9th Slide Set
Computer Networks

Prof. Dr. Christian Baun

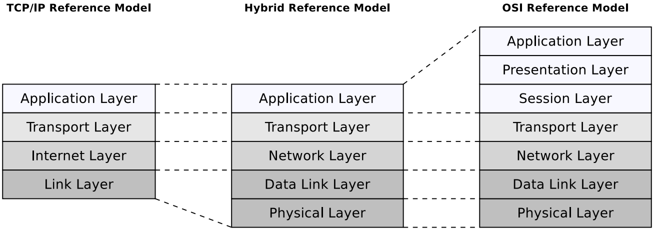
Frankfurt University of Applied Sciences
(1971–2014: Fachhochschule Frankfurt am Main)
Faculty of Computer Science and Engineering
christianbaun@fb2.fra-uas.de

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**



- 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 $\Longrightarrow$ 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 receiver should be able to control the transmission rate of the sender $\Longrightarrow$ flow 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**

Addressing in the Transport Layer

- Every application, which uses TCP or UDP, has a **port number** assigned
  - It specifies, which service is accessed
  - For TCP and UDP, the size of port numbers is 16 bits
    - Thus, the range of possible port numbers is from 0 to 65,535
- In principle, port numbers can be assigned as wished
  - Conventions exist, that specify, which ports are used by common used applications

Port number	Service	Description
21	FTP	File transfer
22	SSH	Encrypted terminal emulation (secure shell)
23	Telnet	Terminal emulation for remote control of computers
25	SMTP	E-mail transfer
53	DNS	Resolution of domain names into IP addresses
67	DHCP	Assignment of the network configuration to clients
80	HTTP	Webserver
110	POP3	Client access to E-mail server
143	IMAP	Client access to E-mail server
443	HTTPS	Webserver (encrypted)
993	IMAPS	Client access to E-mail server (encrypted)
995	POP3S	Client access to E-mail server (encrypted)

- The table contains only a small selection of well-known port numbers

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)

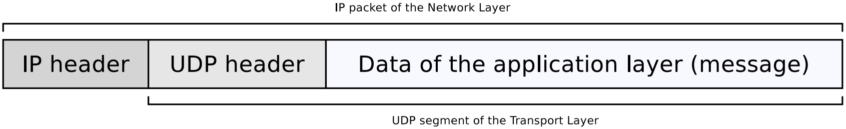
Pipes, message queues and shared memory $\Longrightarrow$ see Operating Systems course

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

User Datagram Protocol (UDP)

- Maximum size of an UDP segment: 65,535 Bytes
  - Reason: The size of the **length** field inside the UDP header, which contains the segment length, is 16 bits
    - The maximum representable number with 16 bits is 65,535
  - UDP segments of this size are transmitted fragmented by IP

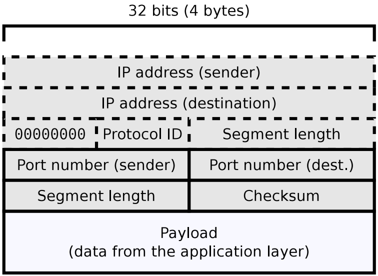


UDP standard: RFC 768 from 1980
`http://tools.ietf.org/rfc/rfc768.txt`

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)

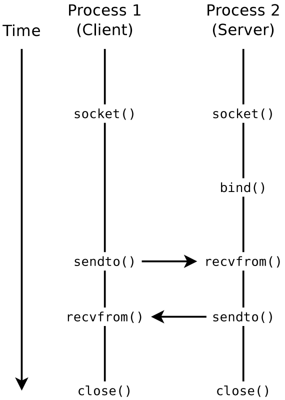
- A pseudo-header is created, which includes the IP addresses of sender and destination, as well as some Network Layer information
  - Protocol ID of UDP = 17
- The pseudo-header is not transmitted
  - But it is used for the checksum calculation



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

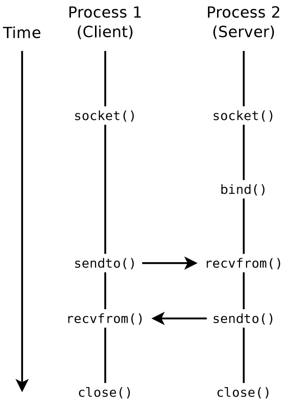


- **Client**
  - Create socket (`socket`)
  - Send (`sendto`) and receive data (`recvfrom`)
  - Close socket (`close`)
- **Server**
  - Create socket (`socket`)
  - Bind socket to a port (`bind`)
  - Send (`sendto`) and receive data (`recvfrom`)
  - Close socket (`close`)

Sockets via UDP – Example (Server)

```
#!/usr/bin/env python
# Server: Receives a message via UDP

import socket                    # Import module socket

# For all interfaces of the host
HOST = ''                        # '' = all interfaces
PORT = 50000                     # Port number of server

# Create socket and return socket deskriptor
sd = socket.socket(socket.AF_INET, socket.SOCK_DGRAM)

try:
  sd.bind((HOST, PORT))          # Bind socket to port
  while True:
    # Receive data
    data = sd.recvfrom(1024)
    # Print out received data
    print 'Received:', repr(data)
finally:
  sd.close()                     # Close socket
```

```
$ python udp_server.py
```

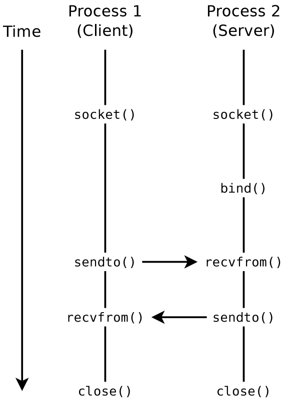


Sockets via UDP – Example (Client)

```
#!/usr/bin/env python
# Client: Sends a message via UDP

import socket                  # Import module socket

HOST = 'localhost'             # Hostname of Server
PORT = 50000                   # Port number of Server
MESSAGE = 'Hello World'        # Message

# Create socket and return socket deskriptor
sd = socket.socket(socket.AF_INET, socket.SOCK_DGRAM)

# Send message to socket
sd.sendto(MESSAGE, (HOST, PORT))

sd.close()                     # Close socket
```

```
$ python udp_client.py
```

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



Transmission Control Protocol (TCP)

- **Connection-oriented Transport Layer protocol**
- Makes connections via IP reliable in a way, which is desired or required for many applications
- Ensures that segments reach their destination **completely** and in the **correct order**
  - Lost or unacknowledged TCP segments are requested by the receiver at the sender and sent again
- TCP connections are opened and closed like files
  - Equal to files, the position in the data stream is exactly specified

TCP standard: RFC 793 from 1981
`http://tools.ietf.org/rfc/rfc793.txt`

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

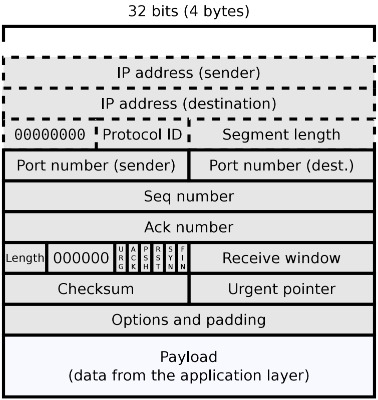
HEADER	Segment 1	HEADER	Segment 2	HEADER	Segment 3	HEADER	Segment 4
	0 ... 1.459		1.460 ... 2.919		2.920 ... 4.379		4.380 ... 4.999
	Sequence number: 0		Sequence number: 1.460		Sequence number: 2.920		Sequence number: 4.380

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 Segement 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 ($\leq$ 1,500 bytes for Ethernet)



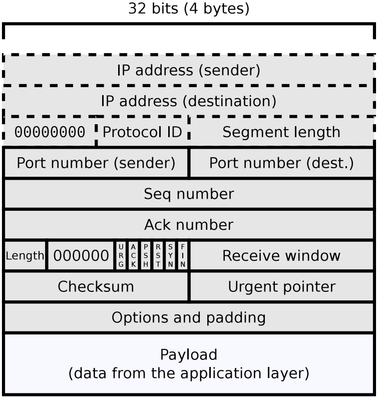
- The header of TCP segments is more complex compared with UDP segments

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

$\Longrightarrow$ 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)

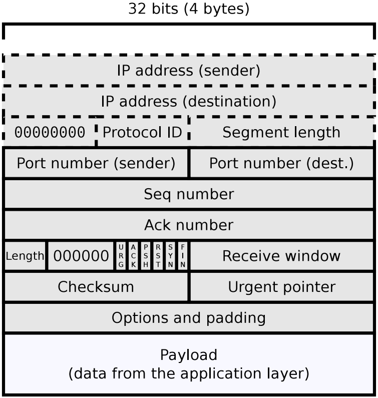


- One field contains the port number of the sender process
- Another field contains the port number of the process which is expected to receive the segment
- **Seq number** contains the sequence number of the current segment
- **Ack number** contains the sequence number of the next expected segment

- 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

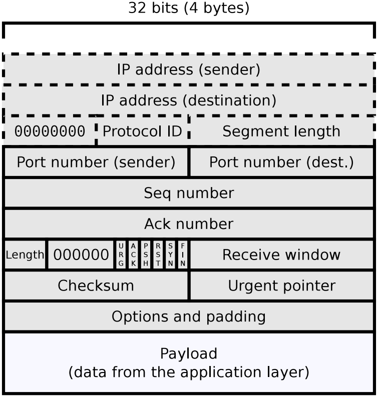


- The 6 fields with a size of 1 bit each are required for connection establishment, data exchange and connection termination
  - The functionality of these fields is described with the assumption that the fiels contain the value 1 $\Longrightarrow$ it is *set*

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)



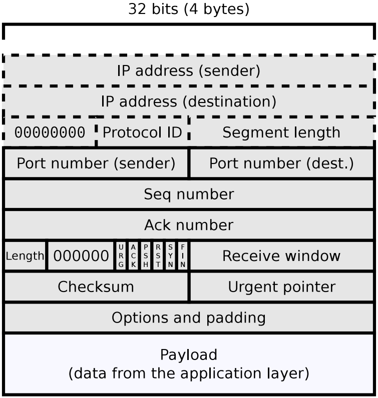
PSH (Push) is not discussed in this course

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

- The field **receive window** contains the number of free bytes in the sender's receive window, which is necessary for flow control

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. . .

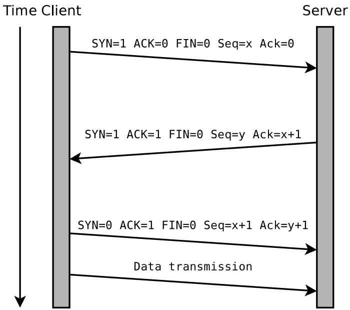
- Each segment has a unique **sequence number**
- The sequence number of a segment is the position of the segments first byte in the data stream

- 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 SYN=1 as a request to synchronize the sequence numbers
   $\Longrightarrow$ *Synchronize*
2. Server sends as confirmation a segment with ACK=1 and requests with SYN=1 to synchronize the sequence numbers too
   $\Longrightarrow$ *Synchronize Acknowledge*
3. Client confirms with a segment with ACK=1 that the connection is established
   $\Longrightarrow$ *Acknowledge*



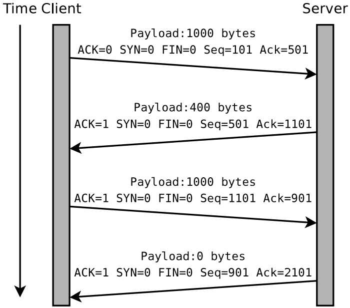
- The initial sequence numbers (x and y) are determined randomly
- No payload is exchanged during connection establishment!

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

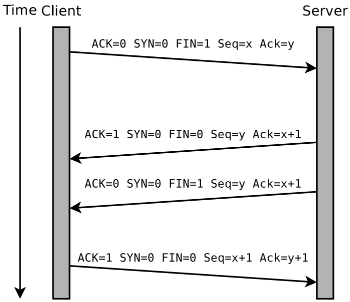
1. The client transmits 1000 bytes payload
2. Server acknowledges with ACK=1 the received payload and requests with the Ack number 1101 the next segment. In the same segment, the server transfers 400 bytes of payload
3. The client transmits another 1000 bytes payload. And it acknowledges the received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



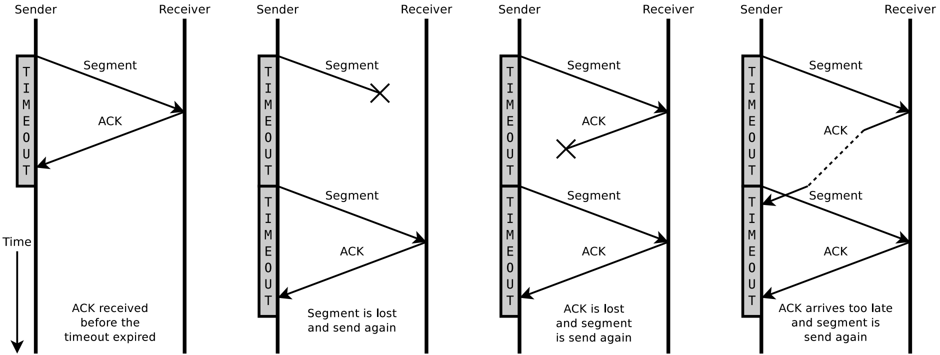
- No payload is exchanged during connection termination!

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 $\Longrightarrow$ timeout
  - Timeout $\Longrightarrow$ segment is sent again



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

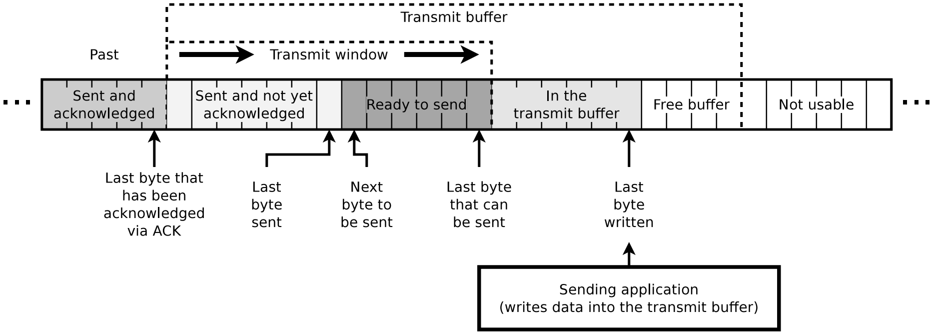
The **Trivial File Transfer Protocol** (RFC 783) operates according to the Stop-and-Wait principle

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 $\Longrightarrow$ **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

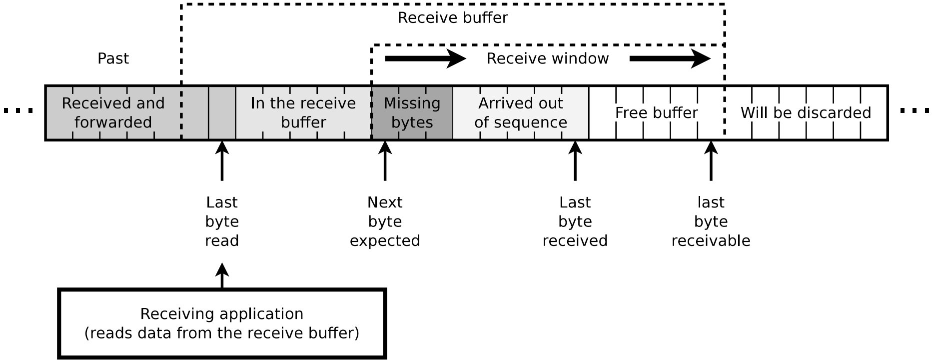
Sliding Window – Method: Sender

- 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



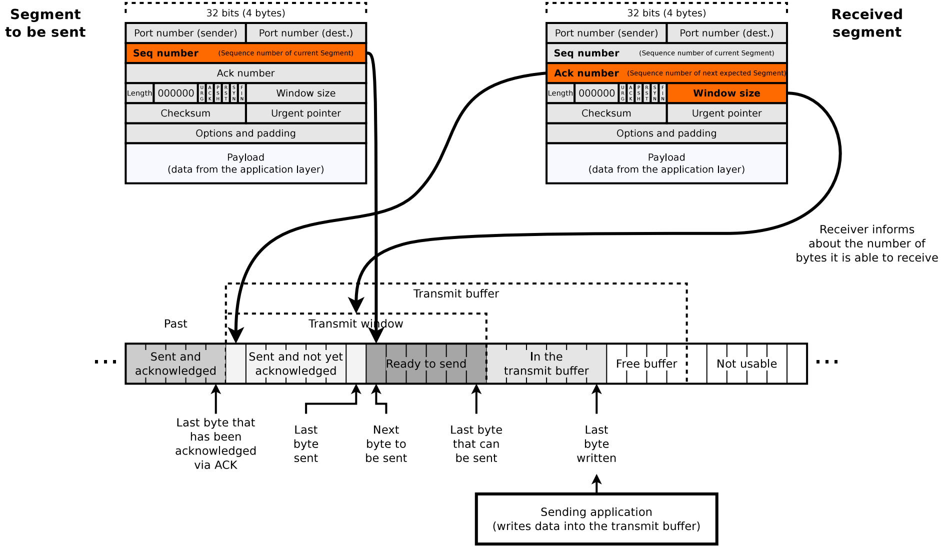
Sliding Window – Method: Receiver

- 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



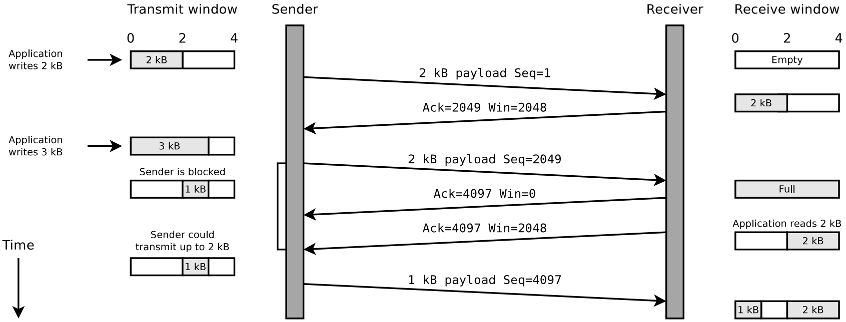
- The receiver informs the sender about the size of its receive window
  - This is important to avoid a buffer overflow!

TCP Flow Control



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 $\Longrightarrow$ **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

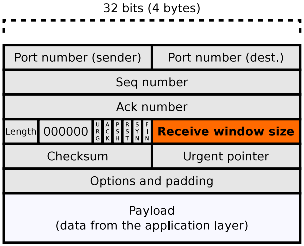
- Possible reasons for the occurrence of congestion:
  1. **Receiver capacity**
     - The receiver can not process the received data fast enough and therefore its receive buffer becomes full
     - Already solved by **flow control**
  2. **Network capacity**
     - Congestion (overload) occurs when the utilization of a computer network exceeds its capacity $\Longrightarrow$ **congestion control**
     - Only useful reaction to congestion: **Reduce the data rate**
     - TCP tries to avoid congestion by changing the window size dynamically $\Longrightarrow$ **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 that 12 kB are sent, it transmits only 12 kB

- How does the sender know the capacitance of the network?
  $\Longrightarrow$ how does the sender determine the size of the congestion window?

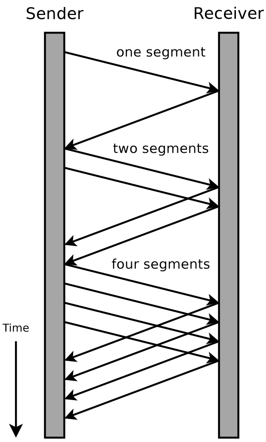
Determine the Size of the Congestion Window

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

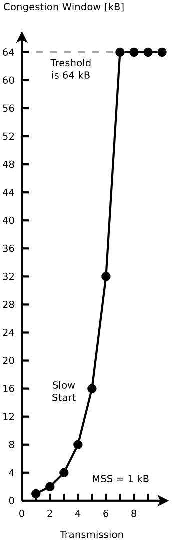
- Challenge for the sender: **What is the size of the congestion window?**
  - The sender never knows for sure the capacity of the network
  - The capacity of computers networks is not static
    - It depends among others of the network utilization and of the occurrence of network faults
- Solution: The sender must **incrementally** try to identify the network capacity

Determine the Congestion Window Size – Connection Establishment



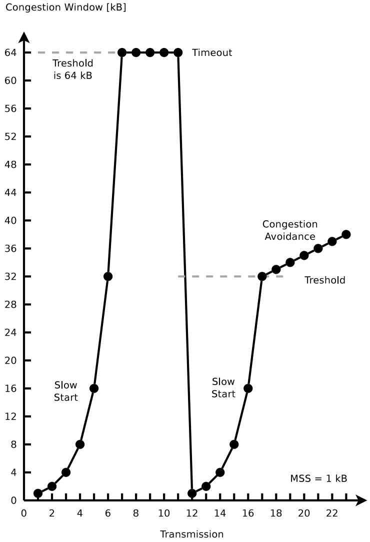
- During connection establishment, the sender initializes the congestion window to maximum segment size (MSS)
- Method:
  - 1 segment of size MSS is sent
    - If the segment is acknowledged before the timeout expires, the congestion window is doubled
  - 2 segments of size MSS are sent
    - If both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

Determine the Congestion Window Size – Slow Start



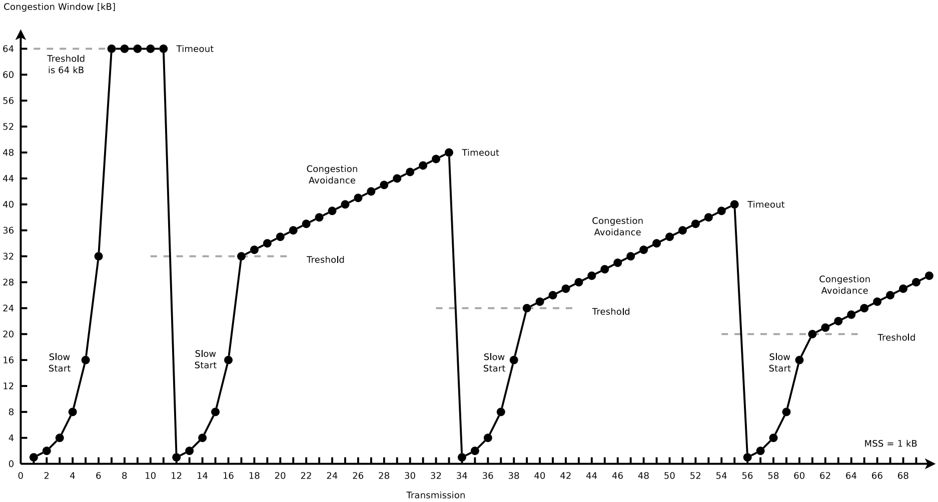
- The congestion window grows exponentially until...
  - the size of the receive window is reached, which has been determined by the sender
  - or the **threshold** is reached
  - or a **timeout** expires
- The exponential growth phase is called **slow start**
  - Reason: The low transmission rate of 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 $= 64\,\text{kB}$, 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



- 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**

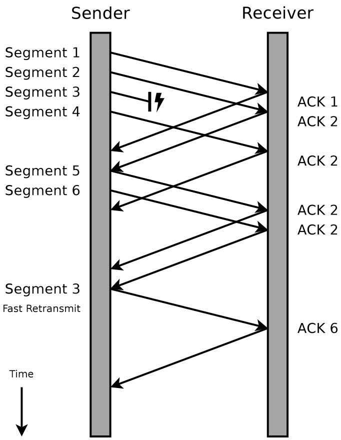
Possible Continuation of the Example



Reasons why a Timeout expires and reasonable Proceeding

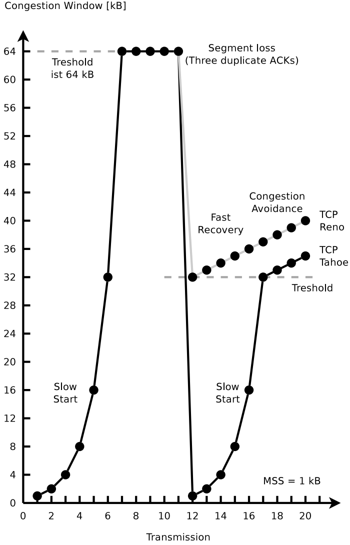
- An expired **timeout** can have different reasons
  - Congestion ($\Longrightarrow$ delay)
  - Loss of a transmission
  - Loss of an acknowledgment (ACK)
- Not only delays due to congestion, but also each loss event reduces the congestion window to size 1 MSS
  - This way works the obsolete TCP version *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
- After 3 duplicate ACKs arrived, TCP *Reno* (1990) sends the lost segment again $\Longrightarrow$ **fast retransmit**

Fast Recovery



- TCP *Reno* also avoids the slow start phase after 3 duplicate ACKs arrived $\Longrightarrow$ **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

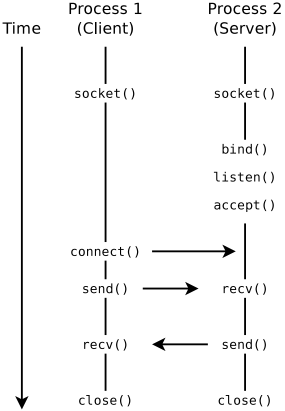
Additive Increase / Multiplicative Decrease (AIMD)

- 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 ($\Longrightarrow$ 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)

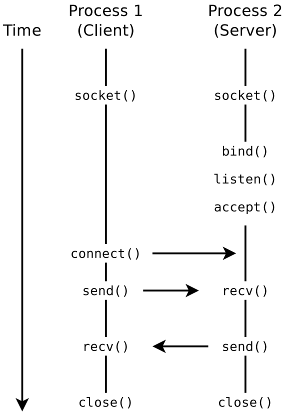
Connection-oriented Communication via Sockets – TCP



- **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)

```python
#!/usr/bin/env python
# Echo Server via TCP
import socket                 # Import module socket

HOST = ''                     # '' = all interfaces
PORT = 50007                  # Port number of server

# Create socket and return socket deskriptor
sd = socket.socket(socket.AF_INET, socket.SOCK_STREAM)
# Bind socket to port
sd.bind((HOST, PORT))
# Make socket ready to receive
# Max. number of connections = 1
sd.listen(1)
# Socket accepts connections
conn, addr = sd.accept()

print 'Connected by', addr

while 1:                      # Infinite loop
    data = conn.recv(1024)    # Receive data
    if not data: break        # Break infinite loop
    conn.send(data)           # Send back received data

conn.close()                  # Close socket
```



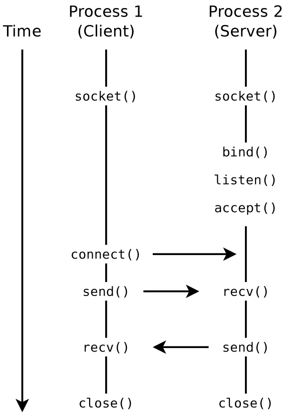

```
$ python tcp_server.py
```

Sockets via TCP – Example (Client)

```python
#!/usr/bin/env python
# Echo Client via TCP

import socket                    # Import module socket

HOST = 'localhost'               # Hostname of Server
PORT = 50007                     # Port number of server

# Create socket and return socket deskriptor
sd = socket.socket(socket.AF_INET, socket.SOCK_STREAM)
# Connect with server socket
sd.connect((HOST, PORT))

sd.send('Hello, world')          # Send data
data = sd.recv(1024)             # Receive data
sd.close()                       # Close socket

# Print out received data
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
  $\Longrightarrow$ 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