topic:
TCP Over Wireless

- Reducing Link-Layer Overheads

TCP Over Wireless

- Slow start until reaches the SSThresh size;
  set timer for every burst
- If no ACK $\Rightarrow$ congestion;
  halve the window size; slow start again

Wireless Link-layer Overhead



Jacobson/Karels Algorithm

- Difference = SampleRTT − EstimatedRTT
- EstimatedRTT = EstimatedRTT + (× Difference)
- Deviation = Deviation + (|Difference| − Deviation)
- TimeOut = μ × EstimatedRTT + × Deviation
  - where based on experience, μ is typically set to 1 and is set to 4. Thus, when the variance is small, TimeOut is close to EstimatedRTT; a large variance causes the deviation term to dominate the calculation.

Slow Start

Initial CW = 1.

After each ACK, CW += 1;

Continue until:

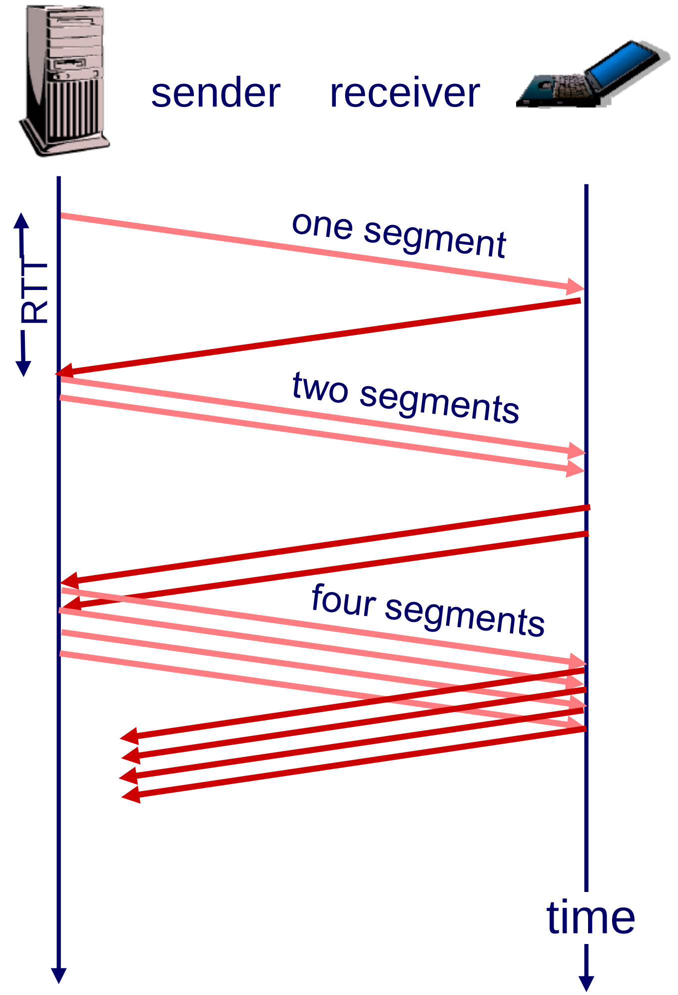
    Loss occurs OR

    CW > slow start threshold

Then switch to congestion avoidance

If we detect loss, cut CW in half

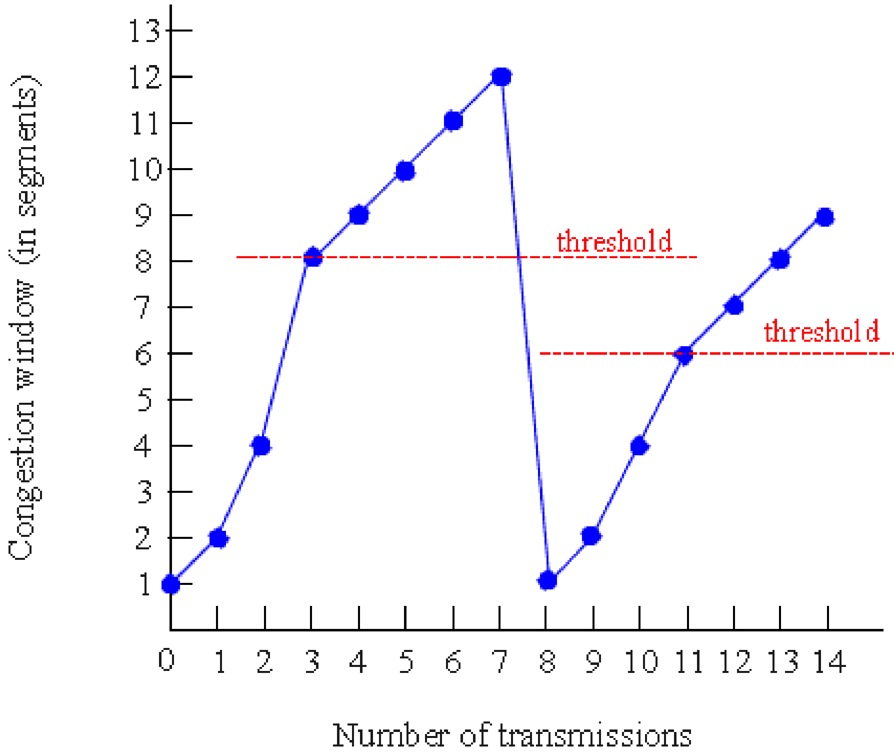
Exponential increase in window size per RTT



Congestion Avoidance

```
Until (loss) {
 after CW packets ACKed:
   CW += 1;
}
ssthresh = CW/2;
Depending on loss type:
  SACK/Fast Retransmit:
    CW/= 2; continue;
  Course grained timeout:
    CW = 1; go to slow start.
```

(This is for TCP Reno/SACK: TCP Tahoe always sets CW=1 after a loss)



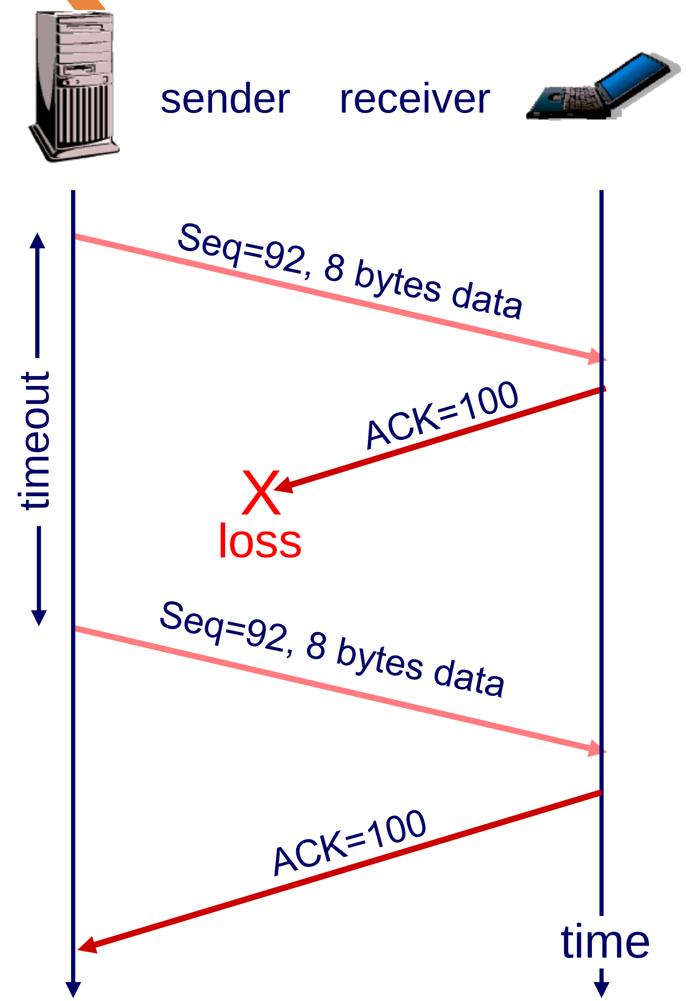
How are losses recovered?

- Say packet is lost (data or ACK!)
- Coarse-grained Timeout:
  - Sender does not receive ACK after some period of time
  - Event is called a retransmission time-out (RTO)
  - RTO value is based on estimated round-trip time (RTT)
  - RTT is adjusted over time using exponential weighted moving average:

    RTT = (1-x)*RTT + (x)*sample

    (x is typically 0.1)

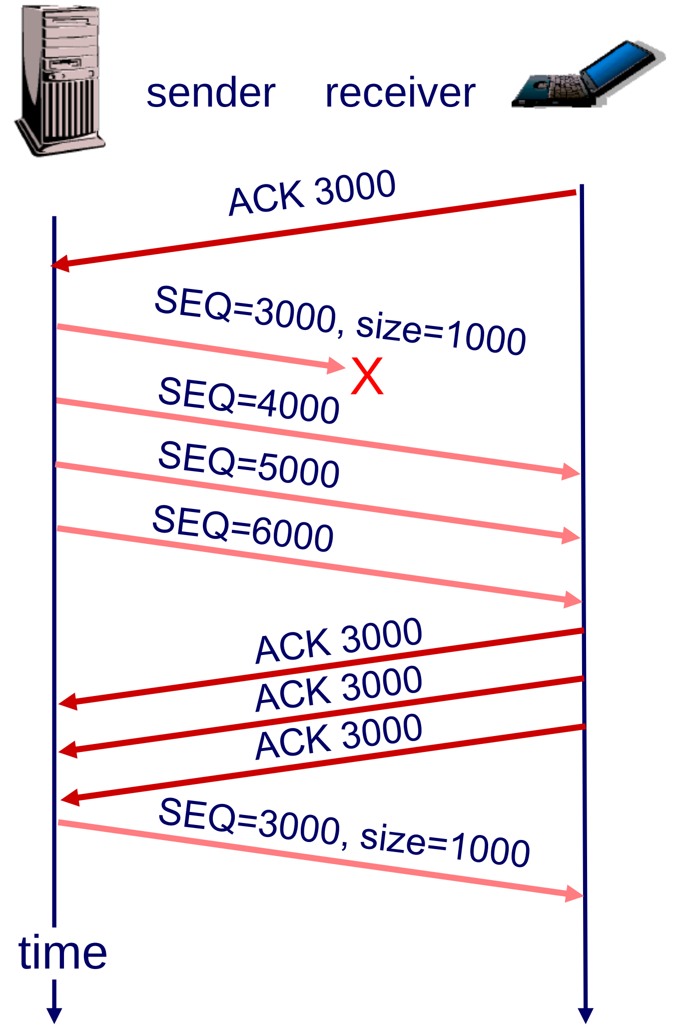
First done in TCP Tahoe



lost ACK scenario

Fast Retransmit

- Receiver expects N, gets N+1:
  - Immediately sends ACK(N)
  - This is called a duplicate ACK
  - Does NOT delay ACKs here!
  - Continue sending dup ACKs for each subsequent packet (not N)
- Sender gets 3 duplicate ACKs:
  - Infers N is lost and resends
  - 3 chosen so out-of-order packets don't trigger Fast Retransmit accidentally
  - Called "fast" since we don't need to wait for a full RTT



TCP Connection Management Modeling

- The states used in the TCP connection management finite state machine.

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
TIMED 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

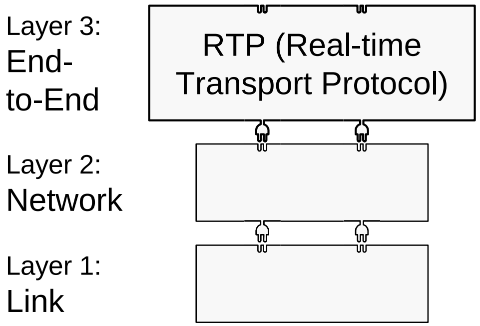
Summary

- We have discussed how to convert host-to-host packet delivery service to process-to-process communication channel.
- We have discussed UDP
- We have discussed TCP

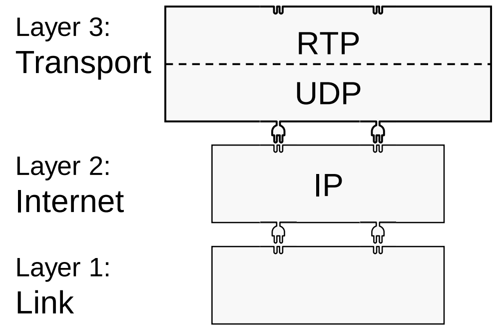
Introduction to Computer Network

More of Chapter 3

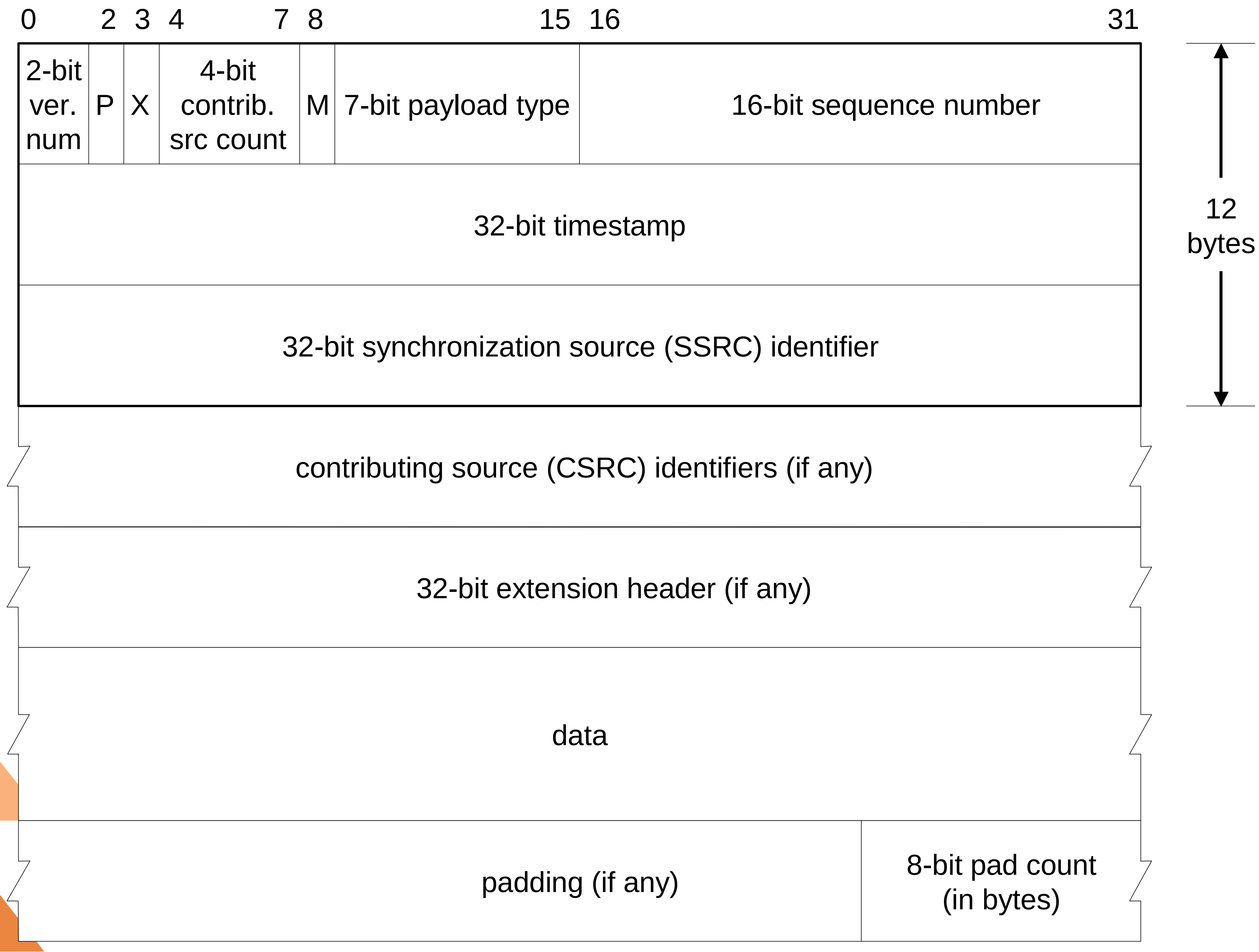
RTP: Real-time Transport Protocol



TCP/IP protocol suite:



RTP Header



0
2
3
4
7
8
15
16
31
2-bit ver. num
P
X
4-bit contrib. src count
M
7-bit payload type
16-bit sequence number
32-bit timestamp
32-bit synchronization source (SSRC) identifier
12 bytes
contributing source (CSRC) identifiers (if any)
32-bit extension header (if any)
data
padding (if any)
8-bit pad count (in bytes)

RTP Header (2)

- P = padding bit indicates if packet is padded to a required size (e.g., for encryption)
- X = extension bit indicates an extension header; rarely used b/c appl. defines own hdr
- CSRC count = # contributing source IDs in the header, if any
- M = marks packets w/ significant events
- Payload type = payload format indicates encoding scheme used for audio, video, etc.

RTP Header (3)

- Sequence number = used by receiver for delay jitter removal
- Timestamp = used with seq. num. to detect pkt loss. Also to synchronize packets from different sources. Represents the sampling (creation) time of the 1st byte. Possible that successive packets have the same timestamp. E.g., a single video frame is transmitted in multiple RTP packets.
- SSRC identifier = unique synchronization source ID randomly chosen. Same for all packets from the same src/device. Enables receiver to group packets for playback.
- CSRC identifiers = list of contributing sources for the packet payload. Used when a mixer combines several streams of packets. Allows the receiver to identify the original senders.

RTCP Header Format

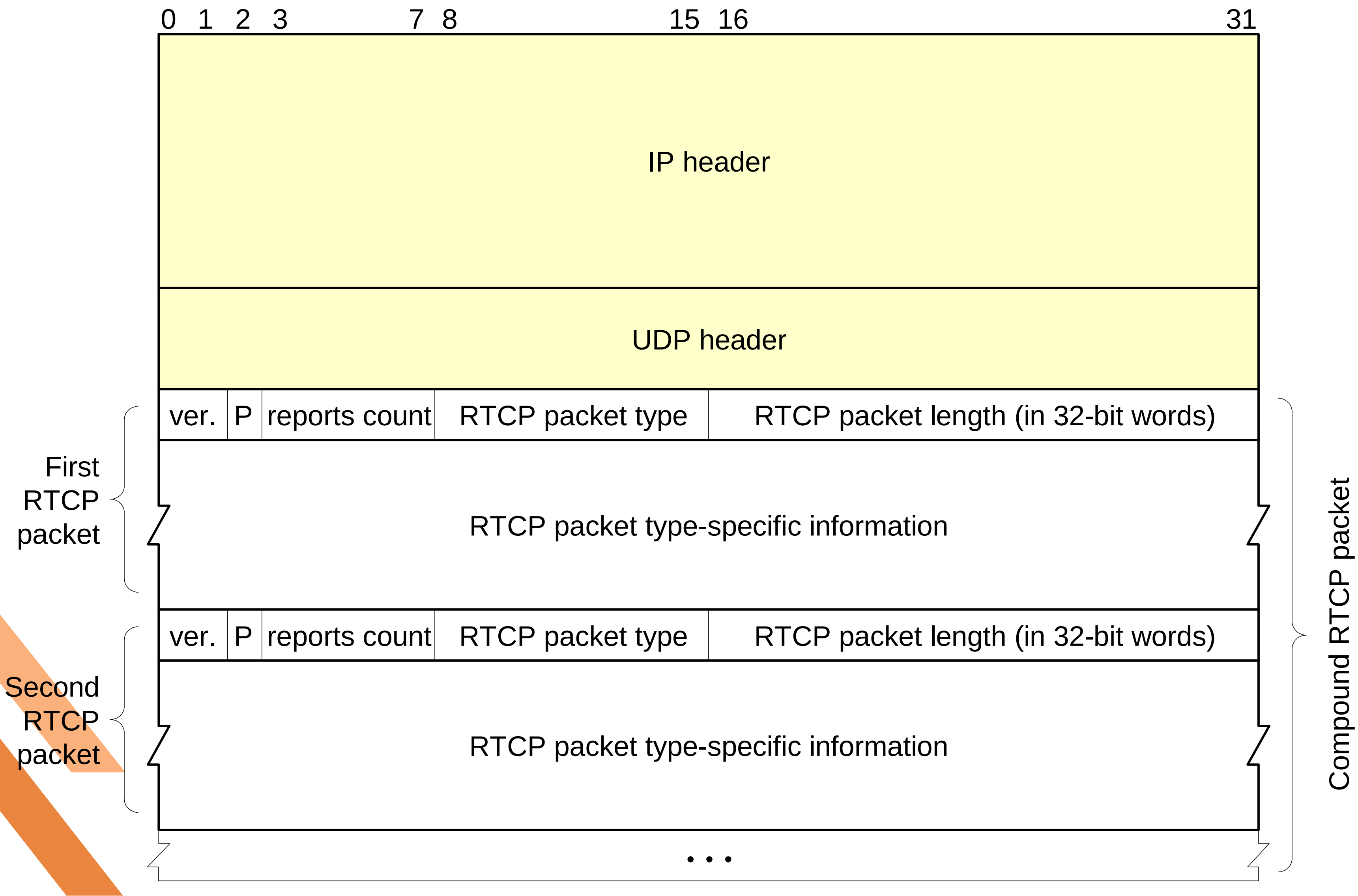
Common sender/receiver REPORT message header:

0 1 2 3
0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1

v=2	*p*	*RR count*	*packet type*	*message length*
SSRC of report sender				

All report messages have the same 8 byte header
» version number (same as RTP)
» padding indicator
» reception report count (5 bits)
» RTCP message type (8 bits)
» RTCP message length (16 bits)
» SSRC for the sender of this report (32 bits)

Compound RTCP Packet Format



Reception Report Blocks

- Each sender and receiver report should contain a reception report block for each synchronization source heard from since the last RTCP report
- Contents:
  - source identifier for the block (SSRC)
  - fraction of RTP packets from this source lost since the last report
  - cumulative number of lost packets
  - extended highest sequence number received
  - estimated average RTP packet interarrival time jitter
  - last SR timestamp received from this source
  - delay since receiving the last SR report from this source

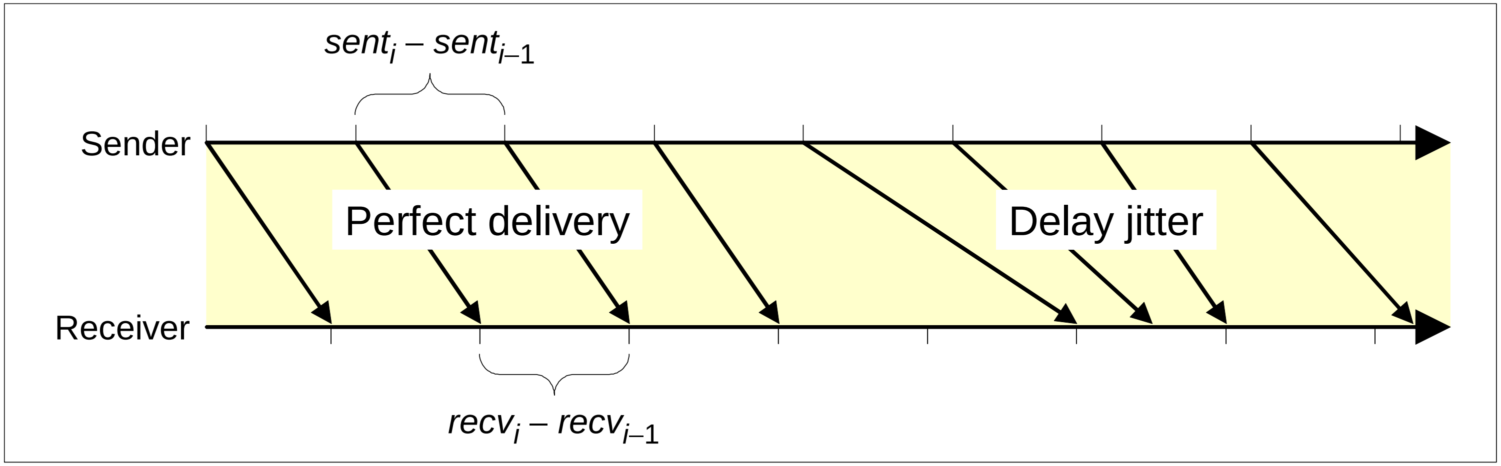
RTCP Receiver Report Format

	0 1	2	3 7	8 15	16 31
header	ver.	P	reports count	packet type = ‘201’	packet length (in 32-bit words)
	SSRC of this reporter				
report block 1	SSRC_1 (SSRC of first source)				
	fraction lost			cumulative number of packets lost	
	extended highest sequence number received				
	interarrival jitter				
	last SR (LSR)				
	delay since last SR (DLSR)				
	report block 2 . . .				
	profile-specific extensions				

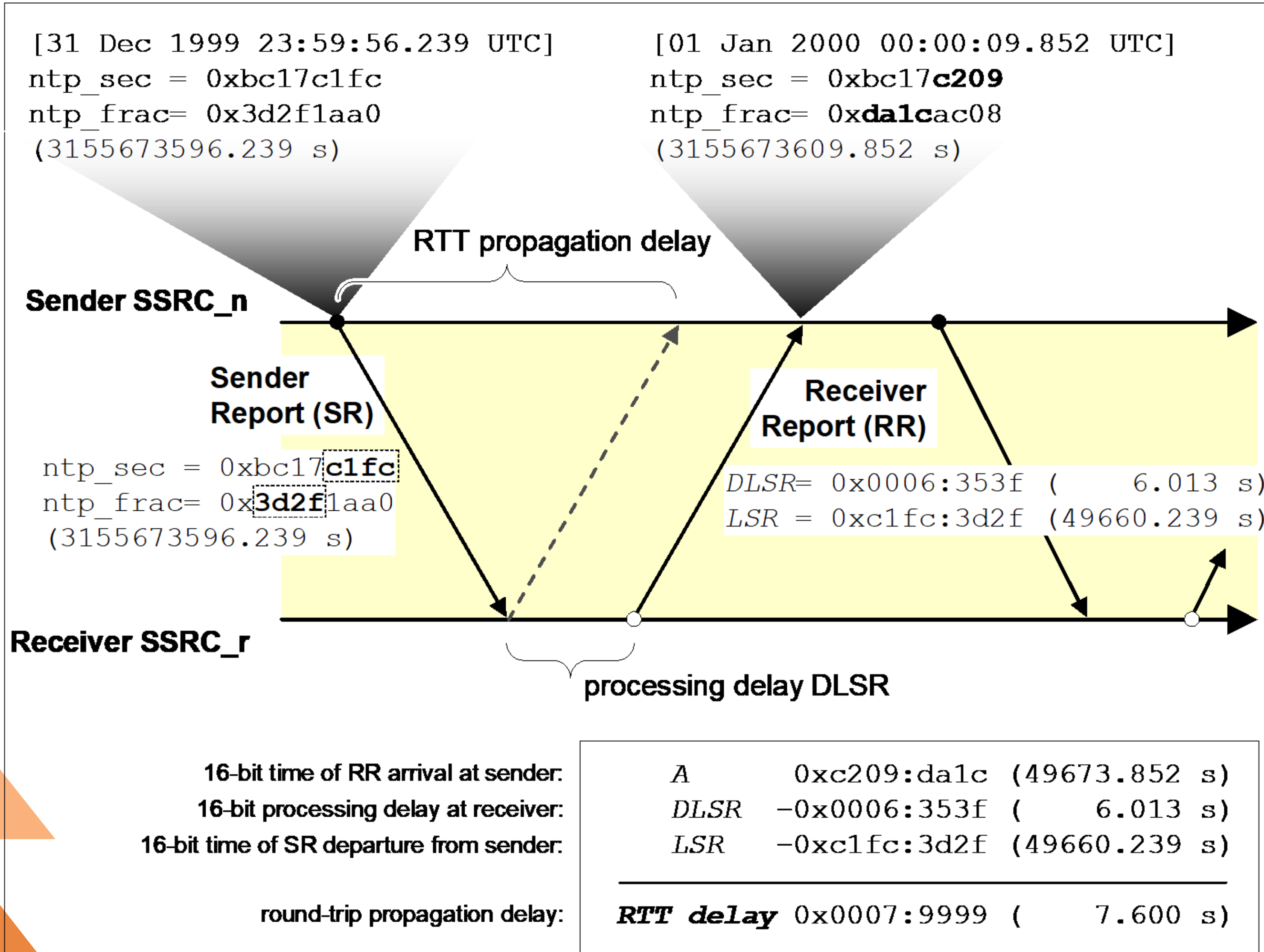
RTCP Sender Report Format

	0 1	2	3 7	8 15	16 31
header	ver.	P	reports count	packet type = ‘200’	packet length (in 32-bit words)
	SSRC of this reporter				
sender info	NTP timestamp, most significant word (seconds)				
	NTP timestamp, least significant word (seconds fraction)				
	RTP timestamp				
	sender’s packet count				
	sender’s byte count				
report block 1	SSRC_1 (SSRC of first source)				
	fraction lost			cumulative number of packets lost	
	extended highest sequence number received				
	interarrival jitter				
	last SR (LSR)				
	delay since last SR (DLSR)				
	report block 2 ...				
	profile-specific extensions				

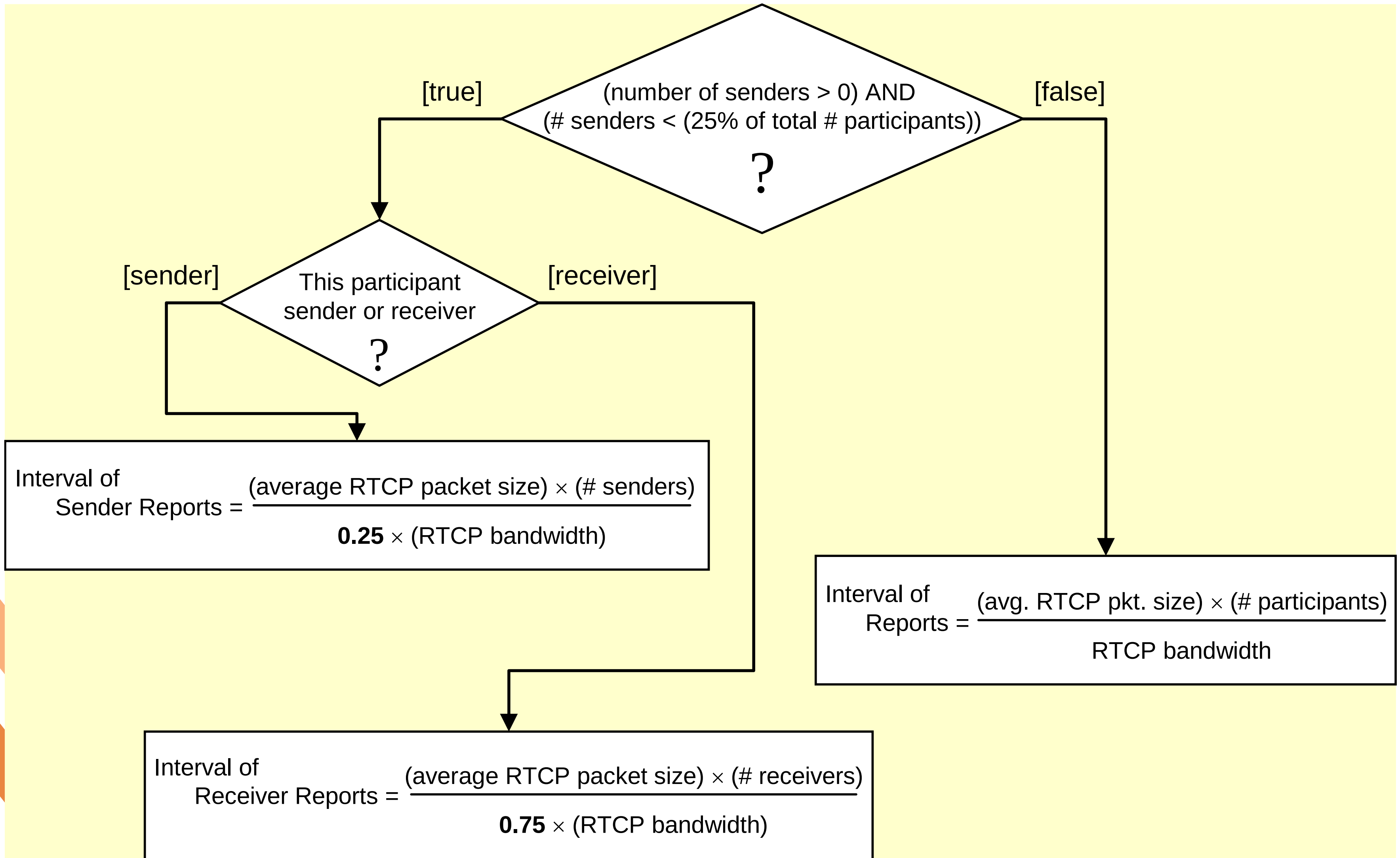
Inter-arrival Time Jitter



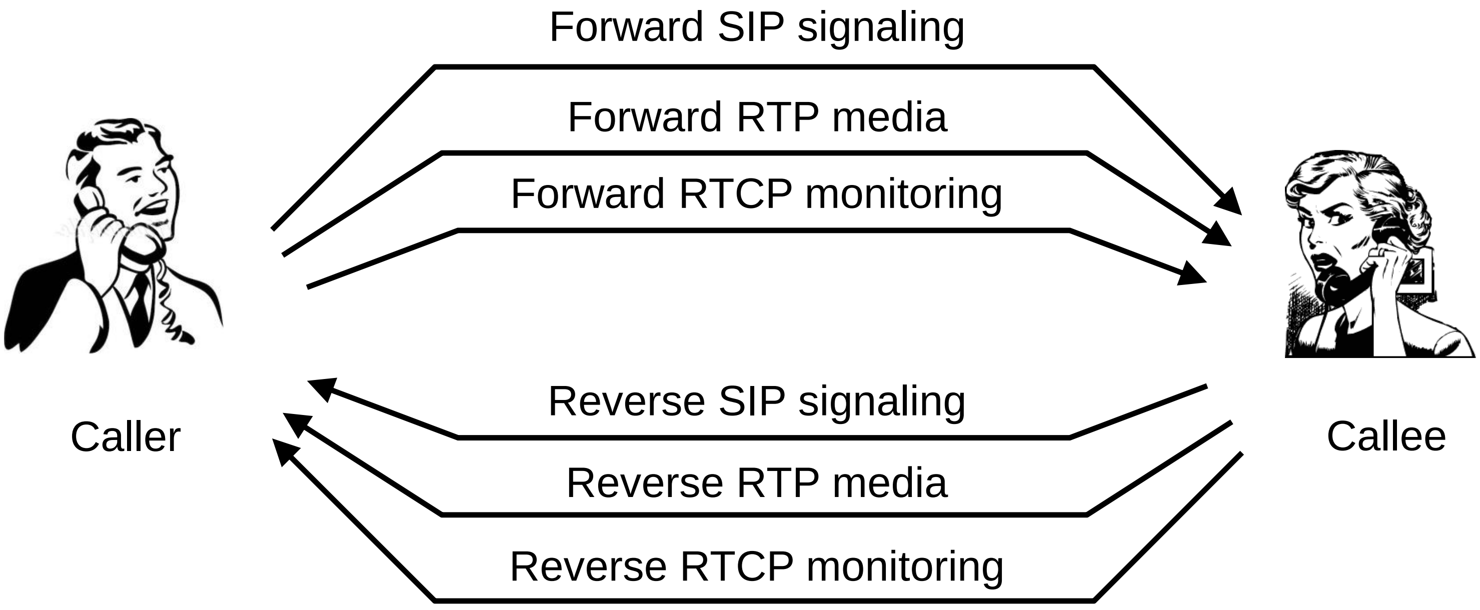
Example for RTT computation



Algorithm for RTCP Reporting Interval



VoIP Phone Call Session



Logical channels between the caller and callee during a VoIP call