Chapter 5
End-to-End Protocols

Transport Level

- Underlying best-effort network
  - drop messages
  - re-orders messages
  - delivers duplicate copies of a given message
  - limits messages to some finite size
  - delivers messages after an arbitrarily long delay

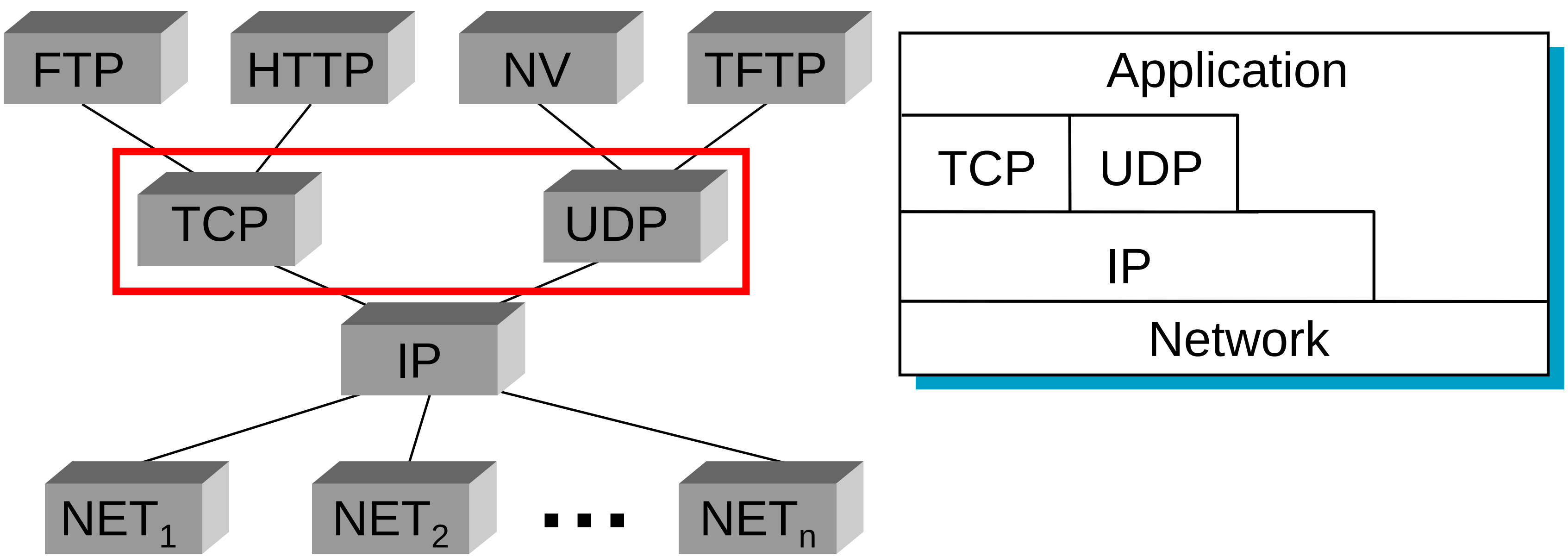
Transport Level

- **Transport level protocols:** support communication between the end application programs (the **end-to-end** protocol)
- Some properties are expected to provide for transport protocols:
  - **Guarantees** message delivery
  - Delivers messages in the **same order** they are 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

Simple Demultiplexer (UDP)

Internet Architecture

- The Internet architecture is also called the **TCP/IP architecture**
- The transport protocols are
  - UDP protocol
  - TCP protocol



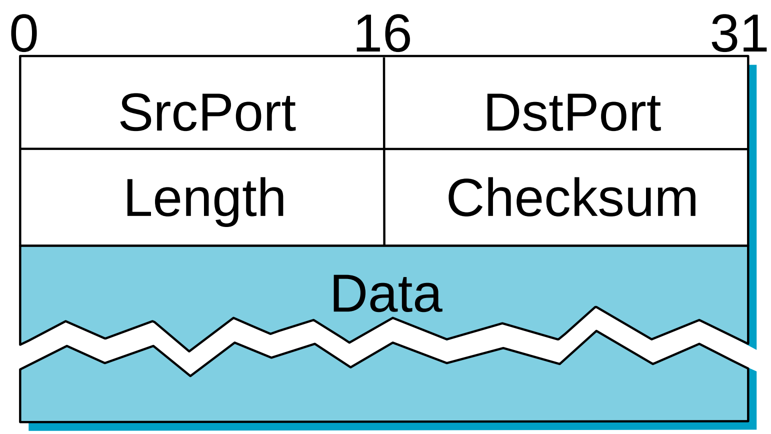
Simple Demultiplexer

- The **simplest** transport protocol extends the host-to-host delivery service of the underlying network into a **process-to-process** communication service
  - Many processes running on any given host
  - A level of **demultiplexing** is required for multiple processes on each host to share the network
  - The simplest transport protocol adds **no other functionality** to the **best-effort service** provided by the underlying network
- The Internet's **User Datagram Protocol (UDP)** is an example of such a transport protocol
- The only issue is the form of the address used to **identify the target process**

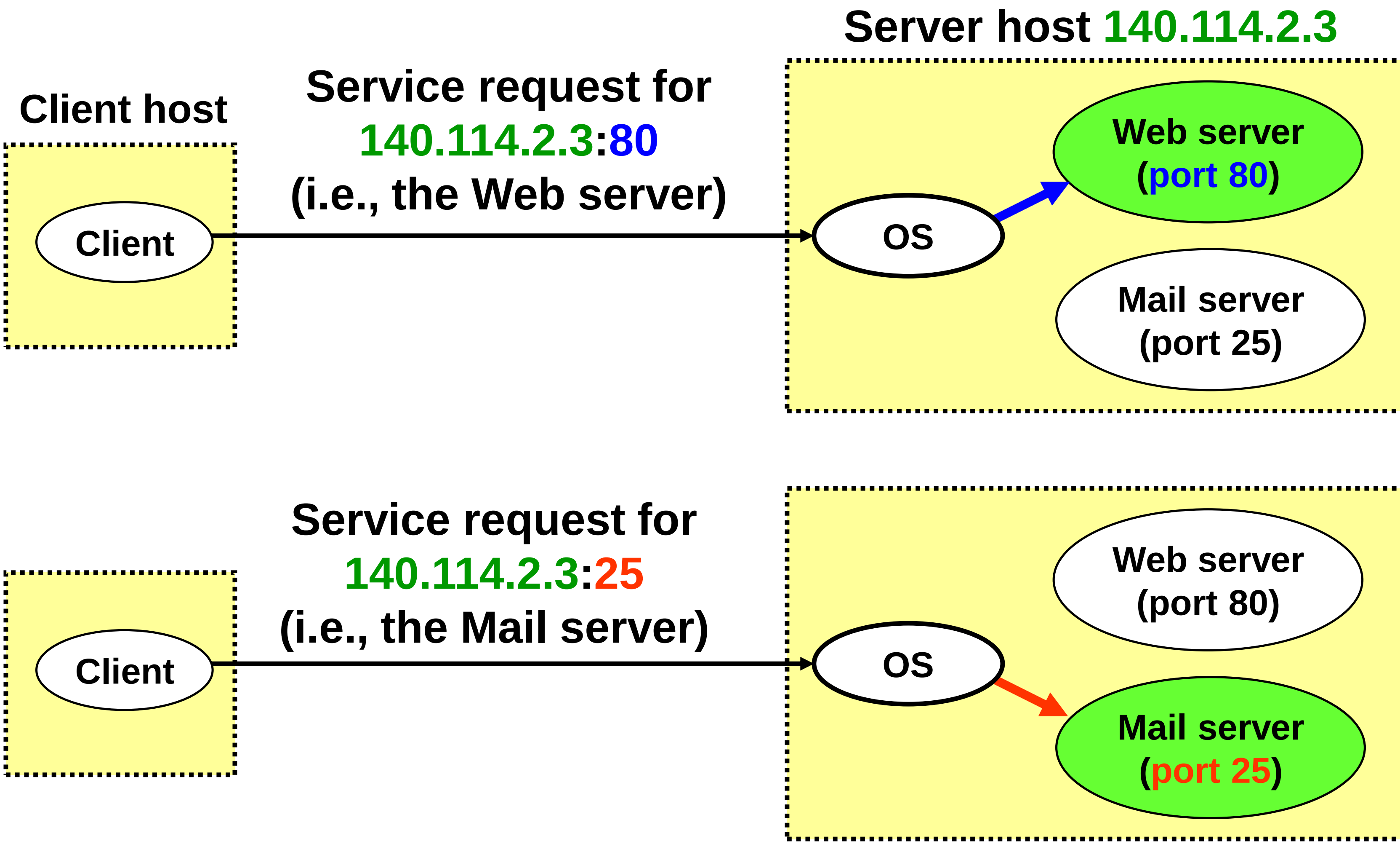
Simple Demultiplexer (UDP)

- The approach used by UDP is using an **abstract locator**
  - Called a **port** or **mailbox**
  - For a source process to send a message **to a port**, or for a destination process to receive the message **from a port**
- The **UDP port field** is **16 bits** long ⇒ up to **64 K** possible ports on a single host

Format for UDP header



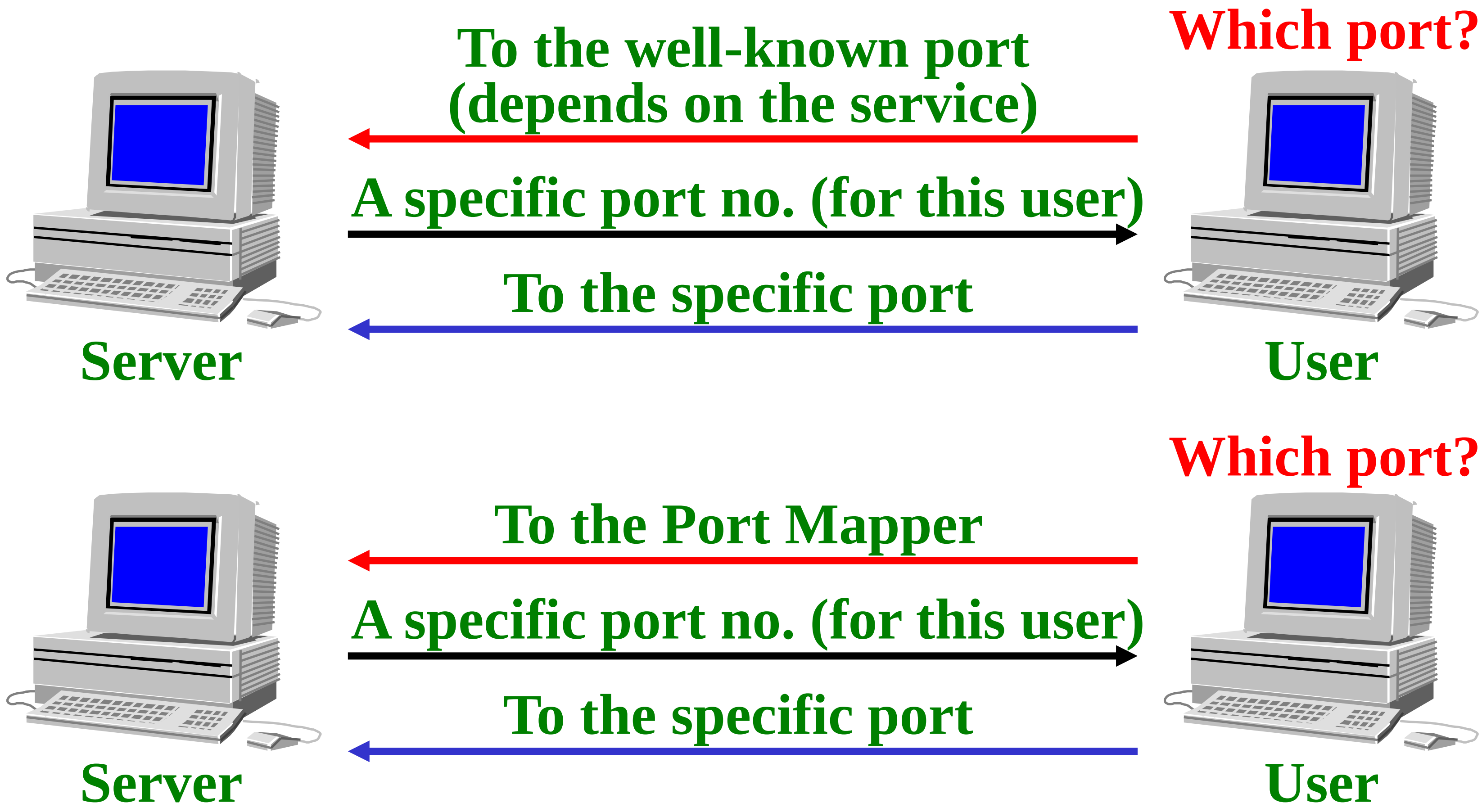
Port


Server host 140.114.2.3
Client host
Service request for
140.114.2.3:80
(i.e., the Web server)
Client
OS
Web server
(port 80)
Mail server
(port 25)
Service request for
140.114.2.3:25
(i.e., the Mail server)
Client
OS
Web server
(port 80)
Mail server
(port 25)

Simple Demultiplexer (UDP)

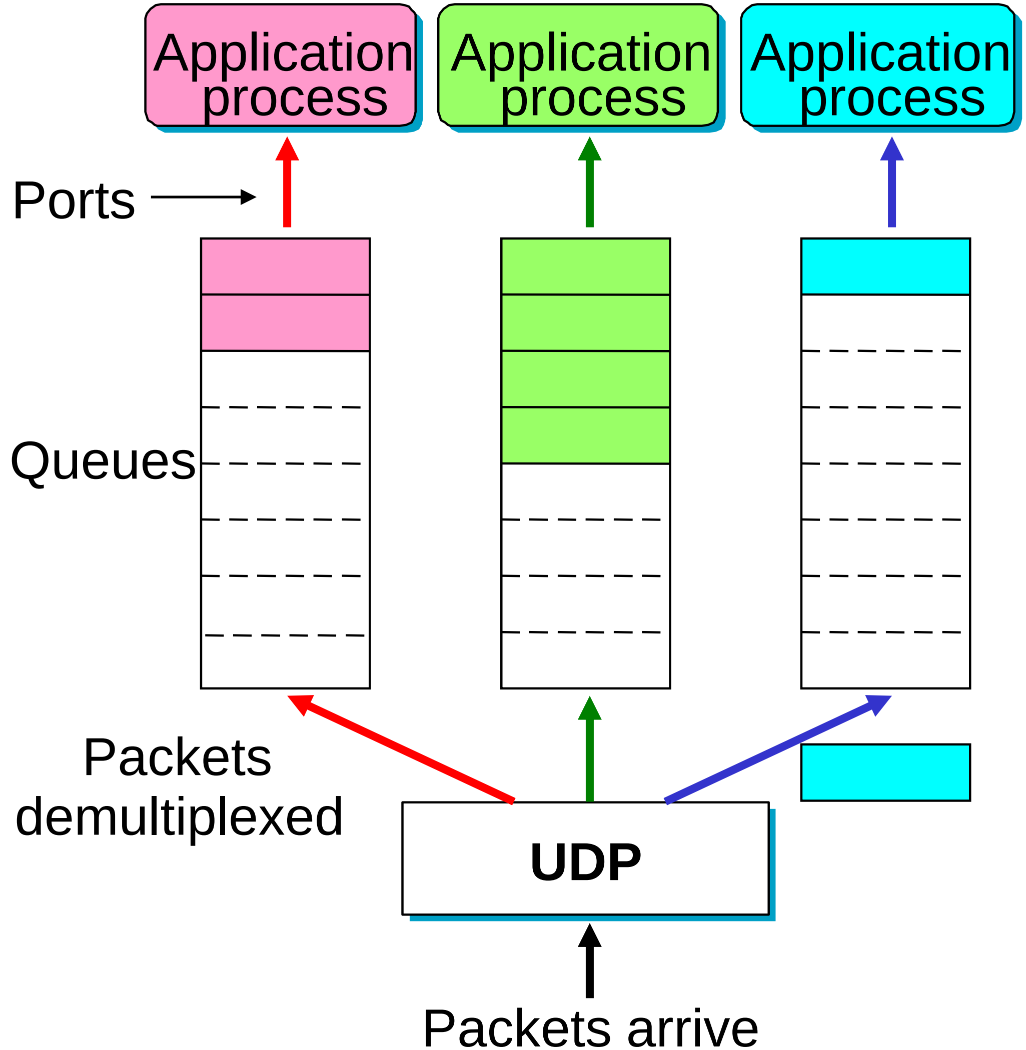
- How does the client learn the server's port in the first place?
- A common approach is for the server to accept messages at a **well-known port**, i.e. some fixed port widely published
  - **Domain Name Server (DNS): port 53**
  - **The mail server: port 25**
  - **The Unix talk program: port 517**
- A well-known port is the **starting point** for communication:
  - The client and server use the well-known port to **agree on some other port** for subsequent communications
- An alternative strategy is using only a well-known port for the **Port Mapper** service to accept messages
  - A client send a message to ask for the port it should use

Simple Demultiplexer (UDP)



Simple Demultiplexer (UDP)

- A port is implemented by a **message queue**
- For an arrived message, the protocol appends it to the end of the queue
- When a process wants to **receive a message**, one is removed from the front of the queue
- If the queue is empty, the process **blocks** until a message becomes available



Reliable Byte Stream (TCP)

Reliable Byte Stream (TCP)

- A **reliable, connection-oriented, byte-stream service:**
  - Do not need to worry about **missing** or **reordered** data
- **TCP:** the Internet's **Transmission Control Protocol**
  - Guarantees the **reliable, in-order delivery** of a stream of bytes
  - A **full-duplex** protocol: each TCP connection supports a pair of byte streams
  - A **flow-control** mechanism: allows the receiver to limit the amount of data that the sender can transmit at a given time
  - A **demultiplexing** mechanism
  - A **congestion-control** mechanism

End-to-End Issue (Variant RTT)

- The **sliding window algorithm** in TCP runs over the Internet
  - Which is quite different to point-to-point link
- TCP needs an explicit **connection establishment phase**
  - The two sides agree to exchange data with each other
  - The two parties establish some **shared state** to enable the sliding window algorithm to begin
- TCP also has an explicit **connection teardown phase**
  - For each host to know it is OK to **free this state**
- Different connections may have **widely different RTTs**
  - The TCP protocol must be able to support all conditions with different round-trip times
  - The **timeout mechanism** that triggers retransmissions must be **adaptive**

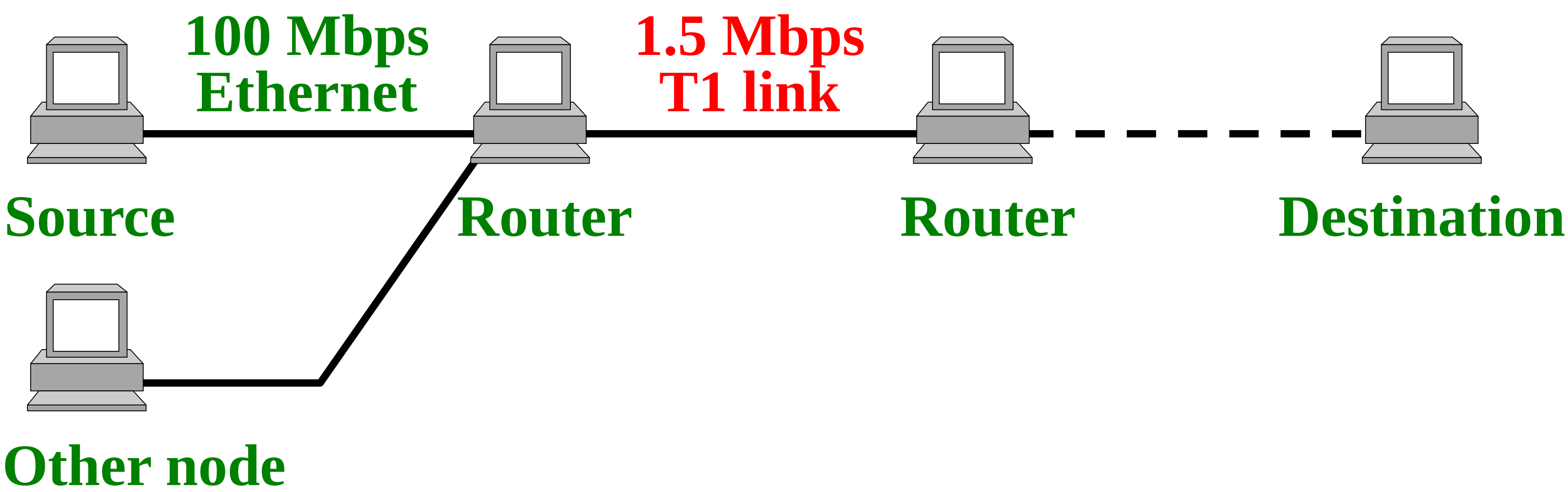
End-to-End Issue (Flow-control)

- The packets may be **reordered** as they cross the Internet
  - Packets that are **slightly out of order** can be correctly reordered by using the **sequence number**
  - If a packet is delayed until IP's time to live **(TTL) field expires**, the packet will be **discarded**
- The amount of **resources** dedicated to any one TCP connection is **highly variable**
  - Each side must **"learn"** what resources (e.g. buffer space) **the other side** is able to apply to the connection

  $\Rightarrow$ The **flow-control** mechanism

End-to-End Issue (Network Congestion)

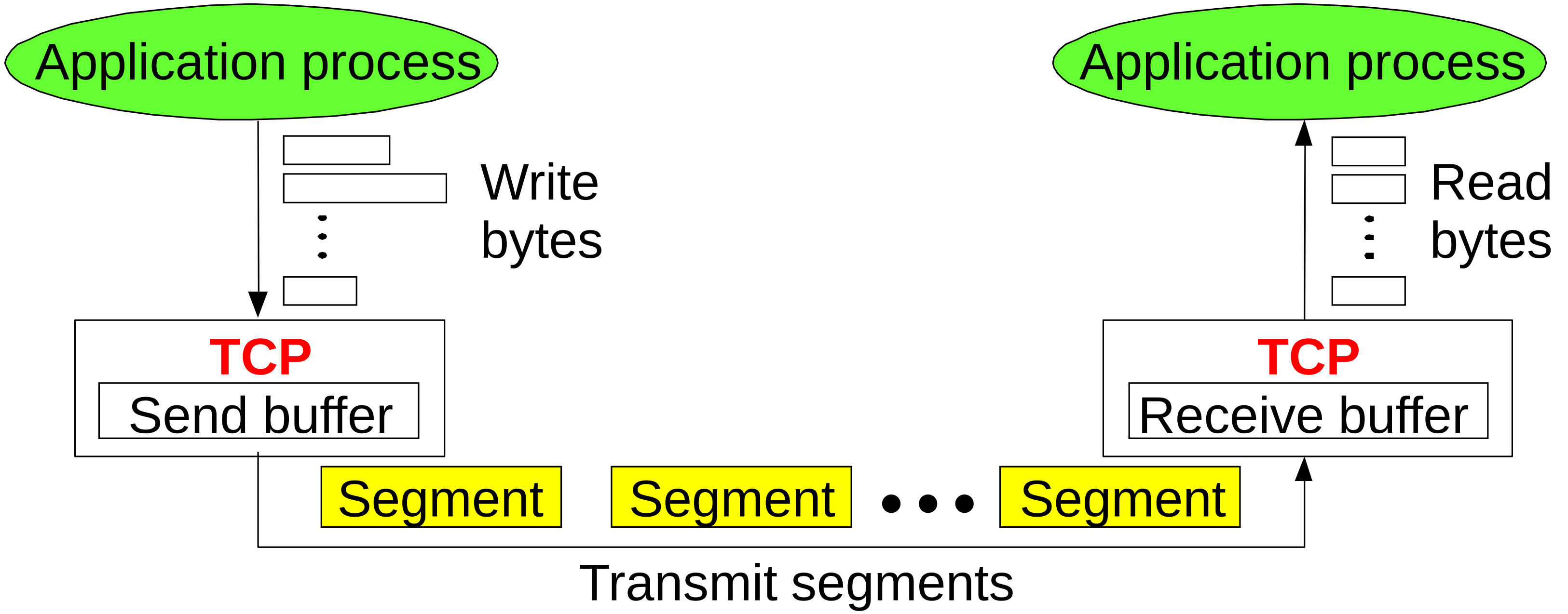
- The sending side of a TCP connection has no idea what links will be traversed to reach the destination
  - 100 Mbps fast Ethernet ↔ 1.5 Mbps T1 link ↔ …
  - This leads to the problem of **network congestion**



Segment Format

Segment

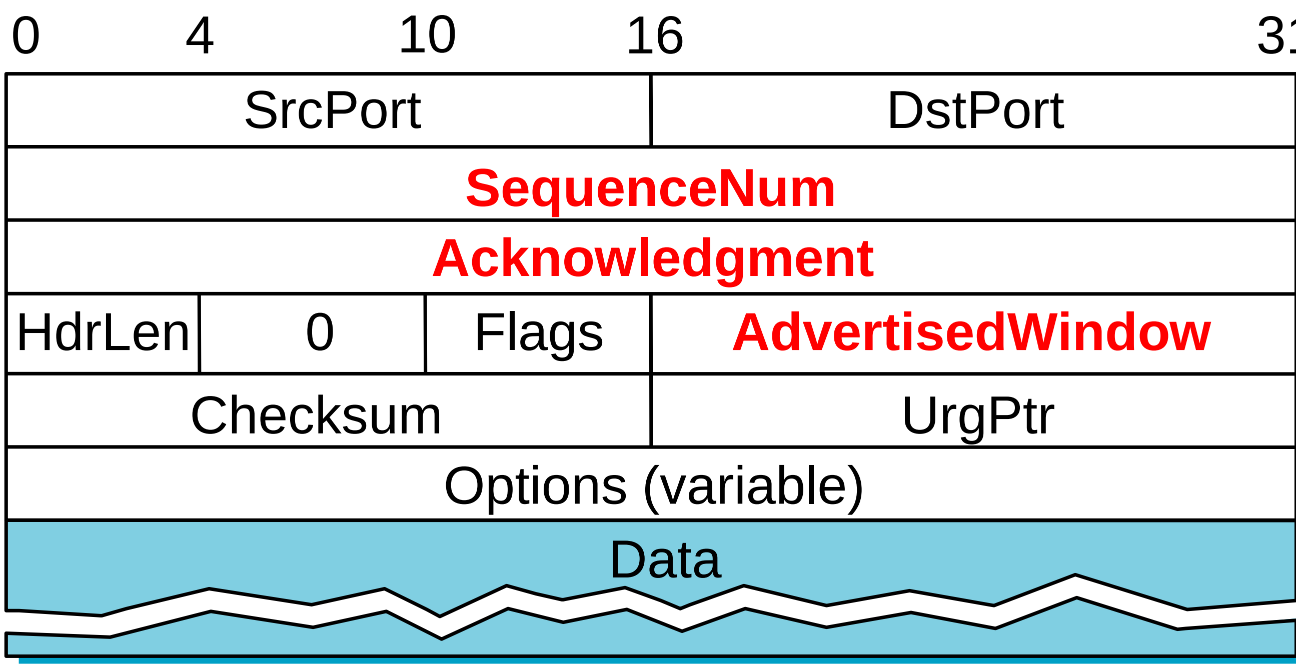
- TCP connection supports **byte streams flowing** in both direction
  - The source host buffers enough bytes from the sending process to fill a reasonably sized packet
  - The packet is called **segment**



Segment Format

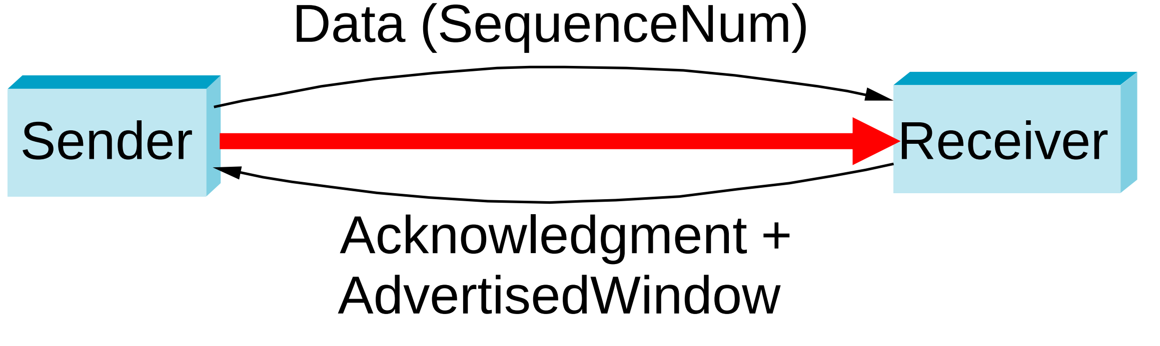
- **SrcPort** and **DstPort:**
  - The source and destination ports
- **Acknowledgment**, **SequenceNum**, and **AdvertisedWindow:**
  - All involved in TCP's sliding window algorithm

TCP header format



Segment Format

- **SequenceNum**:
  - Contains the sequence number for the **first byte of data** carried in the segment
  - Each byte of data has a sequence number
- **Acknowledgment** and **AdvertisedWindow:**
  - Carry information about the flow of data going in **the other direction**



Segment Format

- **HdrLen field:**
  - The length of the header in **32-bit words**
- The **6-bit Flags field:**
  - Used to relay **control information** between TCP peers
- **UrgPtr field:**
  - Indicates where the **nonurgent data** contained in this segment begins
  - **Urgent data** is contained in the front of a segment
- **Checksum field:**
  - Error detection

Segment Format

- The possible flags include **SYN, FIN, RESET, PUSH, URG** and **ACK** (**6 bits $\Rightarrow$ 6 flags**)
  - **SYN:** is used when **establishing** a TCP connection
  - **FIN:** is used when **terminating** a TCP connection
  - **RESET:** is used when the receiver has become confused, and so wants to **abort the connection**
  - **PUSH:** is used when the sending process **invokes the push operation** to efficiently flush the buffer of unsent bytes
  - **URG:** is used when this segment contains **urgent data**
  - **ACK:** is set when the **Acknowledgment** field is valid

Connection Establishment and Termination

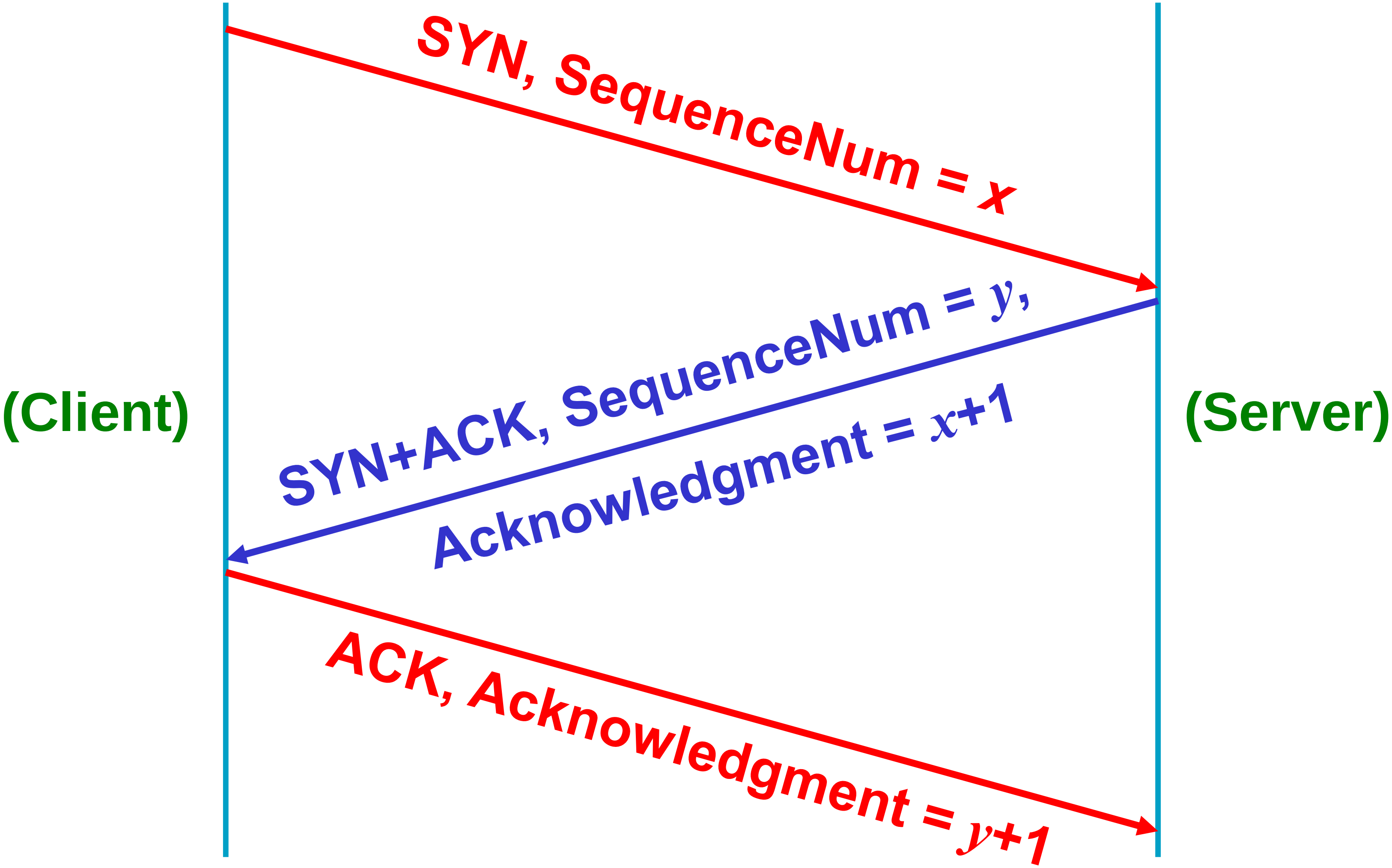
Connection Establishment and Termination

- A TCP connection begins with a **client** (caller) doing an active open to a **server** (callee)
- The two sides engage in an **exchange of messages** to establish the connection
- Only after this connection establishment phase is over, the two sides can begin sending data
- The algorithm used by TCP to establish and terminate a connection is called a **three-way handshake**
  - Involves **the exchange of three messages** between the client and the server

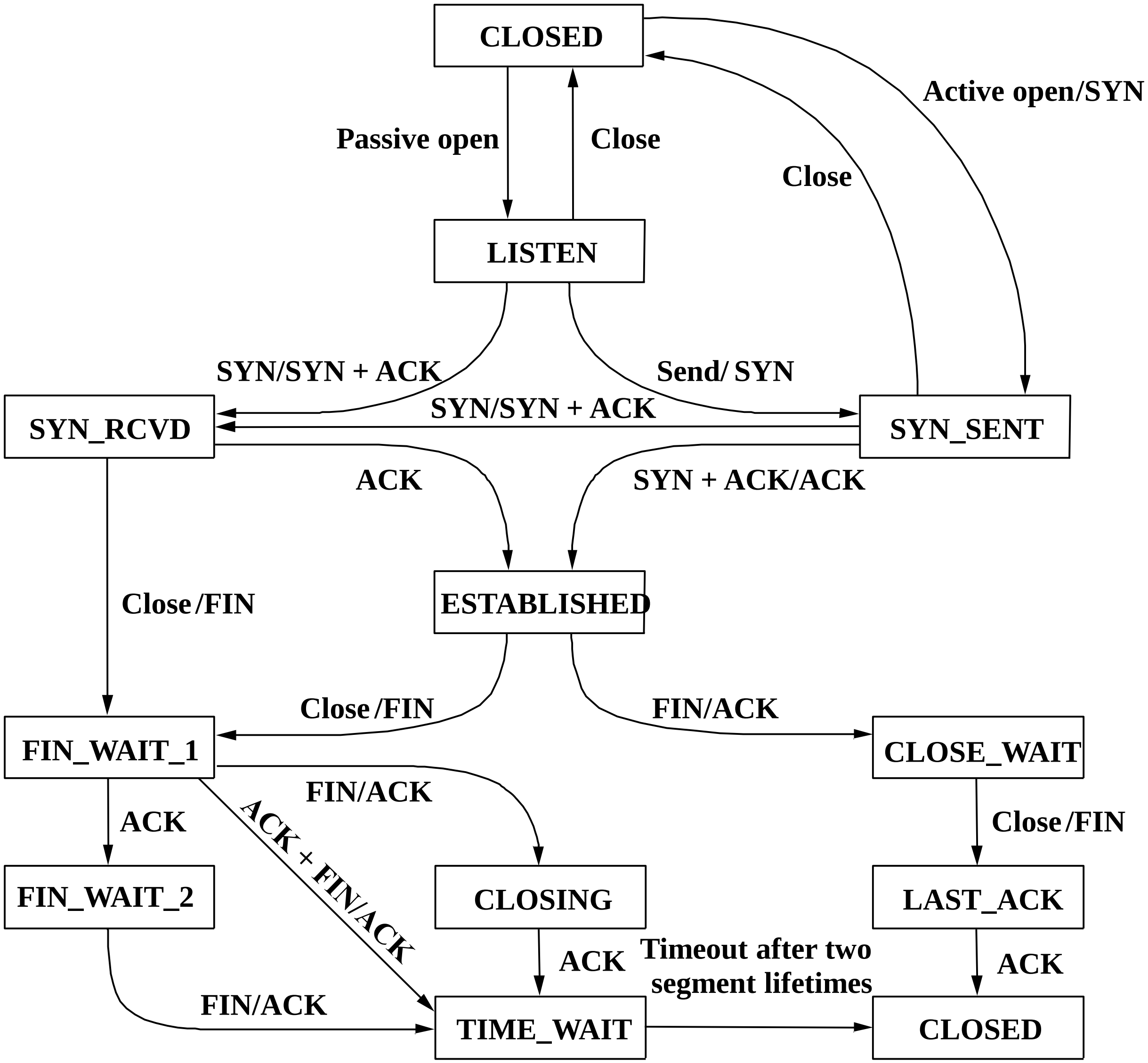
Connection Establishment and Termination

- The client sends a segment to the server stating the **initial sequence number**
  - Flags = **SYN**, SequenceNum = ***x***
- The server responds with a single segment
  - To **acknowledge** the client's sequence number
    - Flags = **ACK**, Ack = ***x*+1** (next sequence number expected is *x*+1)
  - To state its own **beginning sequence number**
    - Flags = **SYN**, SequenceNum = ***y***
- The client responds with a segment that **acknowledges** the server's sequence number
  - Flags = **ACK**, Ack = ***y*+1**

Connection Establishment and Termination



State Transition Diagram



Sliding Window Algorithm

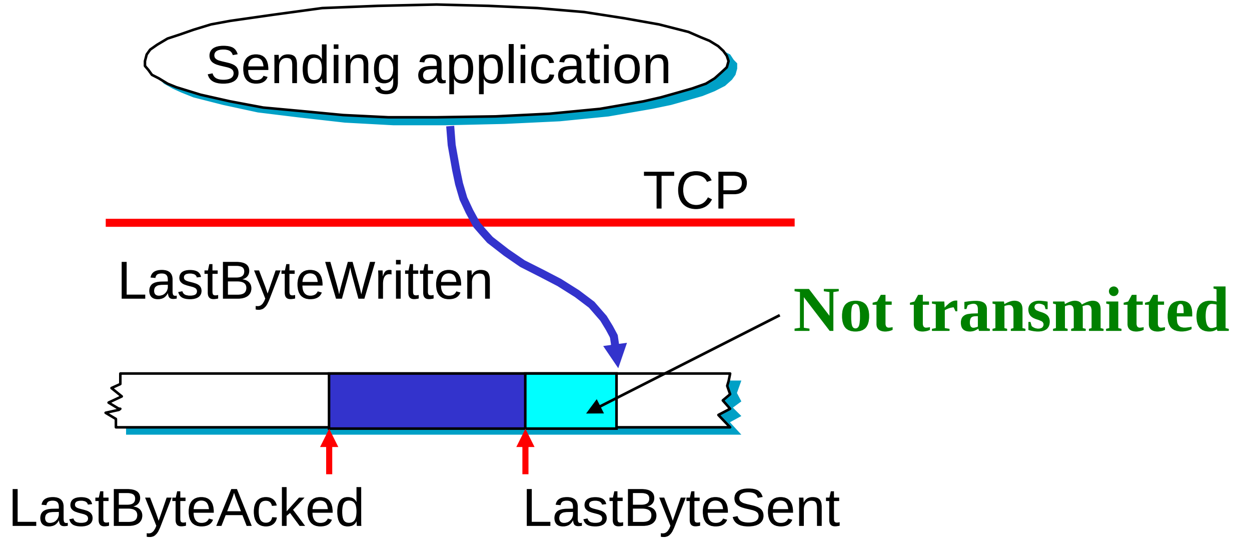
Sliding Window Algorithm

- **TCP sliding window algorithm:**
  - It guarantees the **reliable delivery** of data
  - It ensures that data is delivered **in order**
  - It enforces **flow control** between the sender and the receiver
- Rather than having a fixed-size sliding window, the receiver **advertises a window size** to the sender
  - Based on the **amount of memory** allocated to the connection for the purpose of buffering data
  - Using the **AdvertisedWindow** field in the TCP header
- The sender is limited to having **no more than** a value of AdvertisedWindow bytes of **unacknowledged data**

Sliding Window Algorithm (Sending Side)

- TCP on the sending side maintains a **send buffer** used to store
  - The data that **has been sent** but **not yet acknowledged**
  - The data that **has been written** by the sending application, but **not transmitted**

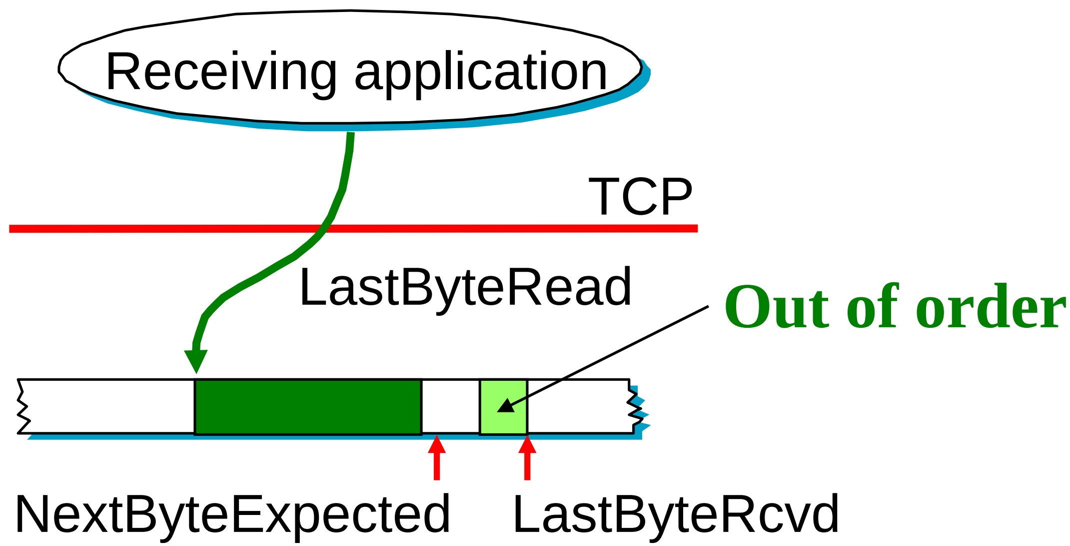
TCP send buffer



Sliding Window Algorithm (Receiving Side)

- TCP on the receiving side maintains a **receive buffer** used to hold
  - The data that **arrives out of order**
  - The data that is in the correct order, but that the application process **has not yet had the chance to read**

TCP receive buffer

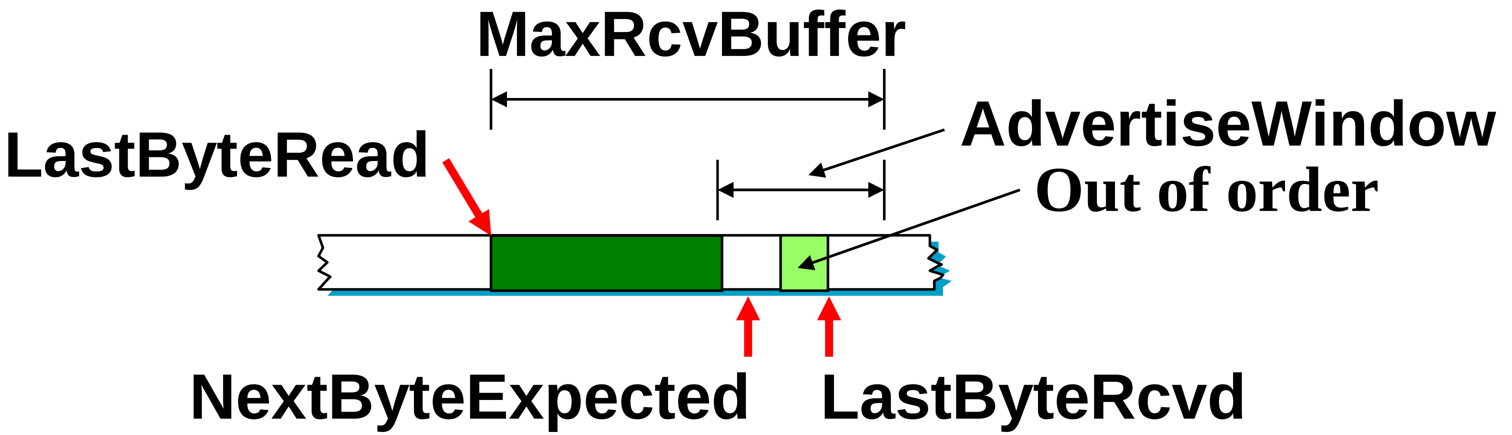


Sliding Window Algorithm

- In the sending side, three pointers are maintained into the send buffer: **LastByteAcked**, **LastByteSent**, and **LastByteWritten**
  - **LastByteAcked ≤ LastByteSent**
  - **LastByteSent ≤ LastByteWritten**
- In the receiving side, three pointers are maintained into the receive buffer: **LastByteRead**, **NextByteExpected**, and **LastByteRcvd**
  - **LastByteRead < NextByteExpected**
  - **NextByteExpected ≤ LastByteRcvd + 1**
    - "=" holds when there is no out of order byte

Flow Control (Receive Buffer)

- The buffer sizes are finite: **MaxSendBuffer, MaxRcvBuffer**
- To avoid overflowing the **receive buffer**
  - **LastByteRcvd – LastByteRead ≤ MaxRcvBuffer**
- The receiver advertises a window size representing the amount of **free space** remaining in its buffer
  - **AdvertiseWindow = MaxRcvBuffer – ((NextByteExpected – 1) – LastByteRead)**

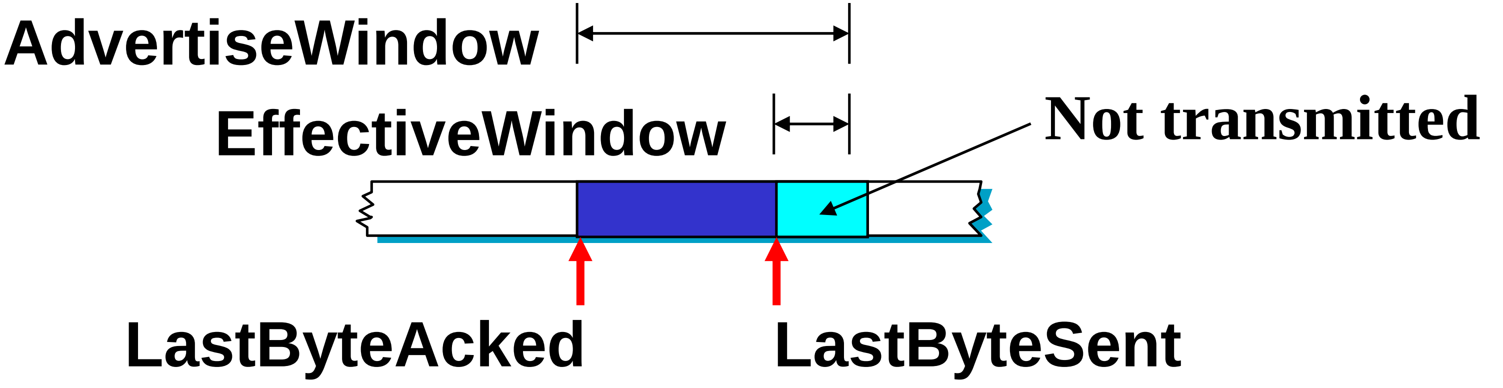


Flow Control (Receive Buffer)

- If the local process is reading data just **as fast as it arrives**
  - The advertised window stays open
    - **AdvertiseWindow = MaxRcvBuffer**
- If the receiving process **falls behind**
  - The advertised window grows smaller until it goes to **0**

Flow Control (Receive Buffer)

- TCP on the sending side must ensure that
  - **LastByteSent–LastByteAcked ≤ AdvertiseWindow**
- To avoid overflowing the **receive buffer**, the sender computes an effective window that limits how much data it can send:
  - **EffectiveWindow = AdvertiseWindow – (LastByteSent – LastByteAcked)**

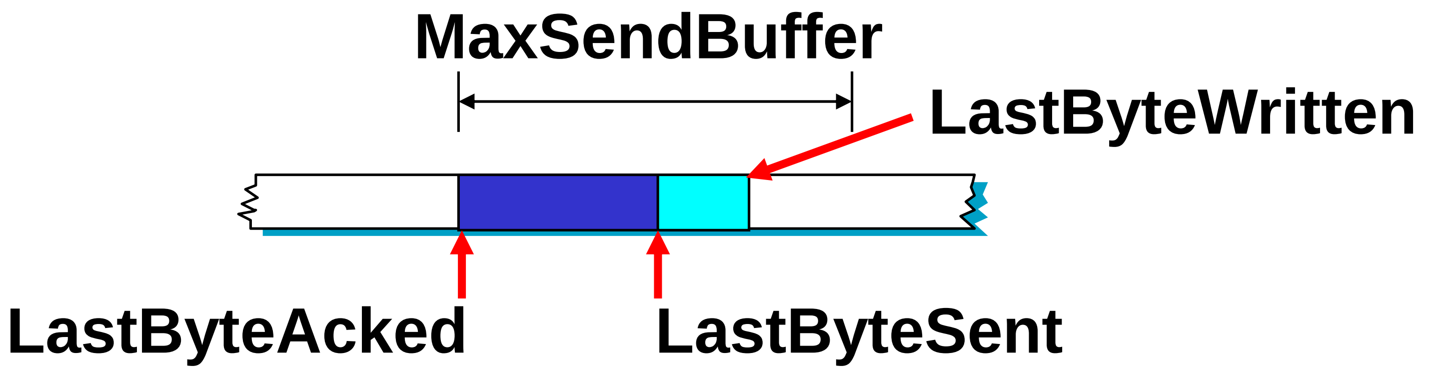


Flow Control (Receive Buffer)

- **EffectiveWindow** must be **greater than 0** before the source can send more data
- If a segment arrives acknowledging ***x*** bytes and the receiving process **was not** reading any data
  - The receive buffer **does not** free any buffer space
  - The advertise window is ***x*** bytes smaller
  - The sender can increase **LastByteAcked** by ***x***
  - The sender would be able to **free** buffer space, but **not to send** any more data

Flow Control (Send Buffer)

- The sending side must also make sure that the local application process **does not overflow** the send buffer
  - **LastByteWritten–LastByteAcked ≤ MaxSendBuffer**
- If the sending process ties to write ***y*** bytes to TCP, but
  - **LastByteWritten – LastByteAcked + *y* > MaxSendBuffer**
  - Then TCP **blocks** the sending process



Flow Control (Send Buffer)

- A **slow** receiving process ultimately stops a **fast** sending process
  - The receive buffer fills up

  $\Rightarrow$ The advertise window shrinks to 0

  $\Rightarrow$ The sending side cannot transmit any data

  $\Rightarrow$ The send buffer fills up

  $\Rightarrow$ TCP blocks the sending process
- TCP is designed to make the receive side **as simple as possible**
  - It simply responses to segments from the sender

Flow Control

- How does the sending side know that **the advertised window is no longer 0**?
- TCP **always** sends a segment in response to a received segment
  - Contains the latest values for the **Acknowledge** and **AdvertiseWindow** fields
- Whenever the receiving side advertises a window size of 0
  - The sending side persists in sending a **probe segment** with **1 byte** of data
  - Each probe segment **triggers a response** containing the **current** advertised window
  - Eventually, a response reports a **nonzero** advertised window

Protection Against Wrap Around

- 32-bit **`SequenceNum`**

Bandwidth	Time Until Wrap Around
T1 (1.5 Mbps)	6.4 hours
Ethernet (10 Mbps)	57 minutes
T3 (45 Mbps)	13 minutes
FDDI (100 Mbps)	6 minutes
STS-3 (155 Mbps)	4 minutes
STS-12 (622 Mbps)	55 seconds
STS-24 (1.2 Gbps)	28 seconds

Keeping the Pipe Full

- 16-bit **`AdvertisedWindow`**

Bandwidth	Delay x Bandwidth Product
T1 (1.5 Mbps)	18KB
Ethernet (10 Mbps)	122KB
T3 (45 Mbps)	549KB
FDDI (100 Mbps)	1.2MB
STS-3 (155 Mbps)	1.8MB
STS-12 (622 Mbps)	7.4MB
STS-24 (1.2 Gbps)	14.8MB

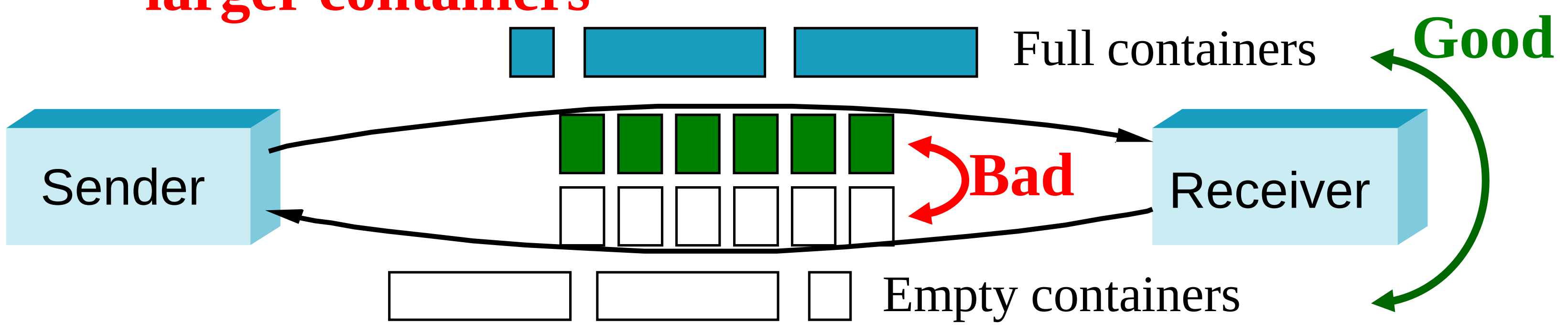
Triggering Transmission

Triggering Transmission

- TCP has three mechanisms to trigger the transmission of a segment
  - It sends a segment as soon as it has **collected MSS (maximum segment size)** bytes from the sending process
    - MSS is generally set to the size of the largest segment TCP can send **without causing IP fragmentation**
  - It sends a segment when the **sending process** has **asked** it to do so
    - TCP supports a **PUSH operation** and the sending process invokes it to **flush** the buffer of unsent bytes
  - It sends a segment when a **timer fires**
    - The resulting segment contains **all** bytes that are currently buffered for transmission

Triggering Transmission

- **Data segment:** full containers; **ACKs:** empty containers;
  - **MSS-sized** segments: large container; **1-byte** segments: small container
- **Silly window syndrome:** If the sender aggressively fills an empty container **as soon as it arrives**
  - Any **small container** introduced into the system remains in the system indefinitely
  - It never coalesces with adjacent containers to create **larger containers**

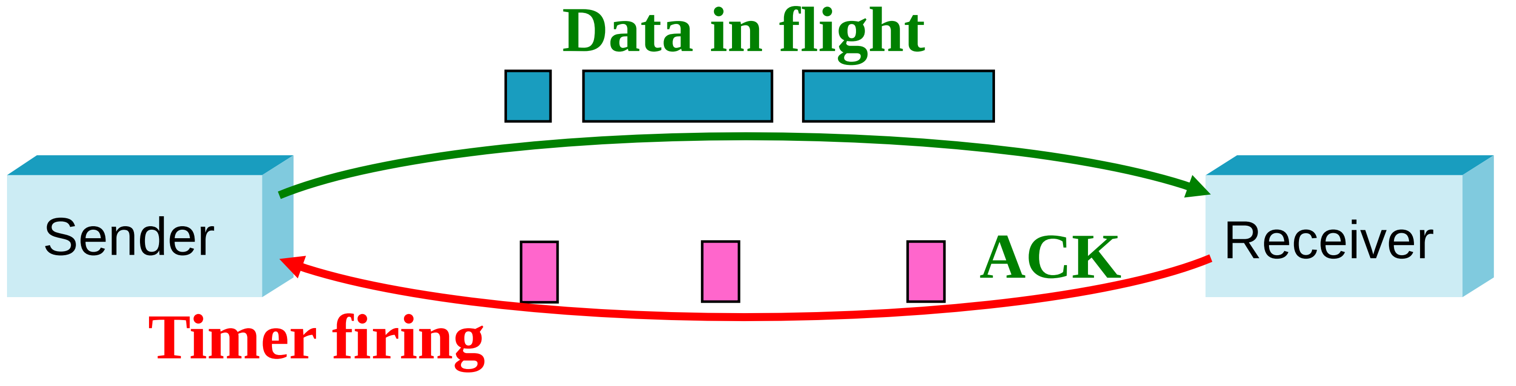


Triggering Transmission (Window Size)

- Triggering transmission is applied to keep the receiver from introducing a small container:
  - After advertising a zero window, the **receiver** must **wait for space equal to an MSS** before it advertises an open window
- Some mechanisms are also introduced to coalesce small containers
  - The receiver can do this by **delaying ACKs** — sending one **combined ACK** rather than multiple smaller ones
    - Reply a large window size

Triggering Transmission (Sender)

- If there is data to send but the window is open **less** than MSS
  - It waits some amount of time before sending the data:
    - Introduce a **timer**
  - It transmits when the timer expires
- A **self-clocking** solution: **Nagle's algorithm**
  - If TCP has any data in flight, the sender will eventually receive an ACK – treated like a timer firing



Triggering Transmission (Sender)

- **Nagle's algorithm:**
  - It's **always OK** to send a **full** segment if the window allows
  - It's OK to send a small amount of data if there are currently **no segments in transit**
  - If there is anything in flight, the sender must **wait for an ACK** before transmitting the next segment

Adaptive Retransmission

Adaptive Retransmission

- TCP **retransmits** each segment if an ACK is not received in a certain period of time
- TCP sets this **timeout** as a function of
  - The **RTT** it expects between the two ends of the connection
- Since the RTTs are **various with time**, TCP uses an **adaptive retransmission mechanism**
  - To keep a **running average** of the RTT
  - Then compute the timeout as a function of this RTT

Adaptive Retransmission

- Every time TCP sends a data segment, it records the time
- When an ACK for that segment arrives, TCP reads the time again and then takes the difference as a **SampleRTT**
- TCP then computes an **EstimatedRTT** as a **weighted average** between the previous estimate and this new sample
  - **EstimatedRTT = $\alpha \times$ EstimatedRTT + (1–α) $\times$ SampleRTT**
  - α is selected to **smooth** the **EstimatedRTT**
- TCP then uses **EstimatedRTT** to compute the timeout:
  - **TimeOut = 2 $\times$ EstimatedRTT**

Adaptive Retransmission

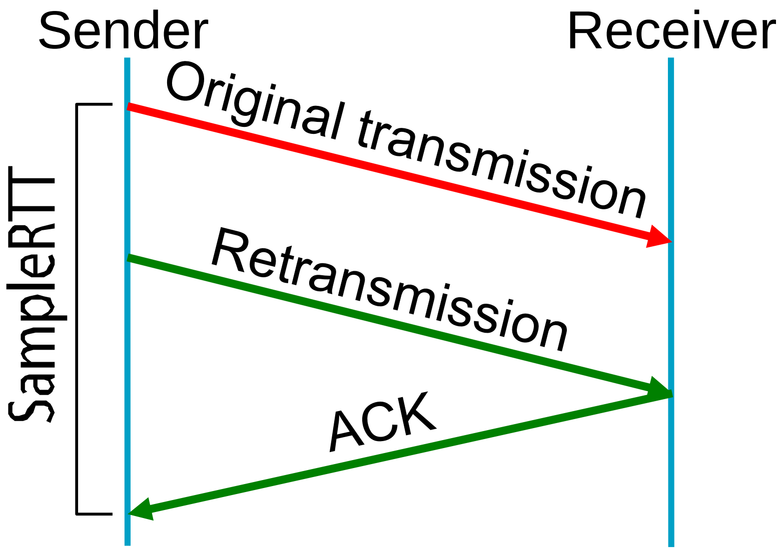
- The setting of α:
  - A **small α** tracks changes in the RTT but is heavily influenced by **temporary fluctuations**
  - A **large α** is more **stable** but is not quick enough to adapt to real change
  - It recommended a setting of α between **0.8 and 0.9**

- Problem: An ACK does not really acknowledge a transmission
  - It actually acknowledges the **receipt** of data

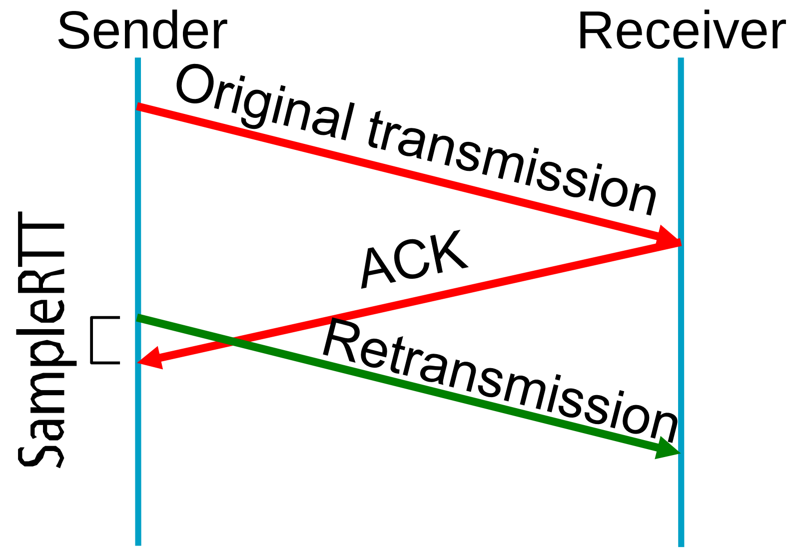
Adaptive Retransmission

- Whenever a segment is **retransmitted** and then an ACK arrives at the sender
  - It is impossible to determine if this ACK should be associated with the **first or** the **second** transmission

SampleRTT too large



SampleRTT too small



Adaptive Retransmission

- **Karn/Partridge algorithm:**
  - Whenever TCP **retransmits** a segment, it **stops** taking samples of the RTT
  - It only measures **SampleRTT** for segments that have been **sent only once**
  - Each time TCP retransmits, it sets the next timeout to be **twice the last timeout** (rather than the last EstimatedRTT)
    - TCP use **exponential backoff**
- Problem: If the variation among samples is **small**
  - Then the EstimatedRTT can be **better trusted**
- If the variation among samples is **large**
  - Then the timeout value **should not** be too tightly coupled to the EstimatedRTT

Adaptive Retransmission

- **Jacobson/Karels algorithm:**
  - The sender measures a new **SampleRTT** as before
  - The timeout is calculated as follows:

Difference = SampleRTT – EstimatedRTT

EstimatedRTT = EstimatedRTT + ($\delta \times$ Difference)

Deviation = Deviation + δ(|Difference| – Deviation)

  - δ is a fraction between **0 and 1**
  - TCP then computes the timeout value as follows:

TimeOut = $\mu \times$ EstimatedRTT + $\phi \times$ Deviation

  - μ is typically set to **1** and ϕ is set to **4**
- When the variance is **small**, TimeOut is close to EstimatedRTT
- When the variance is **large**, Deviation will dominate TimeOut

Transport for Real-Time Application (RTP)

Real-time Transport Protocol (RTP)

- RTP contains a considerable amount of functionality that is specific to multimedia applications
  - Runs **on top** of one of the transport-layer protocols **UDP**
  - Provides **common end-to-end functions** to a number of applications
- Multimedia applications are sometimes divided into two classes:
  - **Conferencing applications**
  - **Streaming applications**
- RTP can run over many lower-protocols, but **commonly UDP**

Application
RTP
UDP
IP
Subnet

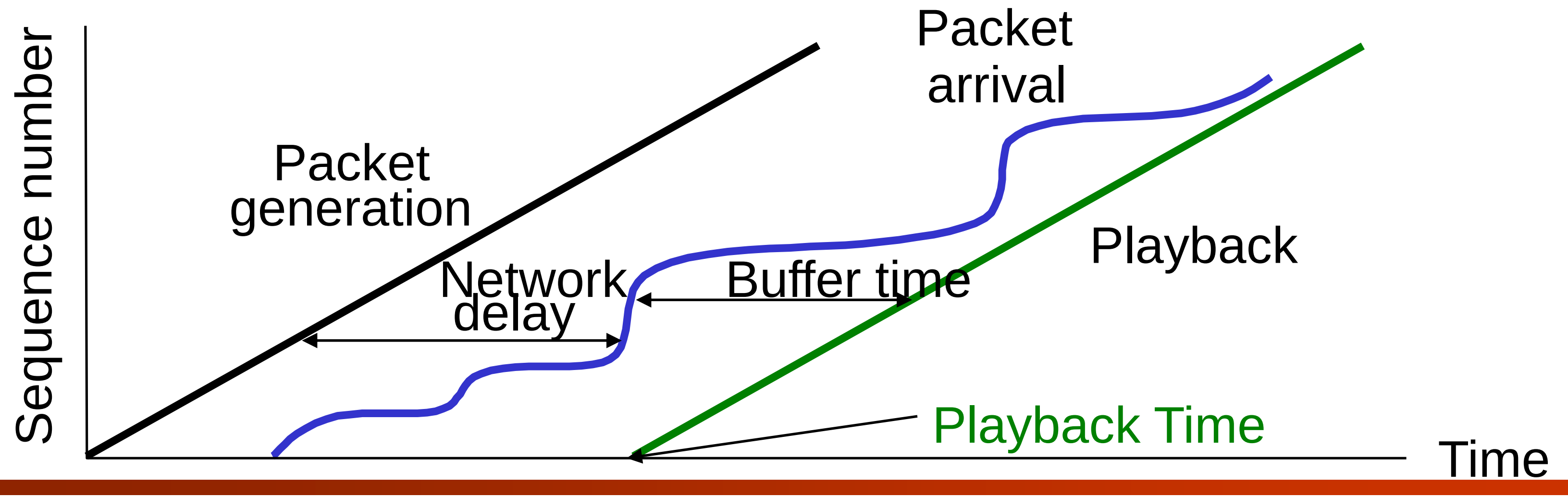
Protocol stack for multimedia applications using RTP

Requirements for RTP

- The most basic requirement for a general-purpose multimedia protocol is that it allow **similar applications** to **interoperate** with each other
  - Two **independently** implemented applications to communicate with each other
- **Coding schemes agreement:** A sender tell a receiver the **used coding scheme**, and negotiate until a scheme is identified
  - There are only quite a few different coding schemes

Requirements for RTP

- **Timing:** To enable the recipient of a data stream to determine the **timing relationship** among the received data
  - **Real-time applications:** need to place received data into a **playback buffer** to smooth out the jitter introduced into the data stream during transmission
    - Some sort of **timestamping** of the data is necessary for the receiver to play it back at the appropriate time



Requirements for RTP

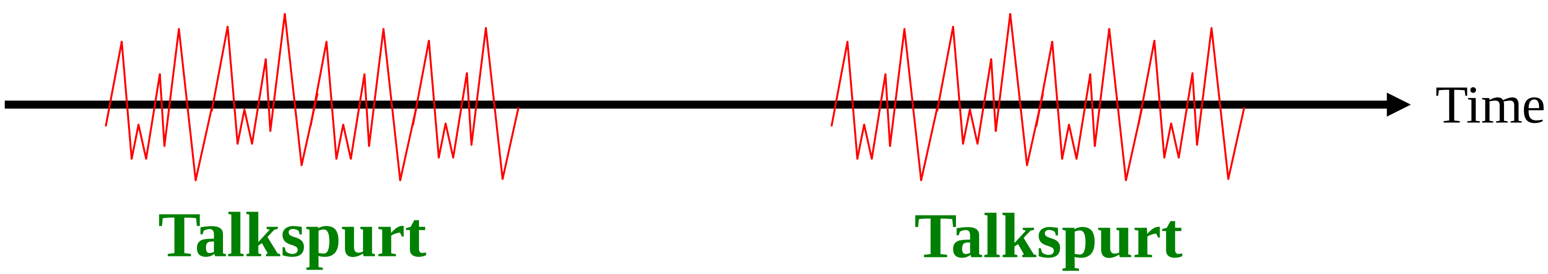
- **Synchronization:** To synchronize **multiple media** in a conference
  - For example to synchronize an **audio** and **video** stream that are originating from the same sender
- **Indication of packet loss:** An application with **tight** latency bounds generally cannot use a reliable transport like TCP
  - **Retransmission** of data to correct for loss would probably cause the packet to **arrive too late** to be useful
  - The application must be able to deal with **missing packets**
  - For example, a video application using MPEG encoding will need to take different actions when a packet is lost
    - Depending on whether the packet came from an **I frame**, a **B frame**, or a **P frame**

Requirements for RTP

- **Congestion-avoidance:** multimedia applications generally **do not run over TCP**
  - Miss out on the congestion-avoidance features of TCP
  - Multimedia applications should **respond to congestion**
    - For example, by changing the parameters of the coding algorithm to **reduce the bandwidth** consumed
  - The receiver needs to notify the sender that **losses** are occurring

Requirements for RTP

- **Frame boundary indication:**
  - Notify a video application that a certain set of packets **correspond to a single frame**
  - Mark the beginning of a **"talkspurt,"** which is a collection of sounds or words followed by **silence**
    - Identify the silences between talkspurts
    - Use them as opportunities to **move the playback point**
    - Slight **shortening** or **lengthening** of the spaces between words are not noticeable to users



Requirements for RTP

- **Identifying senders:** Should be a way more **user-friendly** than an IP address
  - Such as display strings such as Joe User **(user@domain.com)**
- **Efficient use of bandwidth:** **Do not** introduce a lot of extra bits **(long header)** that need to be sent with every packet
  - Long packets would mean **high latency** due to packetization
  - Audio packets tend to be small
    - Bad bandwidth efficiency is obtained if long header is used

RTP Details

- The RTP standard actually defines a pair of protocols
  - **Real-time Transport Protocol (RTP):** is used for the exchange of **multimedia data**
  - **Real-time Transport Control Protocol (RTCP):** is used to periodically send **control information** associated with a certain data flow
- When running over UDP, the RTP data stream and the associated RTCP control stream use **consecutive** transport-layer ports
  - The RTP data uses an **even** port number
  - The RTCP control information uses the **next higher (odd)** port number

RTP Control Protocol

- This **control stream** provides three main functions:
  - To feedback data on the **performance** of the application and the network
  - To **correlate** and **synchronize** different media streams coming from the same sender
  - To convey **the identity of a sender** for display on a user interface
- The performance data is useful for **rate-adaptive** applications
  - Use a more **aggressive compression** scheme to reduce congestion
  - Send a **higher-quality** stream for little congestion