Computer Networks

Transport Layer

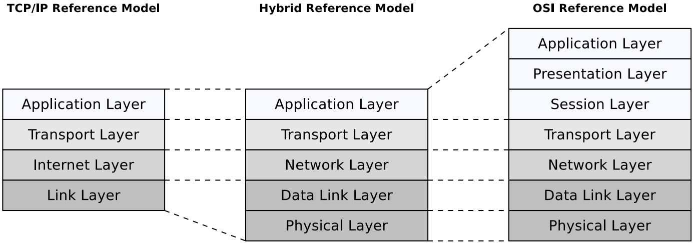
Prof. Dr. Oliver Hahm

Frankfurt University of Applied Sciences
Faculty 2: Computer Science and Engineering
oliver.hahm@fb2.fra-uas.de
https://teaching.dahahm.de

January 13, 2023

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, QUIC

Challenges for Transport Layer Protocols

- The Network Layer protocol IP works connectionless and best effort
  - IP packets are *routed* independently of each other to the destination site
  - Advantage: Simple, little overhead
- Drawbacks from the user/application 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...
  - Multiplexing/demultiplexing of multiple services on one host
  - guaranteed data transmission $\longrightarrow$ end-to-end reliability control
  - ensuring the correct delivery order
  - support for data transmissions of any size
  - the sender must not overload the receiver $\longrightarrow$ end-to-end flow control
  - the sender must not overload the network $\longrightarrow$ congestion control

Addressing in the Transport Layer

- Every application using a transport layer service has a port number assigned
  - The port specifies which service is accessed
  - For TCP and UDP the size of port numbers is 16 bits $\Longrightarrow$ the range of possible port numbers is from 0 to 65,535
- Port numbers[1] can be grouped into ...
  - 0 – 1023: well-known ports or *system ports*
    - These are permanently assigned to applications and commonly known
  - 1024 – 49151: registered ports or *user ports*
    - Application developers can register port numbers in this range for own applications
  - 49152 – 65535: ephemeral ports or *private ports*
    - These port numbers are not registered and can be used freely

[1] https://www.iana.org/assignments/service-names-port-numbers/service-names-port-numbers.xhtml

Well-known Port Numbers

A small selection of well-known port numbers:

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)

- Well-known ports and registered ports are assigned by the IANA
- In Linux/UNIX systems: `/etc/services`
- 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

Tools to monitor the open ports and sockets with...

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

Agenda

- TCP
  - Basics and Structure
  - Functioning of TCP
  - Flow Control
  - Congestion Control
  - Enhancements
  - Connection-oriented Communication via Sockets
  - Denial-of-Service Attacks via SYN Flood
- UDP
  - Basics
  - Connectionless Communication via Sockets
- Other Transport Layer Protocols
  - SCTP
  - DCCP
  - QUIC

Agenda

- TCP
  - Basics and Structure
  - Functioning of TCP
  - Flow Control
  - Congestion Control
  - Enhancements
  - Connection-oriented Communication via Sockets
  - Denial-of-Service Attacks via SYN Flood
- UDP
  - Basics
  - Connectionless Communication via Sockets
- Other Transport Layer Protocols
  - SCTP
  - DCCP
  - QUIC

Agenda

Transmission Control Protocol (TCP)

- Connection-oriented transport layer protocol
- Enables connections over IP in a reliable way
- 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 specification: RFC 793 from 1981
`http://tools.ietf.org/rfc/rfc793.txt`

Sequence Numbers in TCP

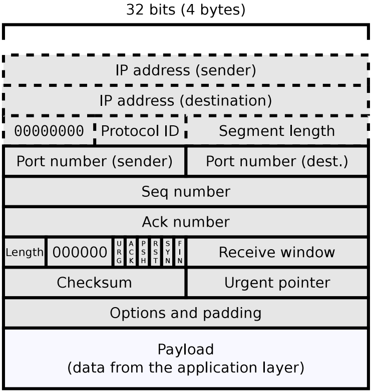
- TCP treats payload as an unstructured, but ordered data stream
- Sequence numbers are used for numbering the bytes in the data stream
  - The sequence number of a segment is the position of the segments first byte in the data stream
- Example
  - The sender splits the application layer data stream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

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

MTU vs. MSS

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)



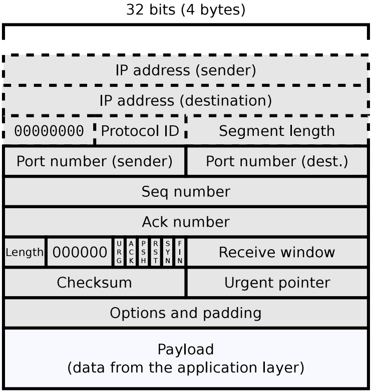
- A TCP segment can contain a maximum of 64 kB payload (data of the Application Layer)
  - Usually, segments are smaller ($\leq$ 1500 bytes for Ethernet)

Overhead

- Size of the TCP header (without the options field): just 20 bytes
- Size of an IPv4 header (without the options field): also just 20 bytes
- Size of an IPv6 header (without extension headers): 40 bytes

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

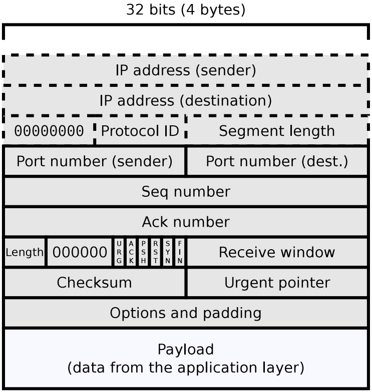
Structure of TCP Segments (2/5)



- The first field contains the source port number (sender process)
- The next field contains the destination port number (receiver process)
- Seq number contains the sequence number of the current segment
- ACK number contains the sequence number of the next expected segment

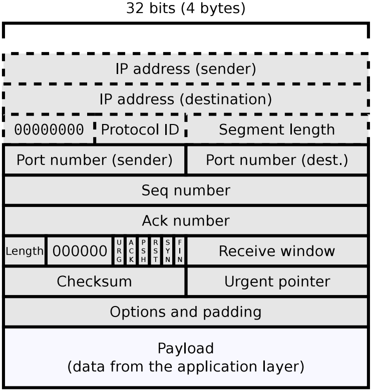
- The length (or data offset) field specifies the size of the TCP header in 32-bit words to tell the receiver where the payload starts in the segment
  - The field is required, because the field options and padding can have a variable length (a multiple of 32 bits)

Structure of TCP Segments (3/5)



- The next field contains 6 bits and is *reserved*, it must contain 000000
- The following six fields contain the Control Bits
- They are required for connection establishment, data exchange, and connection termination
  - The described functionality for these bits assume them to be *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 reception of segments
  - Should be always set after *SYN*

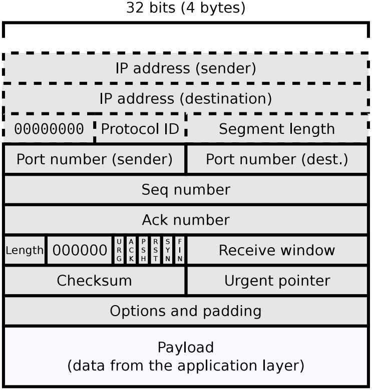
Structure of TCP Segments (4/5)



- 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
  - For 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)



- A pseudo-header is created (but not transmitted), which includes the IP addresses of sender and destination, as well as some Network Layer information
  - 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 fields options and padding must be a multiple of 32 bits and are 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

Agenda

- TCP
  - Basics and Structure
  - Functioning of TCP
  - Flow Control
  - Congestion Control
  - Enhancements
  - Connection-oriented Communication via Sockets
  - Denial-of-Service Attacks via SYN Flood
- UDP
  - Basics
  - Connectionless Communication via Sockets
- Other Transport Layer Protocols
  - SCTP
  - DCCP
  - QUIC

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

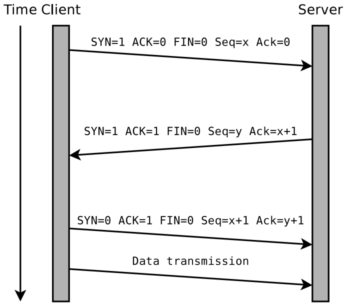
- The sequence number enables the receiver to...
  - correct the order of the segments
  - sort out segments, which arrived twice $\longrightarrow$ duplicate detection
- The length of a segment is known from the IP header
  - This way, missing bytes in the data stream are discovered and the receiver 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)

Server Functionality

- The server waits passively for an incoming connection
  - the server must first bind to and listen at a port before he can accept a connection

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



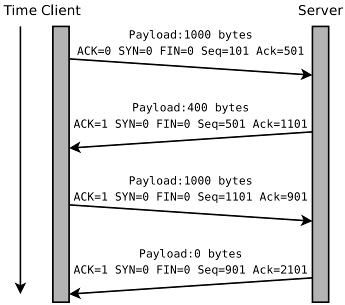
- The initial sequence numbers (x and y) are determined randomly
- No payload is exchanged during connection establishment!

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

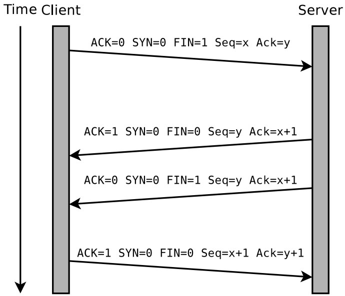
1. The client transmits 1000 bytes payload
2. Server acknowledges with ACK=1 the received payload and requests with the ACK number 1101 the next segment. In the same segment, the server transfers 400 bytes of payload
3. The client transmits another 1000 bytes payload. And it acknowledges the received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

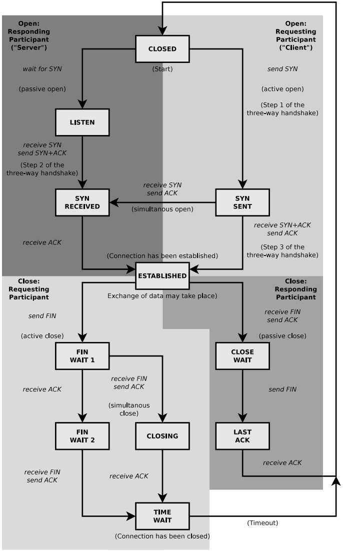
- Connection termination is similar to the connection establishment
- Instead of the `SYN` bit set, the `FIN` bit is used to close the connection, i.e., indicate that the sender will not transmit any more payload

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!

TCP – Simplified Finite State Machine



- CLOSED: Default state. Still no connection
- LISTEN: Waiting for a SYN message
- SYN-SENT: SYN is sent. Waiting for SYN and ACK
- SYN-RECEIVED: Replied with SYN and ACK to SYN. Waiting for ACK
- ESTABLISHED: The TCP connection is established and payload can be exchanged
- CLOSE-WAIT: FIN is received. Local application needs to reply with ACK
- LAST-ACK: ACK has already been sent. Now FIN is sent
- FIN-WAIT-1: FIN is sent. Waiting for ACK
- FIN-WAIT-2: ACK is sent. Waiting for FIN
- CLOSING: FIN is received and ACK is sent back
- TIME-WAIT: Connection is terminated

Source: http://www.tcpipguide.com/free/t_TCPOperationalOverviewandtheTCPFiniteStateMachineF-2.htm

Agenda

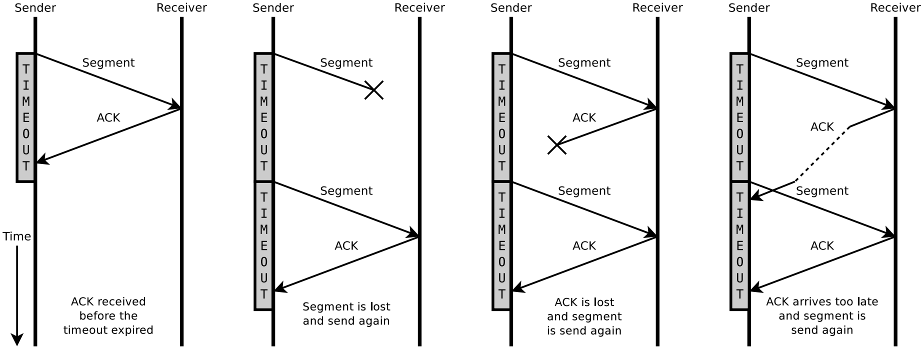
- TCP
  - Basics and Structure
  - Functioning of TCP
  - Flow Control
  - Congestion Control
  - Enhancements
  - Connection-oriented Communication via Sockets
  - Denial-of-Service Attacks via SYN Flood
- UDP
  - Basics
  - Connectionless Communication via Sockets
- Other Transport Layer Protocols
  - SCTP
  - DCCP
  - QUIC

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 $\Longrightarrow$ segment is sent again



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

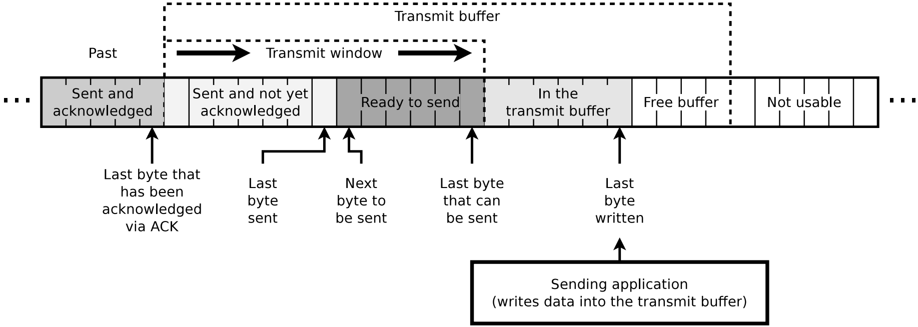
The Trivial File Transfer Protocol (RFC 783) operates according to the Stop-and-Wait principle

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

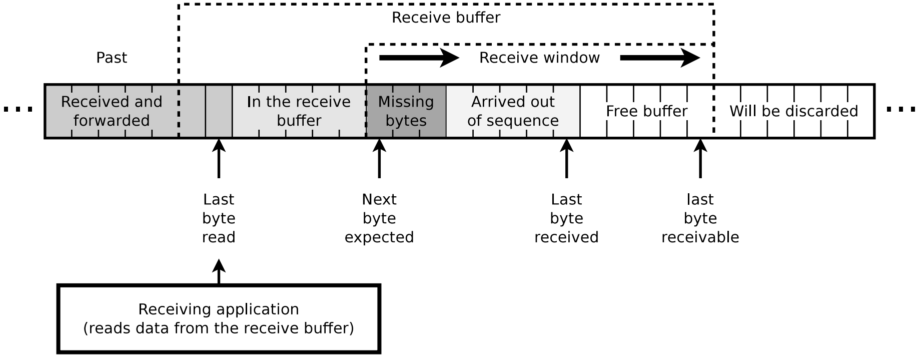
Sliding Window – Method: Sender

- 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

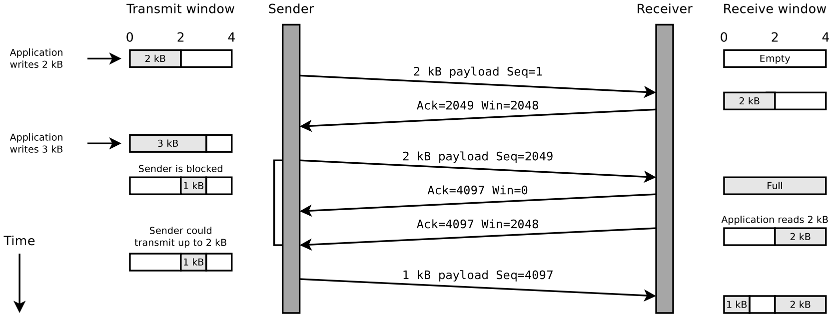
- The receive buffer contains data for the Application Layer, which...
  - is in the correct order, but has not been read
  - has been received out of sequence



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

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 sent, 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

Agenda

- TCP
  - Basics and Structure
  - Functioning of TCP
  - Flow Control
  - Congestion Control
  - Enhancements
  - Connection-oriented Communication via Sockets
  - Denial-of-Service Attacks via SYN Flood
- UDP
  - Basics
  - Connectionless Communication via Sockets
- Other Transport Layer Protocols
  - SCTP
  - DCCP
  - QUIC

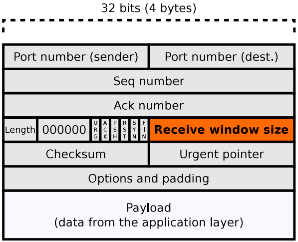
Reasons why Congestion occurs

- 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 than 12 kB are sent, it transmits only 12 kB

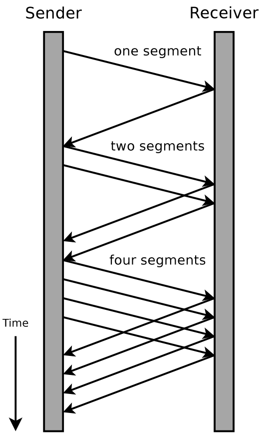
Determine the Size of the Congestion Window

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

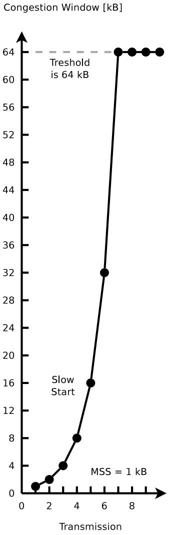
- Challenge for the sender: What is the size of the congestion window?
  - The sender never knows for sure the capacity of the network
  - The capacity of computers networks is not static
    - It depends among others of the network utilization and of the occurrence of network faults
- Solution: The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

Determine the Congestion Window Size – Connection Establishment



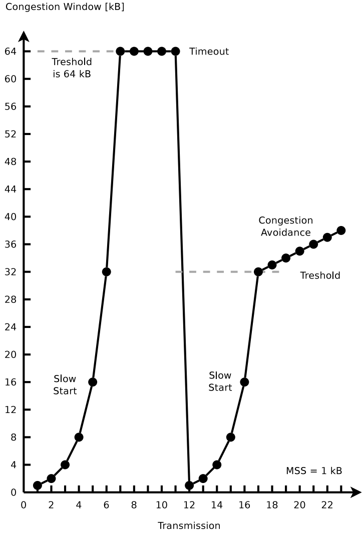
- During connection establishment, the sender initializes the congestion window to maximum segment size (MSS)
- Method:
  - 1 segment of size MSS is sent
    - If the segment is acknowledged before the timeout expires, the congestion window is doubled
  - 2 segments of size MSS are sent
    - If both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

Determine the Congestion Window Size – Slow Start



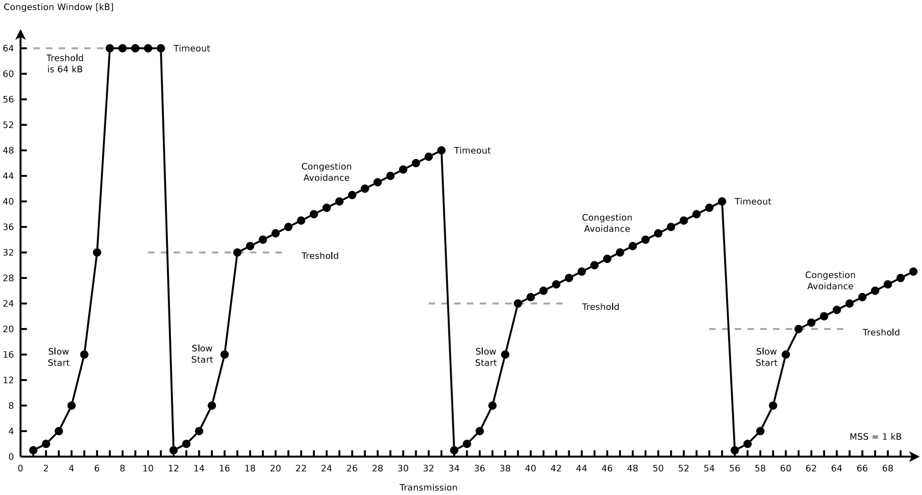
- The congestion window grows exponentially until...
  - the size of the receive window is reached, which has been determined by the sender
  - or the threshold is reached
  - or a timeout expires
- The exponential growth phase is called slow start
  - Reason: The low transmission rate of the sender 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
  - At least in the original congestion control algorithm called *Tahoe* (1988)
- Modern TCP implementations use different congestion control algorithms which differ between...
  - expired timeout caused by congestion of the network
  - and multiple arrival of acknowledgments (ACKs) caused by loss event

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

Agenda

- TCP
  - Basics and Structure
  - Functioning of TCP
  - Flow Control
  - Congestion Control
  - Enhancements
  - Connection-oriented Communication via Sockets
  - Denial-of-Service Attacks via SYN Flood
- UDP
  - Basics
  - Connectionless Communication via Sockets
- Other Transport Layer Protocols
  - SCTP
  - DCCP
  - QUIC

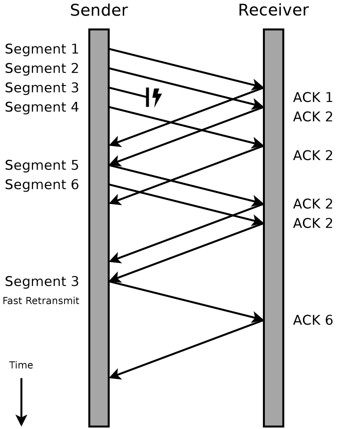
TCP Enhancements

Robustness Principle

„TCP implementations will follow a general principle of robustness: be conservative in what you do, be liberal in what you accept from others.“
Jon Postel, RFC 793, page 13

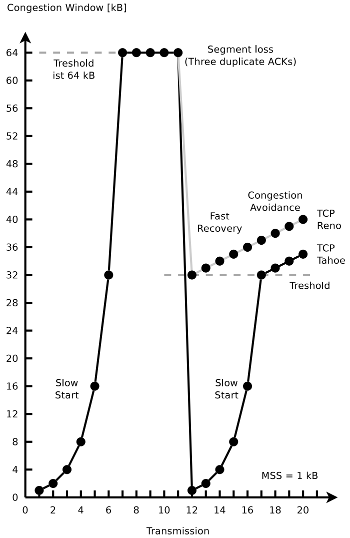
- TCP has no version number
- Continuous enhancements and extensions were necessary over time, in order to ...
  - become more efficient
  - adapt to changing transmission media (e.g., wireless communication)
  - leverage the improving performance of the terminal devices
- The main challenge is to stay compatible

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 $\Longrightarrow$ fast retransmit

Fast Recovery



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

Selective Acknowledgement (SACK)

- TCP only acknowledges continuous segments
  $\longrightarrow$ single segments that get lost inside the sliding window cause the retransmission of the whole window
- Solution: selectively acknowledge discontinuous segment ranges inside a window
- The sender has now the chance to retransmit the unacknowledged segments
- In case of timeout, the sender falls back to resend all segments since the last cumulative ACK
- In case of a cumulative, the sender aborts retransmitting

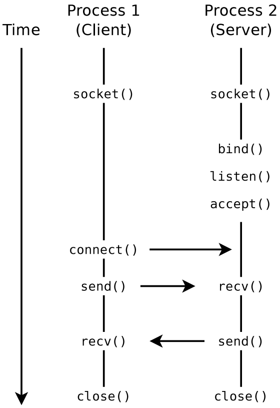
- Specified in RFC 2018
- Is negotiated during connection establishment
- Are part of the TCP header options
- The sender maintains a separate *SACK* table

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)

Agenda

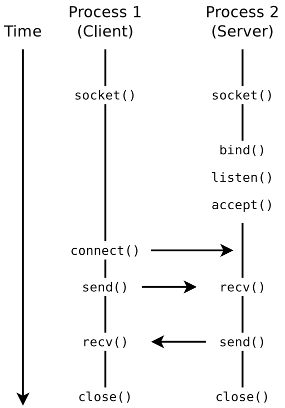
Connection-oriented Communication via Sockets – TCP



- 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

sd.close()                    # Close socket
```



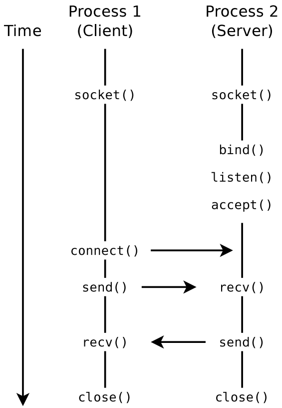

```
$ python tcp_server.py
```

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('Received:', repr(data))
```

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

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



Agenda

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 becomes unresponsive
- Countermeasure: Real-time analysis of the network by intelligent firewalls

Agenda

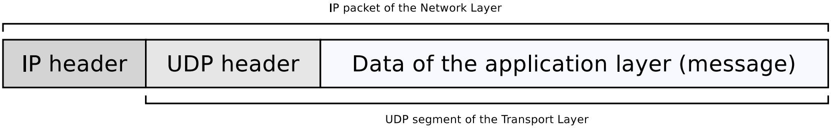
Agenda

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 $\Rightarrow$ more lightweight
  - Only responsible for addressing of the datagrams
  - No guarantees $\longrightarrow$ best effort
  - Datagrams can get lost, duplicated, or arrive out of order
- Depending on the application (e.g., video streaming) this is accepted
- UDP causes lesser delay compared to TCP
- Allows for multicast and broadcast

User Datagram Protocol (UDP)

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

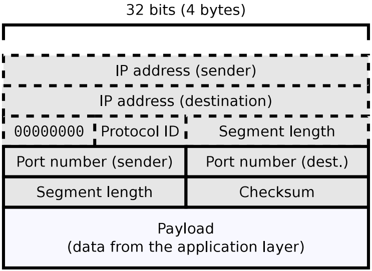


UDP standard: RFC 768 from 1980
`http://tools.ietf.org/rfc/rfc768.txt`

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 datagram (without pseudo-header)
  - Checksum of the complete datagram (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

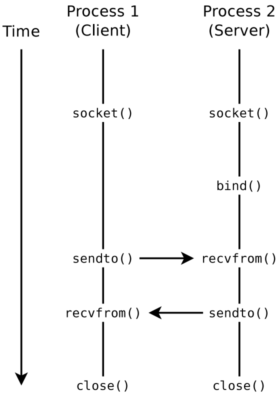


Remember NAT from slide set 8...

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

Agenda

Connectionless Communication via Sockets – UDP

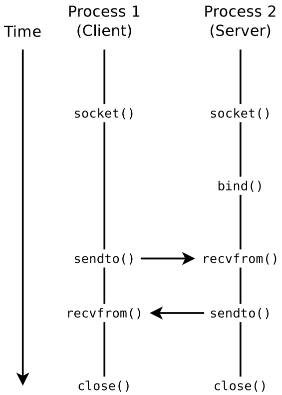


- Client
  - Create socket (`socket`)
  - Send (`sendto`) and receive data (`recvfrom`)
  - Close socket (`close`)
- Server
  - Create socket (`socket`)
  - Bind socket to a port (`bind`)
  - Send (`sendto`) and receive data (`recvfrom`)
  - Close socket (`close`)

Sockets via UDP – Example (Server)

```python
#!/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
```

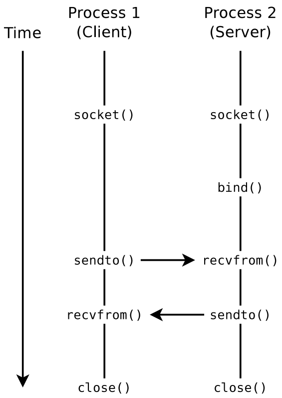


Sockets via UDP – Example (Client)

```python
#!/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))
```



Agenda

- TCP
  - Basics and Structure
  - Functioning of TCP
  - Flow Control
  - Congestion Control
  - Enhancements
  - Connection-oriented Communication via Sockets
  - Denial-of-Service Attacks via SYN Flood
- UDP
  - Basics
  - Connectionless Communication via Sockets
- Other Transport Layer Protocols
  - SCTP
  - DCCP
  - QUIC

Agenda

Streaming Control Transmission Protocol (SCTP)

- Connection-oriented transport layer protocol
- Specified in RFCs 4960, 6096, 6335, and 8260
- Message-oriented
  - Supports messages of arbitrary size (using fragmentation)
  - Smaller messages can be consolidated into one SCTP packet
- Allows for multiple streams per connection
- Stream properties are configurable separately
  - Reliability
  - Order
  - Flow- and congestion control
  - Priorities
- Supports mobility
- Implements SACK
- Design goal: Differentiation between application data
  e.g., real-time audio/video versus large data
- Used for real-time browser-to-browser communication → WebRTC

Agenda

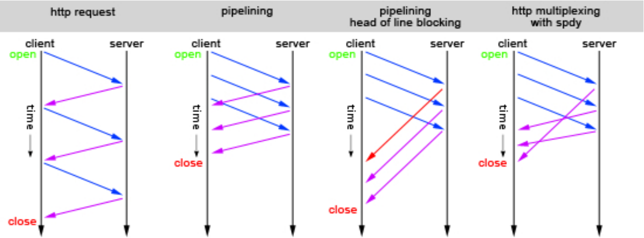
- TCP
  - Basics and Structure
  - Functioning of TCP
  - Flow Control
  - Congestion Control
  - Enhancements
  - Connection-oriented Communication via Sockets
  - Denial-of-Service Attacks via SYN Flood
- UDP
  - Basics
  - Connectionless Communication via Sockets
- Other Transport Layer Protocols
  - SCTP
  - DCCP
  - QUIC

Datagram Congestion Control Protocol (DCCP)

- Connection-oriented transport layer protocol
- Specified in RFC 4340
- Unreliable unicast transport with congestion control
- Reliable connection establishment
- Designed for real-time applications
- Detects packet lost without retransmissions

Agenda

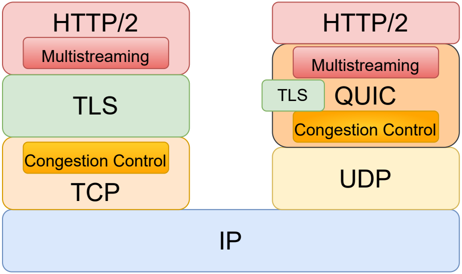
New Challenges: Modern WWW



Source: KSkun's blog, https://ksmeow.moe/quic/

- Problem: Modern web pages consists of multiple individual downloads $\longrightarrow$ sequential loading is slow
- TCP stops the download as soon as the stream is interrupted $\longrightarrow$ head of line blocking
- Solution: UDP plus connection management

Quick UDP Internet Connections (QUIC)



- Goal: reduced latency
- Specified in RFCs 8999 and 9000
- Fast connection management
  - Fast connection establishment
  - Full multiplexing of streams
- Improved packet loss and congestion handling
- Integrated TLS support $\rightarrow$ security
- Multipath support

You should now be able to answer the following questions:

- What are characteristics of a transport layer protocol?
- What is a socket?
- Why do we need multiple transport layer protocols on top of IP?
- What is the difference between TCP and UDP?
- How does flow control and congestion control work in TCP?