

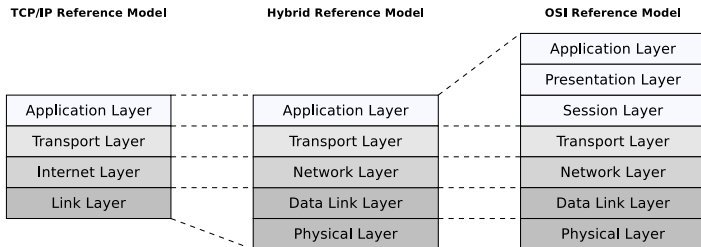


# Learning Objectives of this Slide Set

- Transport Layer
  - Characteristics of Transport Layer protocols
  - Addressing in the Transport Layer
  - User Datagram Protocol (UDP)
    - Format of UDP segments
    - Functioning
  - Transmission Control Protocol (TCP)
    - Format of TCP segments
    - Functioning
    - Flow control
    - Congestion Control
    - Denial-of-service attacks via SYN flood

# Transport Layer

- Functions of the Transport Layer
  - Contains **end-to-end protocols** for inter-process communication
  - In this layer, processes are addressed via **port numbers**
  - Application Layer data is split here into smaller parts **segments**



Exercise sheet 5 repeats the contents of this slide set which are relevant for these learning objectives

- Devices: Gateway
- Protocols: TCP, UDP

## Challenges for Transport Layer Protocols

- The Network Layer protocol IP works **connectionless**
  - IP packets are *routed* independently of each other to the destination site
  - Advantage: Little overhead
- Drawbacks from the Transport Layer perspective
  - IP packets can get **lost** or **discarded** because the TTL has expired
  - IP packets often arrive at the destination site in the **wrong order**
  - **Multiple copies** of IP packets arrive at the destination
- Reasons:
  - Large networks are not static  $\implies$  their infrastructure constantly changes
  - Transmission media can fail
  - The workload varies and therefore the networks' delay
- These problems are common in computer networks
  - Depending on the application, transport protocols need to compensate these drawbacks

# Characteristics of Transport Layer Protocols

- Desired characteristics of Transport Layer protocols include. . .
  - guaranteed data transmission
  - ensuring the correct delivery order
  - support for data transmissions of any size
  - the sender must not overload the network
    - It must be able to adjust its own data flow (data rate)  $\implies$  flow control
  - the receiver should be able to control the transmission rate of the sender  
 $\implies$  congestion control
- Transport Layer protocols are required which convert the networks' negative characteristics into the (positive) characteristics that are expected of Transport Layer protocols
- The most common used Transport Layer protocols:
  - **UDP**
  - **TCP**
- The addressing is realized via **sockets**



## Ports (2/2)

- The port numbers are divided into 3 groups:
    - 0 to 1023 (*Well Known Ports*)
      - These are permanently assigned to applications and commonly known
    - 1024 to 49151 (*Registered Ports*)
      - Application developers can register port numbers in this range for own applications
    - 49152 to 65535 (*Private Ports*)
      - These port numbers are not registered and can be used freely
  - Different applications can use identical port numbers inside an operating system at the same time, if they communicate via different transport protocols
  - In addition, some applications exist, which implement communication via TCP and UDP via a single port number
  - Example: Domain Name System – DNS (see slide set 10)
- 
- Well Known Ports and Registered Ports are assigned by the Internet Assigned Numbers Authority (IANA)
  - In Linux/UNIX systems, the configuration file `/etc/services` exists
    - Here, the applications (services) are mapped to specific port numbers
  - In Windows systems: `%WINDIR%\system32\drivers\etc\services`

# Sockets

- **Sockets** are the platform-independent, standardized **interface** between the implementation of the network protocols in the operating system and the applications
- **A socket consists of a port number and an IP address**
- Stream Sockets and datagram sockets exist
  - **Stream sockets** use the connection-oriented TCP
  - **Datagram sockets** use the connectionless UDP

Tool(s) to monitor the open ports and sockets with...

- Linux/UNIX: `netstat`, `lsof` and `nmap`
- Windows: `netstat`

Alternatives for sockets to implement inter-process communication (IPC)

Pipes, message queues and shared memory  $\implies$  see Operating Systems course



# User Datagram Protocol (UDP)

- **Connectionless Transport Layer protocol**
  - Transmissions take place without previous connection establishment
- More simple protocol in contrast to the connection-oriented TCP
  - Only responsible for addressing of the segments
  - Does not secure the data transmission
- The receiver does not acknowledge transmissions at the sender
  - Segments can get lost during transmission
- Depending on the application (e.g. video streaming) this is accepted
  - If a TCP segment (and therefore some image information) gets lost during the transmission of a video, it is requested again
    - A drawback would be dropouts
  - To compensate for this, a buffer at the receiver site is required
    - Especially video telephony software tries to keep the buffer as small as possible because they cause delays
  - If UDP is used for video transmission or video telephony, the only consequence of losing a segment is losing an image



## Structure of UDP Segments

- The UDP header consists of 4 fields, each of 16 bits size
    - **Port number (sender)**
      - The field can stay empty (value 0), if no response is required
    - **Port number (destination)**
    - **Length** of the complete segment (without pseudo-header)
    - **Checksum** of the complete segment (including pseudo-header)
  - A pseudo-header is created, which includes the IP addresses of sender and destination, as well as some Network Layer information
    - Protocol ID of UDP = 17
  - The pseudo-header is not transmitted
    - But it is used for the checksum calculation
- The diagram illustrates the structure of a UDP segment. It is divided into two main parts: a pseudo-header and the main header/payload.

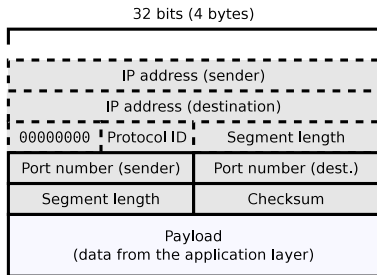
**Pseudo-header (32 bits / 4 bytes):** This section is used for checksum calculation but is not transmitted. It contains:

  - IP address (sender)
  - IP address (destination)
  - 00000000 (Reserved)
  - Protocol ID
  - Segment length

**Main Header:** This section contains the actual header fields of the UDP segment:

  - Port number (sender)
  - Port number (destination)
  - Segment length
  - Checksum

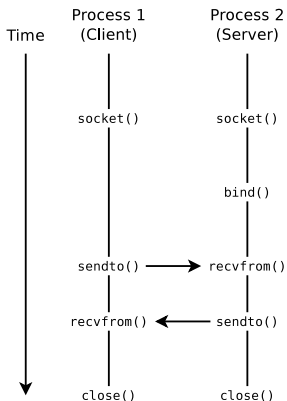
**Payload:** The data from the application, which follows the header.



Remember NAT from slide set 8...

If a NAT device (Router) is used, this routing device also needs to recalculate the checksums in UDP segments when doing IP address translations

## Connectionless Communication via Sockets – UDP



- **Client**

- Create socket (`socket`)
- Send (`sendto`) and receive data (`recvfrom`)
- Close socket (`close`)

- **Server**

- Create socket (`socket`)
- Bind socket to a port (`bind`)
- Send (`sendto`) and receive data (`recvfrom`)
- Close socket (`close`)

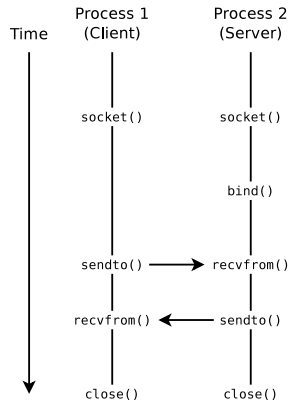
## Sockets via UDP – Example (Server)

```

1 #!/usr/bin/env python
2 # Server: Receives a message via UDP
3
4 import socket                                # Import module socket
5
6 # For all interfaces of the host
7 HOST = ''                                    # '' = all interfaces
8 PORT = 50000                                # Port number of server
9
10 # Create socket and return socket descriptor
11 sd = socket.socket(socket.AF_INET, socket.SOCK_DGRAM)
12
13 try:
14     sd.bind((HOST, PORT))                    # Bind socket to port
15     while True:
16         # Receive data
17         data = sd.recvfrom(1024)
18         # Print out received data
19         print 'Received:', repr(data)
20 finally:
21     sd.close()                                # Close socket

```

```
$ python udp_server.py
```

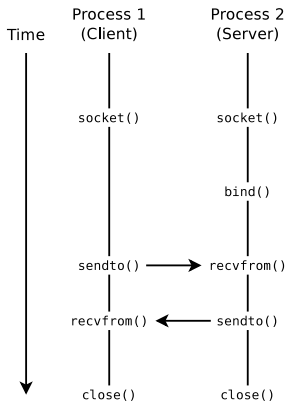


## Sockets via UDP – Example (Client)

```
1 #!/usr/bin/env python
2 # Client: Sends a message via UDP
3
4 import socket                # Import module socket
5
6 HOST = 'localhost'          # Hostname of Server
7 PORT = 50000                 # Port number of Server
8 MESSAGE = 'Hello World'     # Message
9
10 # Create socket and return socket deskriptor
11 sd = socket.socket(socket.AF_INET, socket.SOCK_DGRAM)
12
13 # Send message to socket
14 sd.sendto(MESSAGE, (HOST, PORT))
15
16 sd.close()                   # Close socket
```

```
$ python udp_client.py
```

```
$ python udp_server.py
Received: ('Hello World', ('127.0.0.1', 39834))
```





# Sequence Numbers in TCP

- TCP treats payload as an unstructured, but ordered data stream
- **Sequence numbers** are used for numbering the bytes in the data stream
  - The sequence number of a segment is the position of the segments first byte in the data stream
- Example
  - The sender splits the Application Layer data stream into segments
    - Length of data stream: 5,000 bytes
    - MSS: 1,460 bytes



## Some key figures...

**Maximum Transfer Unit (MTU):** Maximum size of the IP packets

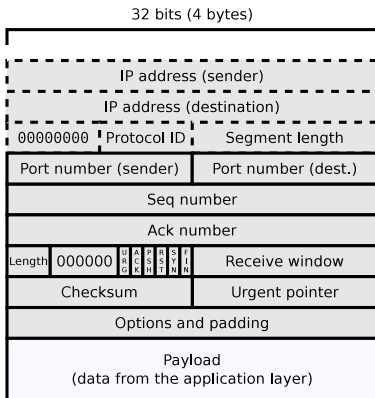
MTU of Ethernet = 1,500 bytes, MTU of PPPoE (e.g. DSL) = 1,492 bytes

**Maximum Segment Size (MSS):** Maximum segment size

MSS = MTU - 40 bytes for IPv4 header and TCP header



## Structure of TCP Segments (1/5)



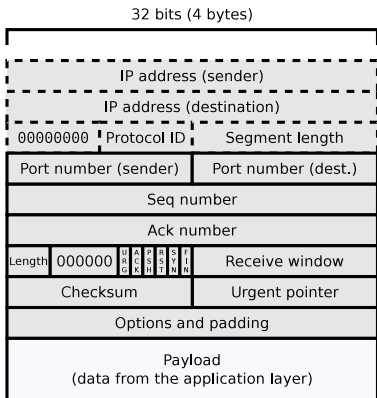
- A TCP segment can contain a maximum of 64 kB payload (data of the Application Layer)
  - Usually, segments are smaller ( $\leq$  1500 bytes for Ethernet)
- The header of TCP segments is more complex compared with UDP segments

## Overhead

- Size of the TCP header (without the options field): just 20 bytes
- Size of the IP header (without the options field): also just 20 bytes

⇒ The overhead, caused by the TCP and IP headers, is small for an IP packet with a size of several kB

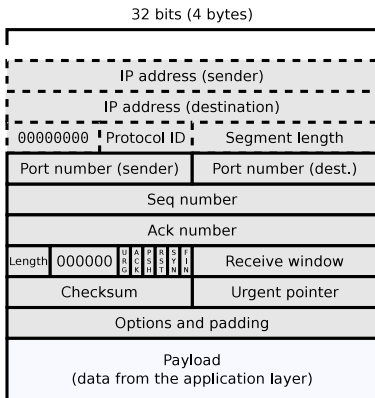
## Structure of TCP Segments (2/5)



- One field contains the port number of the sender process
- Another field contains the port number of the process which is expected to receive the segment
- **Seq number** contains the sequence number of the current segment
- **Ack number** contains the sequence number of the next expected segment

- The **length** field specifies the size of the TCP header in 32-bit words to tell the receiver where the payload starts in the segment
  - The field is required, because the field *options and padding* can have a variable length (a multiple of 32 bits)

## Structure of TCP Segments (3/5)

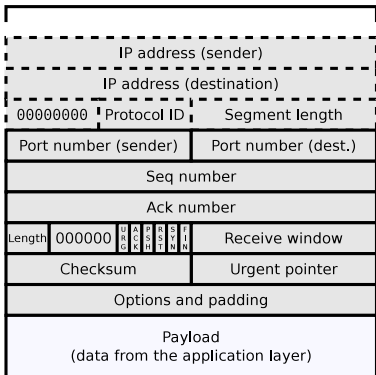
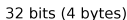


- The field 000000 is 6 bits long and not used
  - It contains per default value zero
- The 6 fields with a size of 1 bit each are required for connection establishment, data exchange and connection termination
  - The functionality of these fields is described with the assumption that the fields contain the value 1  $\implies$  it is set

**URG** (Urgent) is not discussed in this course

- **ACK** (Acknowledge)
  - Specifies that the acknowledgement number in **Ack number** is valid
  - It is also used to acknowledge the receive of segments

## Structure of TCP Segments (4/5)

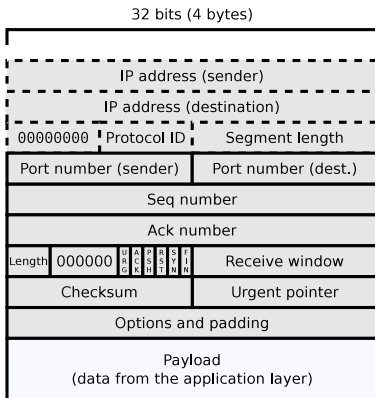


**PSH** (Push) is not discussed in this course

**RST** (Reset) is not discussed in this course

- **SYN** (Synchronize)
  - Requests the synchronization of the sequence numbers
  - That initiates the connection establishment
- **FIN** (Finish)
  - Requests the connection termination and indicates that the sender will not send any more payload

## Structure of TCP Segments (5/5)



- Just as with UDP, for each TCP segment, a pseudo header exists, which is not transmitted
  - But the pseudo header fields are used together with the regular TCP header fields and the payload to calculate the **checksum**
  - **Protocol ID** of TCP = 6

The **urgent pointer** is not discussed in this course

The field **options and padding** must be a multiple of 32 bits and is not discussed in this course

Remember NAT from slide set 8...

If a NAT device (Router) is used, this routing device also needs to recalculate the checksums in TCP segments when doing IP address translations

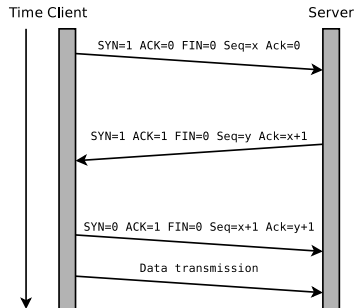
# Functioning of TCP

## You already know...

- Each segment has a unique **sequence number**
- The sequence number of a segment is the position of the segments first byte in the data stream
- The sequence number enables the receiver to...
  - correct the order of the segments
  - sort out segments, which arrived twice
- The length of a segment is known from the IP header
  - This way, missing bytes in the data stream are discovered and the receiver can request lost segments
- To establish a connection, TCP uses a **three-way handshake**, where both communication partners exchange control information in three steps
  - This ensures that the communication partner exists and data transmissions accepts

# TCP Connection Establishment (three-way Handshake)

- The server waits passively for an incoming connection
- ① Client sends a segment with  $\text{SYN}=1$  as a request to synchronize the sequence numbers  
 $\Rightarrow$  *Synchronize*
- ② Server sends as confirmation a segment with  $\text{ACK}=1$  and requests with  $\text{SYN}=1$  to synchronize the sequence numbers too  
 $\Rightarrow$  *Synchronize Acknowledge*
- ③ Client confirms with a segment with  $\text{ACK}=1$  that the connection is established  
 $\Rightarrow$  *Acknowledge*
- The initial sequence numbers ( $x$  and  $y$ ) are determined randomly
- No payload is exchanged during connection establishment!

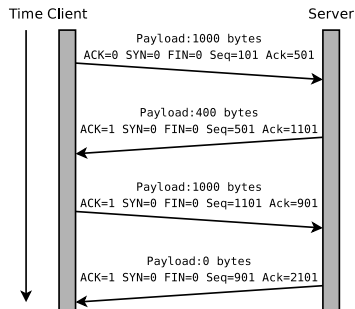


# TCP Data Transmission

To demonstrate a data transmission, **Seq number** (sequence number of the current segment) and **Ack number** (sequence number of the expected next segment) need particular values

- In our example at the beginning of the three-way handshake, the client's sequence number is  $x=100$  and the server's sequence number is  $y=500$
- After completion of the three-way handshake:  $x=101$  and  $y=501$

- 1 The client transmits 1000 bytes payload
- 2 Server acknowledges with  $ACK=1$  the received payload and requests with the Ack number 1101 the next segment. In the same segment, the server transfers 400 bytes of payload
- 3 The client transmits another 1000 bytes payload. And it acknowledges the received payload with the ACK bit set and requests with the Ack number 901 the next segment
- 4 Server acknowledges with  $ACK=1$  the received payload and requests with the Ack number 2101 the next segment

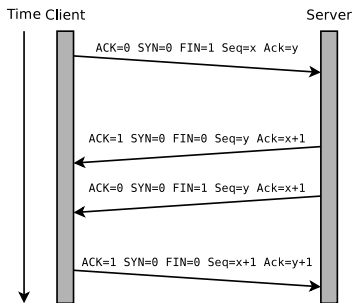




# TCP Connection Termination

- Connection termination is similar to the connection establishment
- Instead of the SYN bit set, the FIN bit is used to indicate that the sender will not transmit any more payload

- 1 The client sends the request for connection termination with FIN=1
- 2 The server sends an acknowledgment with ACK=1
- 3 The server sends the request for connection termination with FIN=1
- 4 The client sends an acknowledgment with ACK=1



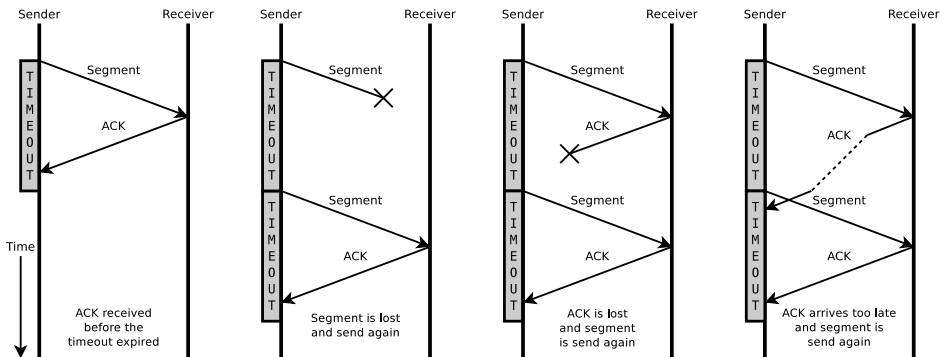
- No payload is exchanged during connection termination!

# Reliable Transmission through Flow Control

- Via Flow control, the receiver controls the **transmission speed** of the sender dynamically, and this way ensures the **completeness of the data transmission**
  - Receivers with a low performance should not be flooded with data they can not process fast enough
    - As result, data would be lost
  - During transmission, lost data is transmitted again
- Procedure: **Transmission retries**, when they are required
- Basic mechanisms:
  - **Acknowledgements** (ACK) as feedback (receipt)
  - **Timeouts**
- Concepts for flow control:
  - **Stop-and-Wait**
  - **Sliding Window**

# Stop-and-Wait

- After transmitting a segment, the sender waits for an ACK
  - If no ACK arrives in a certain time  $\Rightarrow$  timeout
  - Timeout  $\Rightarrow$  segment is sent again



- Drawback: Lesser throughput compared to the transmission-line capacitance

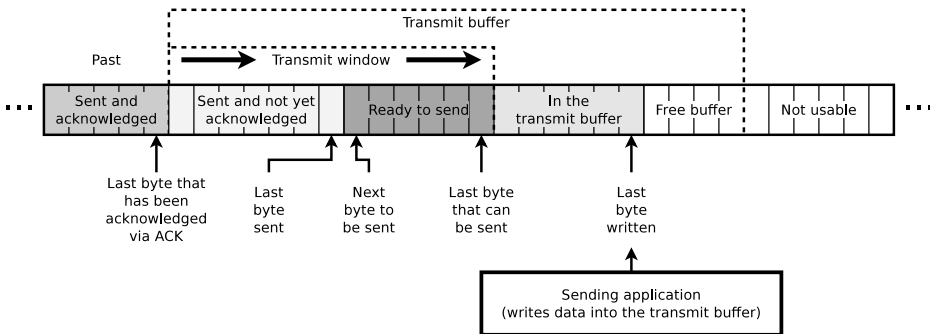
The **Trivial File Transfer Protocol** (RFC 783) operates according to the Stop-and-Wait principle

# Sliding Window

- A **window** allows the sender to transmit a certain number of segments before an acknowledgment is expected
  - Upon arrival of an acknowledgment, the transmit window is moved, and the sender can send further segments
    - The receiver can acknowledge several segments at once  
⇒ **cumulative acknowledgments**
  - If a timeout occurs, the sender transmits all segments in the window again
    - The sender sends everything again beginning from the last not acknowledged sequence number
- Objective: Better utilization of the line capacity and receiver capacity

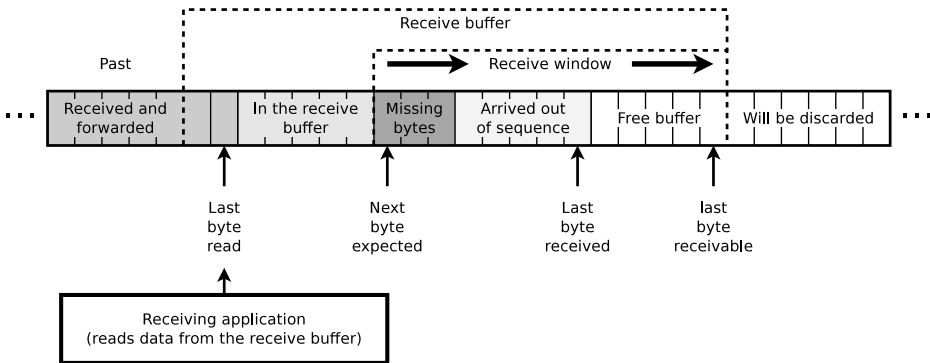
# Sliding Window – Method: Sender

- The transmit buffer contains data of the Application Layer, which...
  - has already been sent but not yet confirmed
  - is ready to be send, but has not been send up to now



## Sliding Window – Method: Receiver

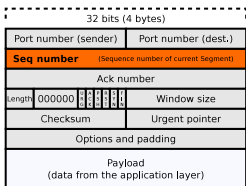
- The receive buffer contains data for the Application Layer, which...
  - is in the correct order, but has not been read
  - has been received out of sequence



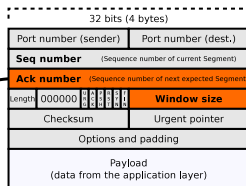
- The receiver informs the sender about the size of its receive window
  - This is important to avoid a buffer overflow!

# TCP Flow Control

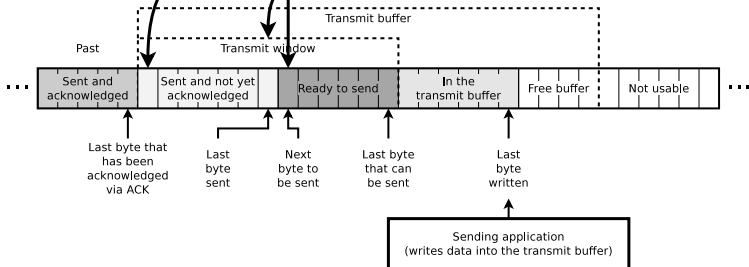
**Segment to be sent**



**Received segment**

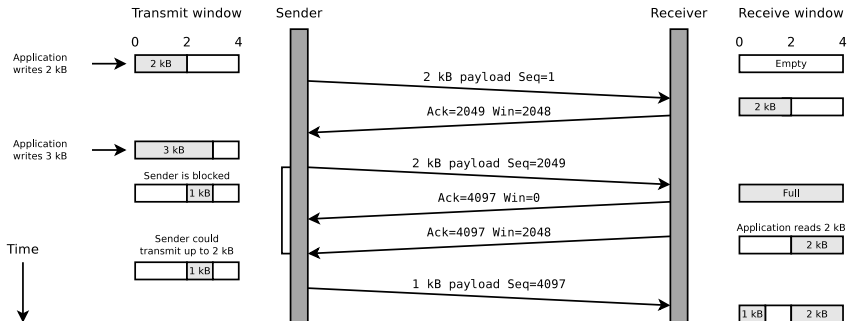


Receiver informs about the number of bytes it is able to receive



# Example of Flow Control in TCP

- The receiver informs the sender in every segment how much free storage capacity its receive window has
- If the receive window has no free capacity, the sender is blocked until it gets informed by the receiver that free storage capacity exists
- If storage capacity in the receive window becomes free  $\Rightarrow$  **A segment with the current free storage capacity is sent**





# Silly Window Syndrome

- The **Silly window syndrome** is a problem where a large number of segments is send, which increases the protocol overhead
  - Scenario
    - A receiver is overloaded and its receive buffer is completely filled
    - Once the application has read a few bytes (e.g. 1 byte) from the receive buffer, the receiver sends a segment with the free storage capacity of the receive buffer
    - For this reason, the sender transmits a segment, which contains just 1 byte payload
    - Overhead: At least 40 bytes for the TCP/IP headers of each IP packet (Required are: 1 segment with the payload, 1 segment for the acknowledgement and eventually another segment which notifies about the current free storage capacity in the receive window)
  - Solution: **Silly window syndrome avoidance**
    - The receiver notifies the sender about free storage capacity in the receive window not before 25% of the receive buffer is free or a segment of size MSS can be received

# Reasons why Congestion occurs

- Possible reasons for the occurrence of congestion:
  - ① **Receiver capacity**
    - The receiver can not process the received data fast enough and therefore its receive buffer becomes full
    - Already solved by **flow control**
  - ② **Network capacity**
    - Congestion (overload) occurs when the utilization of a computer network exceeds its capacity  $\Rightarrow$  **congestion control**
    - Only useful reaction to congestion: **Reduce the data rate**
    - TCP tries to avoid congestion by changing the window size dynamically  $\Rightarrow$  **dynamic sliding window**
- *The one solution*, which solves both causes does not exist
  - Both causes are addressed separately

## Signs of congestion of the network

- Packet losses due to buffer overflows in Routers
- Long waiting times due to full queues in Routers
- Frequent retransmissions due to timeout or packet-/segment loss

# Approach to avoid Congestion

- The sender maintains 2 windows

## ① Advertised Receive Window

- Avoids congestion of the receiver
- Offered (*advertised*) by the receiver

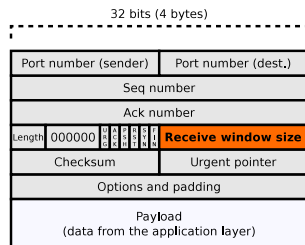
## ② Congestion Window

- Avoids congestion of the network
- Determined by the sender

- The minimum of both windows is the maximum number of bytes, the sender can transmit

- Example:

- If the receive window of the receiver has a free storage capacity of 20 kB, but the sender recognizes that a network congestion occurs when more than 12 kB are sent, it transmits only 12 kB



- How does the sender know the capacitance of the network?

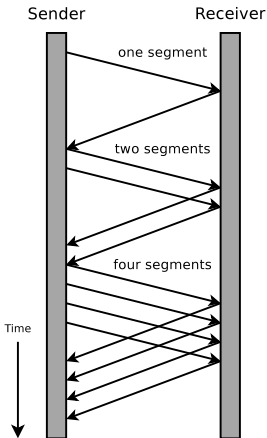
⇒ how does the sender determine the size of the congestion window?

# Determine the Size of the Congestion Window

## You already know...

- The sender can exactly specify the size of the receive window
  - Reason: The receiver informs the sender with every segment, about the free storage capacity of its receive window
- 
- Challenge for the sender: **What is the size of the congestion window?**
    - The sender never knows for sure the capacity of the network
    - The capacity of computers networks is not static
      - It depends among others of the network utilization and of the occurrence of network faults
  - Solution: The sender must **incrementally** try to identify the network capacity

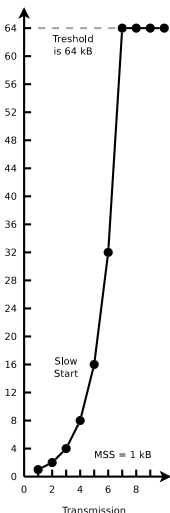
## Determine the Congestion Window Size – Connection Establishment



- During connection establishment, the sender initializes the congestion window to maximum segment size (MSS)
- Method:
  - 1 segment of size MSS is sent
    - If the segment is acknowledged before the timeout expires, the congestion window is doubled
  - 2 segments of size MSS are sent
    - If both segments are acknowledged before the timeout expires, the congestion window is doubled again
  - ...

## Determine the Congestion Window Size – Slow Start

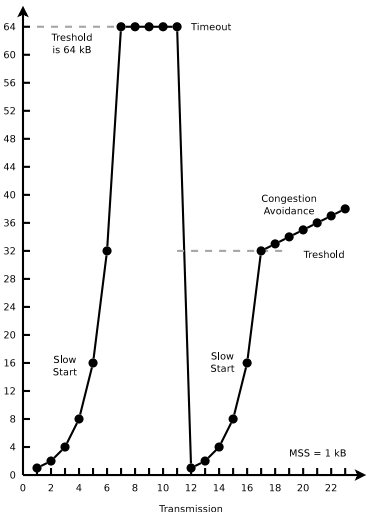
Congestion Window [kB]



- The congestion window grows exponentially until...
  - the size of the receive window is reached, which has been determined by the sender
  - or the **threshold** is reached
  - or a **timeout** expires
- The exponential growth phase is called **slow start**
  - Reason: The low transmission rate of the sender at the beginning
- If the congestion window reaches the size of the receive window, it stops growing
- At the beginning of the transmission, the threshold value is  $2^{16}$  bytes = 64 kB, so that it plays no role at the beginning
  - Maximum size of the receive window:  $2^{16} - 1$  bytes
    - This is determined by the size of the field **window size** in the TCP header

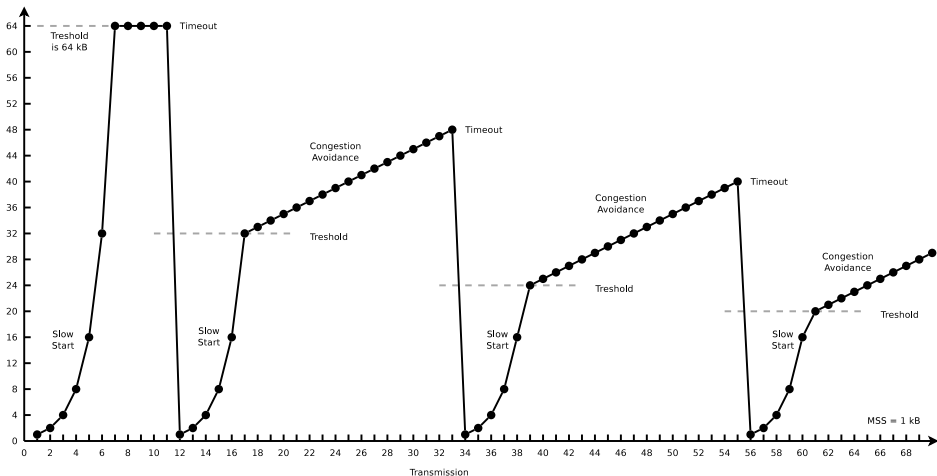
# Determine the Congestion Window Size – Congestion Avoidance

Congestion Window [kB]



- If a timeout expires, . . .
  - the threshold value is set to the half congestion window
  - and the size of the congestion window is reduced to the size 1 MSS
- Then, once again the slow start phase follows
  - If the threshold value is reached, the congestion window grows linear, . . .
    - until the size of the receive window is reached, which is determined by the receiver
    - or until a timeout expires
- The linear growth phase is called **congestion avoidance**

Congestion Window [kB]

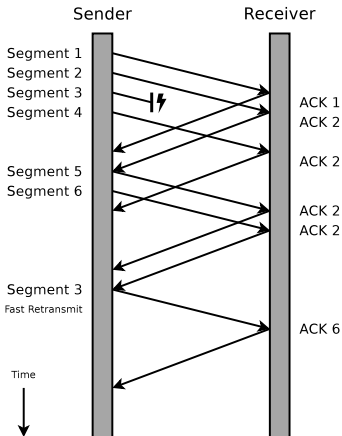




# Reasons why a Timeout expires and reasonable Proceeding

- An expired **timeout** can have different reasons
  - Congestion ( $\implies$  delay)
  - Loss of a transmission
  - Loss of an acknowledgment (ACK)
- Not only delays due to congestion, but also each loss event reduces the congestion window to size 1 MSS
  - This way works the obsolete TCP version *Tahoe* (1988)
- Modern TCP versions differ between...
  - expired timeout caused by congestion of the network
  - and **multiple arrival of acknowledgments** (ACKs) caused by loss event

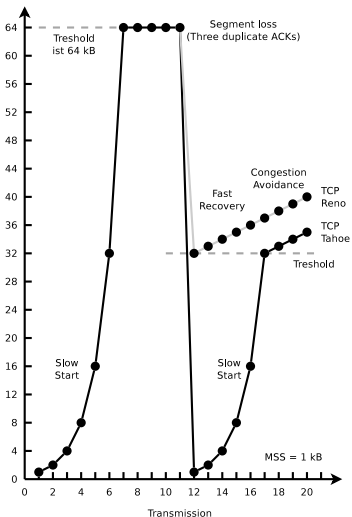
# Fast Retransmit



- A lost segment causes a *gap* in the data stream at receiver site
  - The receiver sends for every additional received segment an ACK for the segment before (the lost segment!)
- If a segment gets lost, a reduction of the congestion window to value 1 MSS is not necessary
  - Reason: A segment loss is not caused by congestion in any case
- If 3 duplicate ACKs arrive, TCP *Reno* (1990) sends the lost segment again  
⇒ **fast retransmit**

# Fast Recovery

Congestion Window [kB]



- TCP *Reno* also avoids the slow start phase if 3 duplicate ACKs arrive  
⇒ **fast recovery**
- If 3 duplicate ACKs arrive, the congestion window is set directly to the threshold value
  - The congestion window grows linear with every acknowledged transmission, . . .
    - until the size of the receive window is reached, which is specified by the receiver
    - or until a timeout expires

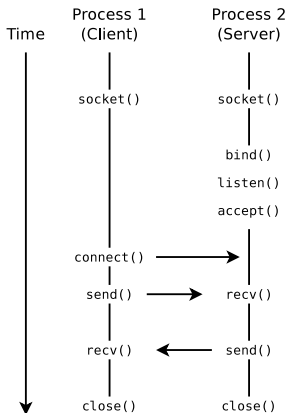
# Additive Increase / Multiplicative Decrease (AIMD)

- The concept of TCP congestion control is called AIMD
  - It stands for **rapid reduction** of the congestion window after a timeout expired or a loss event occurred and **slow (linear) increase** of the congestion window
- Reason for **aggressive reduction** and **conservative increase** of the congestion window:
  - The consequences of a congestion window which is too large in size are worse than for a window which is too small
  - If the window is too small in size, available bandwidth remains unused
  - If the window is too large in size, segments will get lost and must be transmitted again
    - This increases the congestion of the network even more!
- The state of congestion must be left as quick as possible
  - Therefore, the size of the congestion window is reduced significantly

# Summary of Flow Control and Congestion Control

- By using **flow control**, TCP tries to use the available bandwidth of a connectionless network ( $\implies$  IP) efficiently
  - Sliding windows at sender site (**transmit window**) and receiver site (**receive window**) are used as buffers for sending and receiving
  - The receiver controls the transmission behavior of the sender
- Reasons why congestion happens: **receiver capacity** and **network capacity**
  - The receive window avoids congestion of the receiver
  - The congestion window avoids congestion of the network
  - Actual available (used) window = minimum of both windows
- Attempt to maximize the network utilization and react rapidly to indications for congestion
  - Principle of **Additive Increase / Multiplicative Decrease** (AIMD)

# Connection-oriented Communication via Sockets – TCP



## • Client

- Create socket (`socket`)
- Connect client with server socket (`connect`)
- Send (`send`) and receive data (`recv`)
- Close socket (`close`)

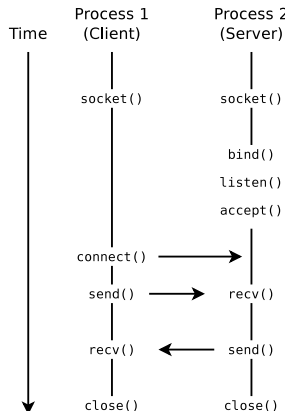
## • Server

- Create socket (`socket`)
- Bind socket to a port (`bind`)
- Make socket ready to receive (`listen`)
  - Set up a queue for connections with clients
- Server accepts connections (`accept`)
- Send (`send`) and receive data (`recv`)
- Close socket (`close`)

# Sockets via TCP – Example (Server)

```
1 #!/usr/bin/env python
2 # Echo Server via TCP
3 import socket                # Import module socket
4
5 HOST = ''                    # '' = all interfaces
6 PORT = 50007                 # Port number of server
7
8 # Create socket and return socket descriptor
9 sd = socket.socket(socket.AF_INET, socket.SOCK_STREAM)
10 # Bind socket to port
11 sd.bind((HOST, PORT))
12 # Make socket ready to receive
13 # Max. number of connections = 1
14 sd.listen(1)
15 # Socket accepts connections
16 conn, addr = sd.accept()
17
18 print 'Connected by', addr
19
20 while 1:                     # Infinite loop
21     data = conn.recv(1024)    # Receive data
22     if not data: break        # Break infinite loop
23     conn.send(data)           # Send back received data
24
25 conn.close()                 # Close socket
```

```
$ python tcp_server.py
```



## Sockets via TCP – Example (Client)

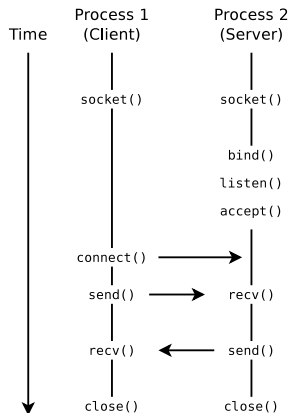
```

1 #!/usr/bin/env python
2 # Echo Client via TCP
3
4 import socket                # Import module socket
5
6 HOST = 'localhost'          # Hostname of Server
7 PORT = 50007                 # Port number of server
8
9 # Create socket and return socket descriptor
10 sd = socket.socket(socket.AF_INET, socket.SOCK_STREAM)
11 # Connect with server socket
12 sd.connect((HOST, PORT))
13
14 sd.send('Hello, world')      # Send data
15 data = sd.recv(1024)         # Receive data
16 sd.close()                   # Close socket
17
18 # Print out received data
19 print 'Empfangen:', repr(data)

```

```
$ python tcp_client.py
Empfangen: 'Hello, world'
```

```
$ python tcp_server.py
Connected by ('127.0.0.1', 49898)
```





# Denial-of-Service Attacks via SYN Flood

- Target: Making services or servers inaccessible
- A client sends multiple connection requests (**SYN**), but does not respond to the acknowledgments (**SYN ACK**) of the server via **ACK**
- The server waits some time for the acknowledgment of the client
  - The confirmation delay could be caused by a network issue
  - During this period, the address of the client and the status of incomplete connection are stored in the memory of the network stack
- By flooding the server with connection requests, the table, which stores the TCP connections in the network stack is completely filled  
 ⇒ the server gets unable to establish new connections
- The memory consumption at the server may become this large that the main memory gets completely filled and the server crashes
- Countermeasure: Real-time analysis of the network by intelligent firewalls