

Lesson 4: End-to-End Protocols

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Fall 2024

Online video link: <https://youtu.be/of5CmizjSW8>

Outline

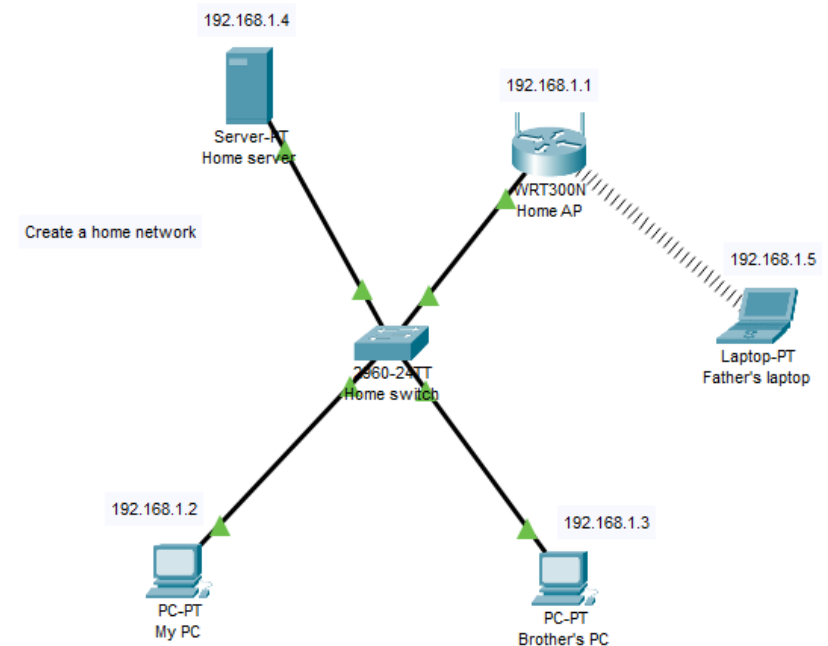
- Lab2: Create a network service/app in Packet Tracer
- Lab3: Create a simple Internet network
- Network protocols
 1. UDP
 2. TCP

Lab2: Service/application

Youtube link for Lab2 reference

<https://youtu.be/azCHIVfY1o>

- This lab is to recall knowledge about application and services
- Based on the first lab, please create a new DNS server with IP 192.168.1.6
- Assign the webserver at 192.168.1.4 with the domain **ccu.edu.tw**
- Test whether PCs can access Web on the Home server by using the domain name
- Activate FTP services on the Home server and create an account admin/admin → Test FTP connection from laptop

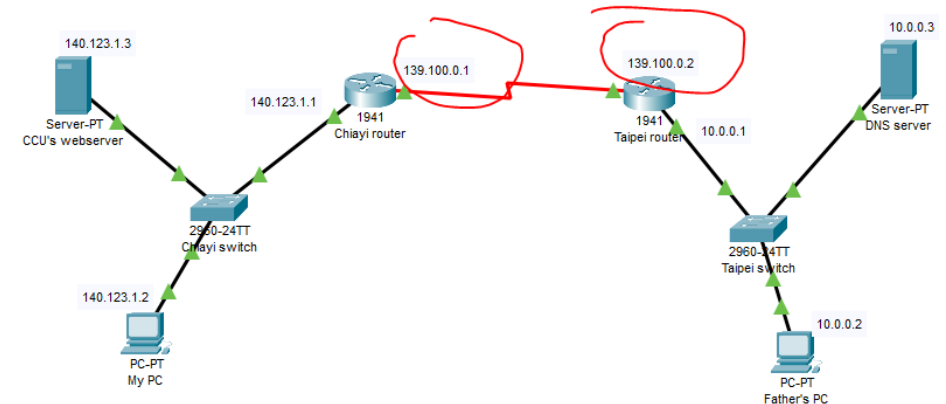


Lab3: Create an Internet network

Youtube link for Lab3 reference

<https://youtu.be/IYusDFMPZWU>

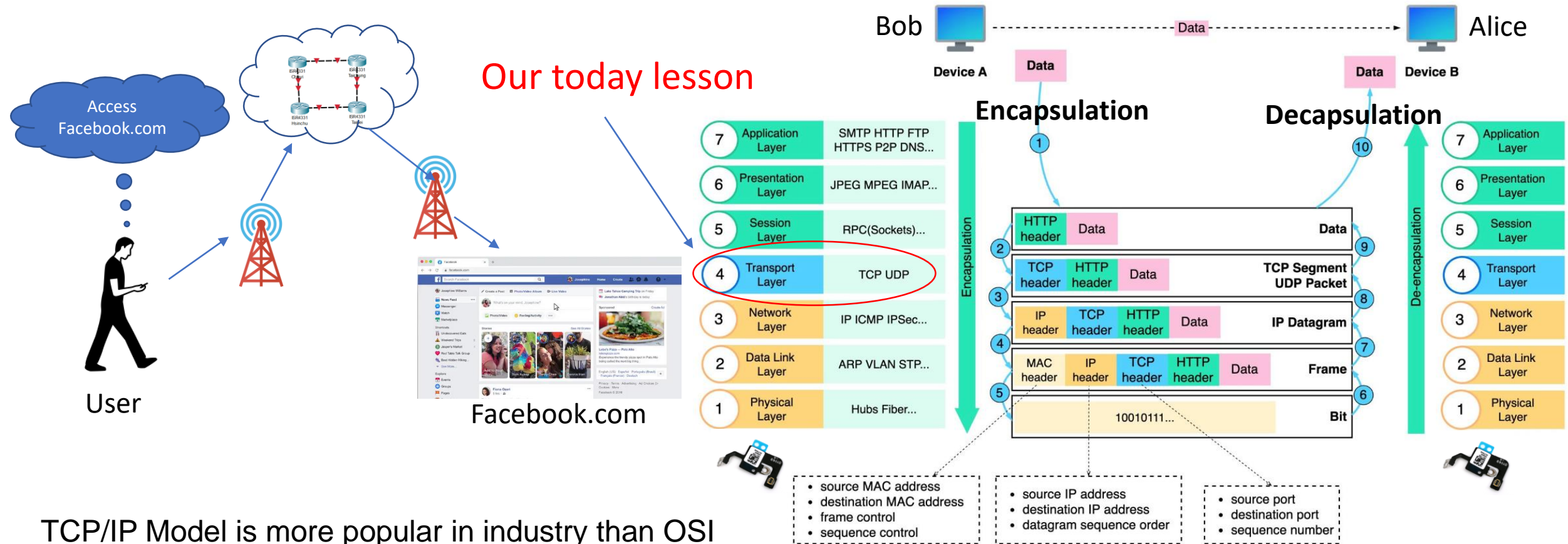
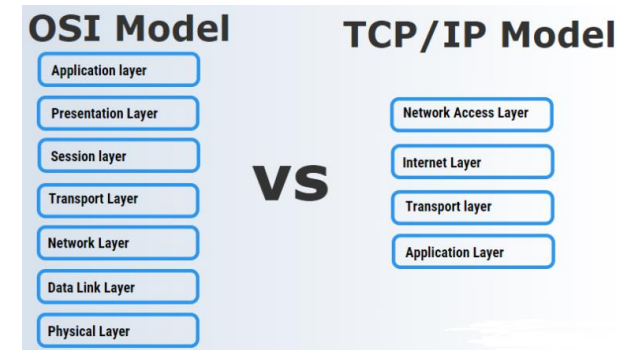
- Create a simple Internet between Chiayi and Taipei
- The Chiayi network starts with the network IP **140.123.1.0/24**
- The Taipei network starts with the network IP **10.0.0.0/16**
- Test whether PCs in the Chiayi network can access PCs at the Taipei network
- Create a Webserver at the Chiayi network (CCU's website) and a DNS server at the Taipei network
- Assign the webserver at 140.123.1.3 with the domain **ccu.edu.tw**
- Create a DNS record for the website at DNS server
- Test whether Father's PC can access CCU's website via domain name



Network IP between Chiayi router and Taipei router must be **different** from the connected networks (**140.123.1.0**, **10.0.0.0**)

In the sample lab, we use IP **139.100.0.0** for that network and the IPs for serial port can start with **139.100.0.1** and **139.100.0.2** as illustrated in the above figure

Encapsulation/Decapsulation

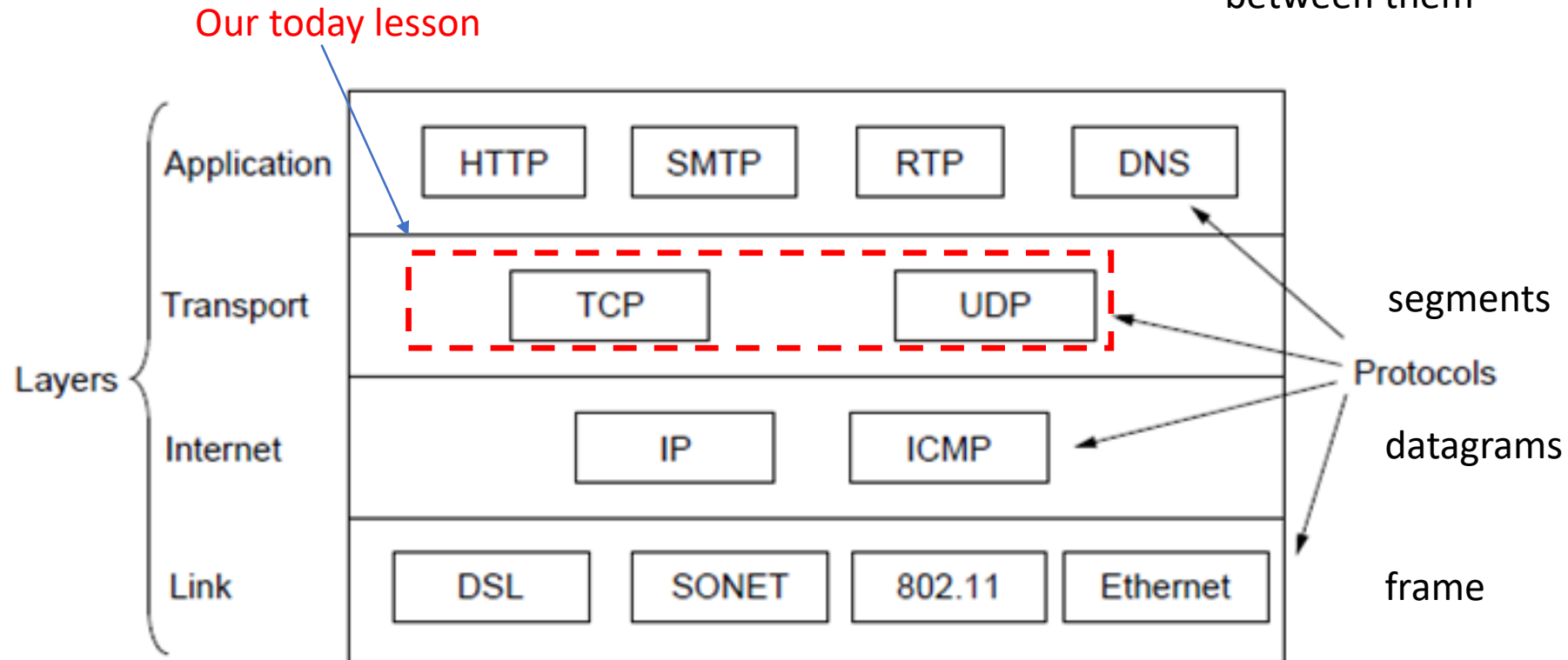


TCP/IP Model is more popular in industry than OSI
OSI is for academic research

<https://blog.bytebytego.com/>

End-to-end Protocols

The transport layer has responsibility for establishing a **temporary communication session** between two applications and delivering **data** between them



<https://www.dcs.bbk.ac.uk/~ptw/teaching/IWT/transport-layer/notes.html>

Message encapsulation through layers

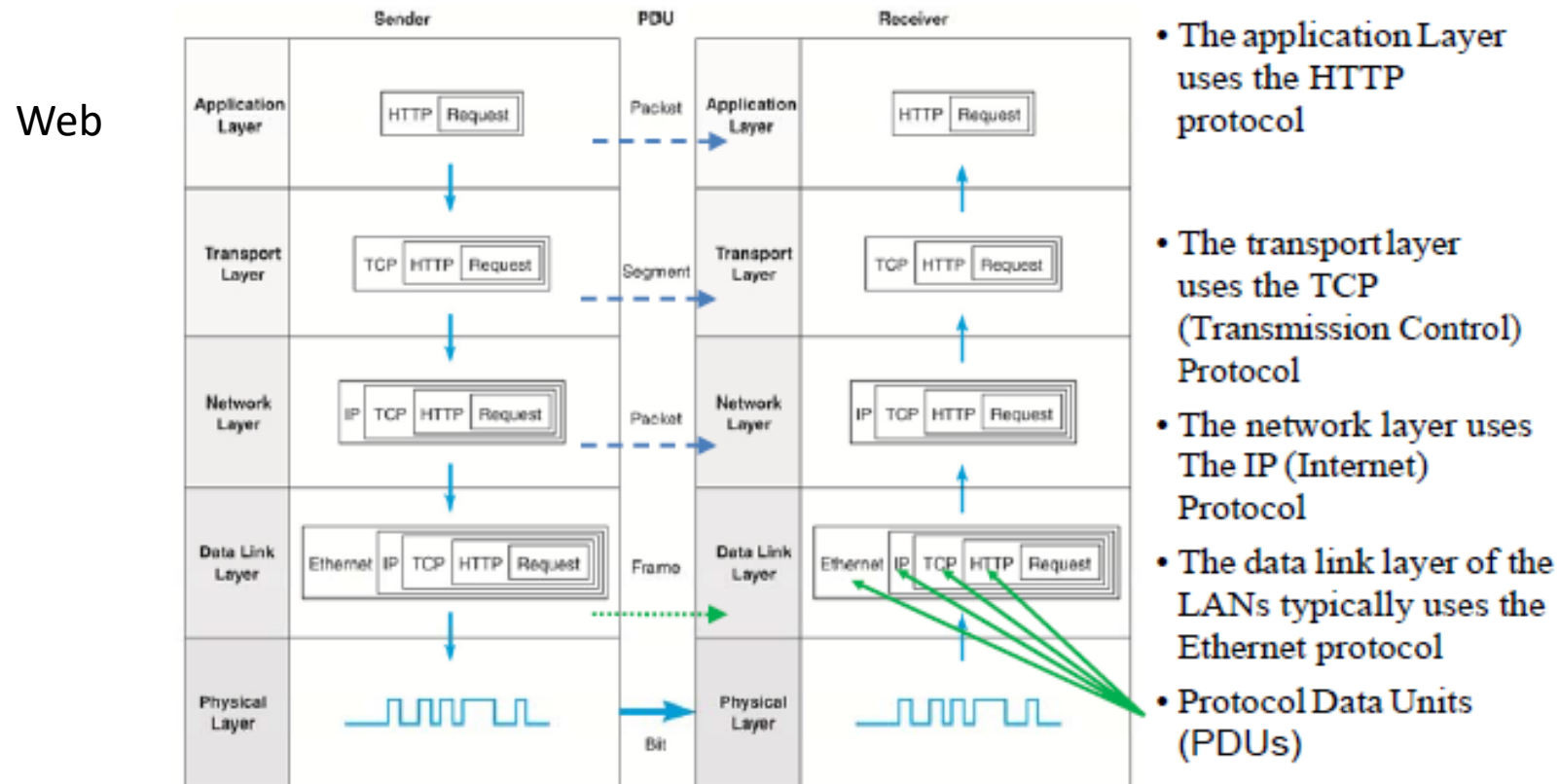


Fig. 1: Message transmission through layers.

End-to-end Protocols

- Below the presentation and session layer is transport layer
- A transport protocol can be expected to provide
 - **Guarantees message delivery**
 - Delivers messages in **the same order** they were sent
 - Delivers at most one copy of each message
 - Supports arbitrarily **large messages**
 - Supports **synchronization** between the sender and the receiver
 - Allows the receiver to apply **flow control** to the sender
 - Supports multiple application processes on each host

End-to-end Protocols

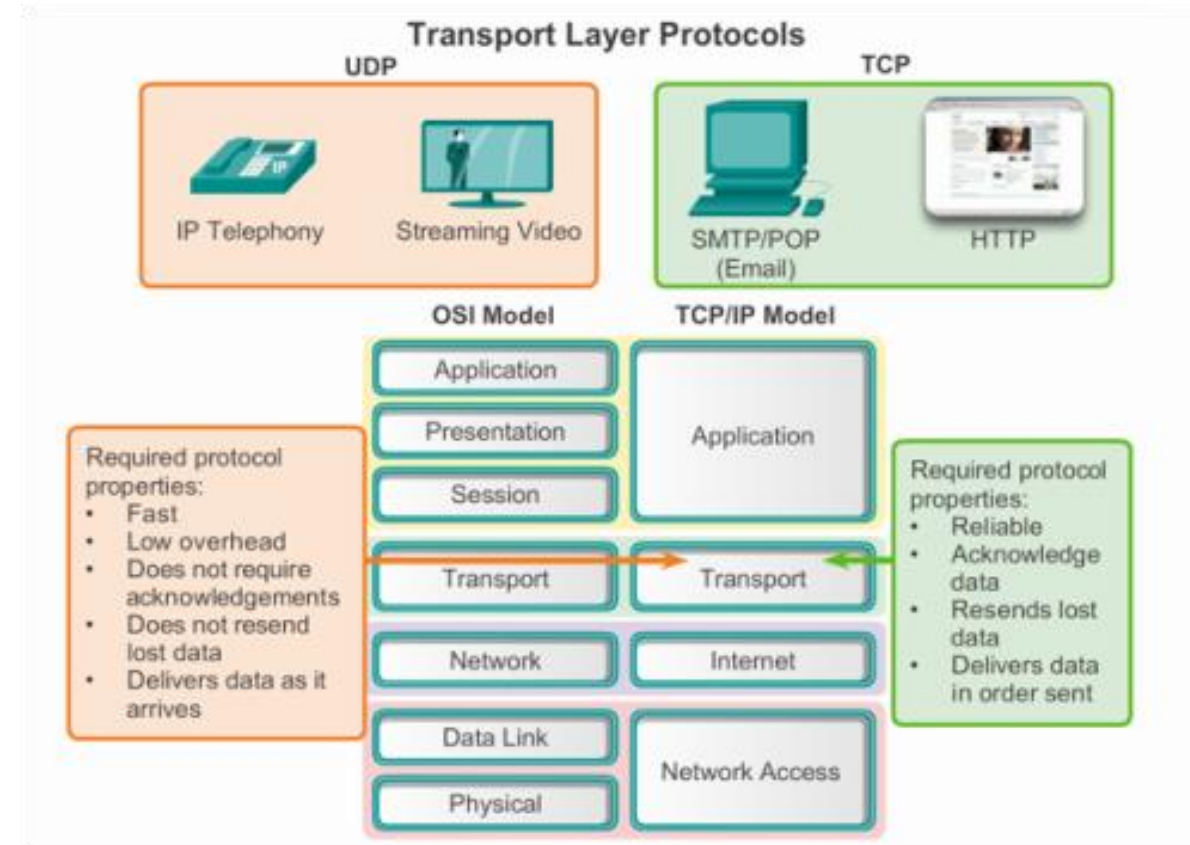
- Typical **limitations** of the network on which transport protocol will operate
 - Drop messages
 - Reorder messages
 - Deliver **duplicate copies** of a given message
 - Limit messages to some **finite size**
 - Deliver messages after an arbitrarily long delay

End-to-end Protocols

- Challenge for Transport Protocols
 - Develop algorithms that turn the less-than-desirable properties of the underlying network into the high level of service required by application programs
- Connectionless transport layer
 1. Treat **each packet** as **an individual** and delivery to the destination
 2. The receiver **doesn't send acknowledgement** of the packet
- Connection oriented transport layer
 1. Establish connection between the sender and the receiver before transmitting data
 2. The receiver sends acknowledgement for each received packet

TCP vs UDP

- Two protocols are used
 1. Connectionless: User Datagram Protocol (UDP)
 2. Connection oriented: Transmission Control Protocol (TCP)



User Datagram Protocol (UDP)

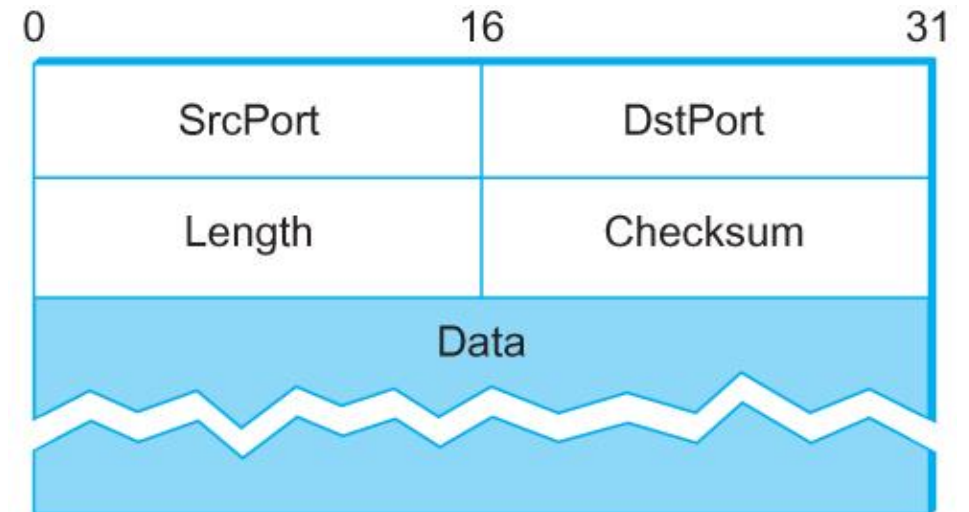
- UDP is suitable for those applications where **data loss doesn't affect the perceived quality of the service**
- Multiplayer Online Games, VoIP, and Video Live Streaming
- Fast but non-guaranteed transfer. It is also called “best effort” transfer

UDP Datagram (header)

- Each UDP segment must include UDP header fields identifying the socket connection
- These header fields are the *source port number field* and the *destination port number field*
- Each port number is a 16-bit number: 0 to 65535
- Port numbers below 1024 are called *well-known ports* and are reserved for standard services

Port number	Application protocol	Description	Transport protocol
21	FTP	File transfer	TCP
23	Telnet	Remote login	TCP
25	SMTP	E-mail	TCP
53	DNS	Domain Name System	UDP
79	Finger	Lookup information about a user	TCP
80	HTTP	World wide web	TCP
110	POP-3	Remote e-mail access	TCP
119	NNTP	Usenet news	TCP
161	SNMP	Simple Network Management Protocol	UDP

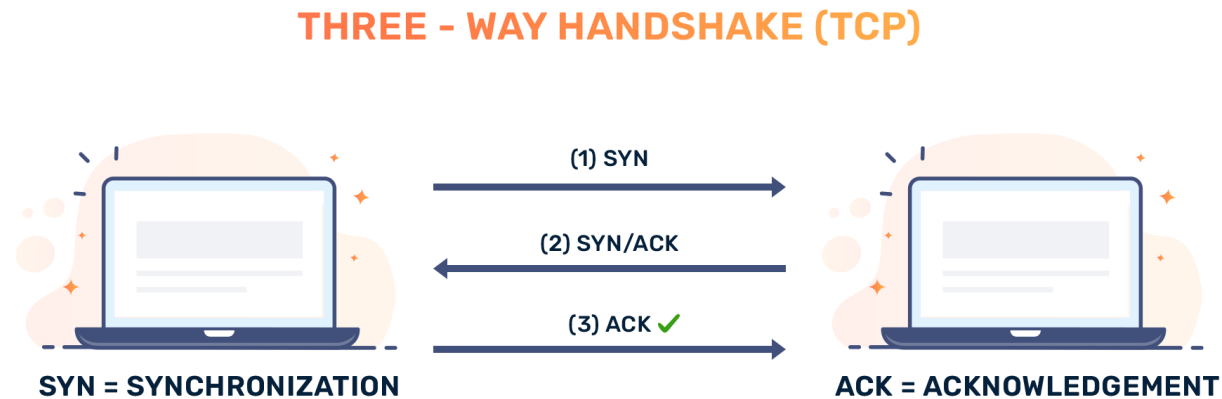
Famous ports for some services



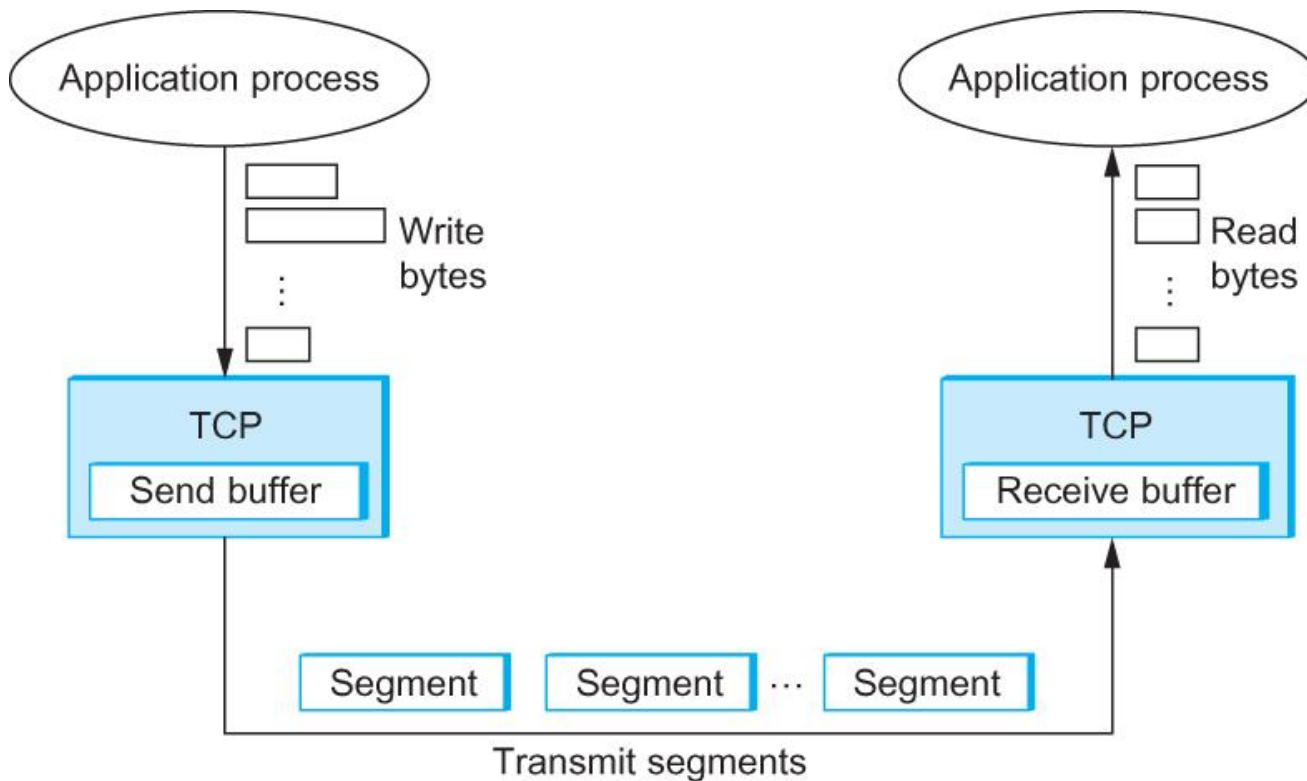
Format for UDP header

Transmission Control Protocol (TCP)

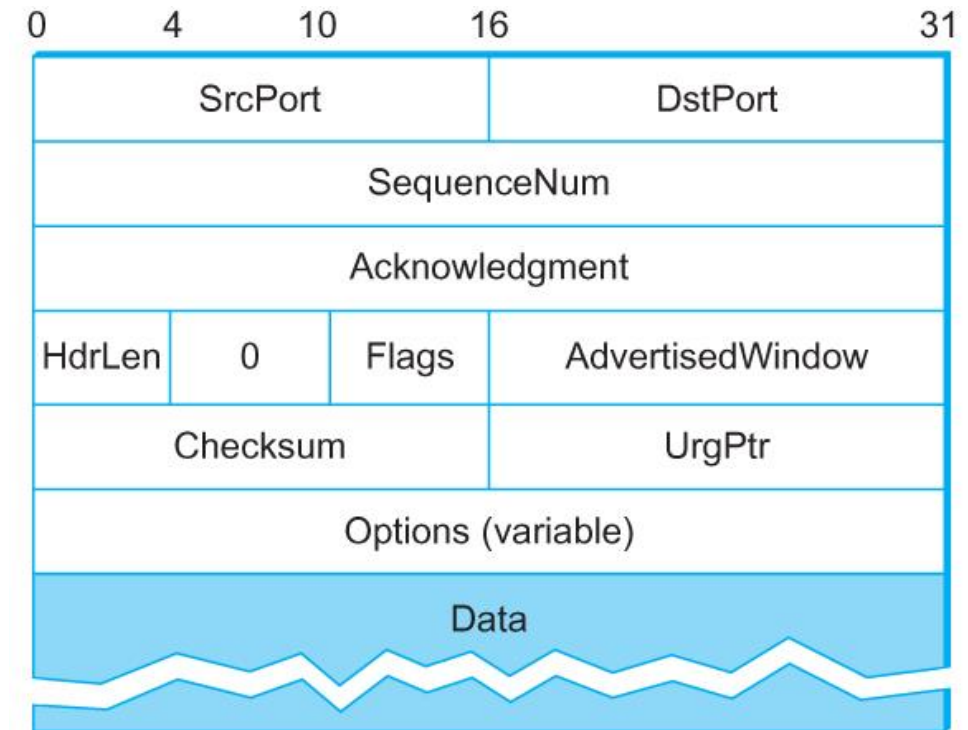
- In contrast to UDP, Transmission Control Protocol (TCP) offers the following services
 - Reliable
 - Connection oriented
 - Byte-stream service
- TCP is also known as a **three-way handshake** protocol



TCP Segment & TCP Header



How TCP manages a byte stream.



TCP Header Format

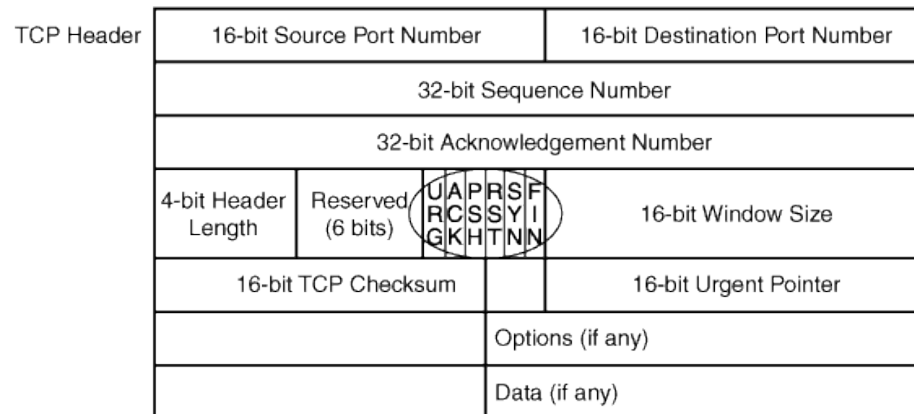
TCP Header

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

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119	NNTP	Usenet news	TCP
161	SNMP	Simple Network Management Protocol	UDP

TCP Header

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



The SYN and FIN flags are set.

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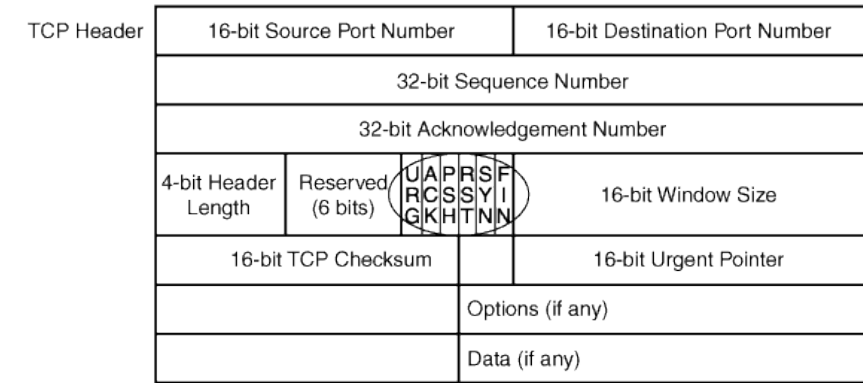
TCP Header

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

```
▼ Flow 1
  SrcAddr: 192.168.88.102
  DstAddr: 188.188.188.188
  Protocol: TCP (6)
  SrcPort: 27127 (27127)
  DstPort: 23 (23)
  ▼ TCP Flags: 0x12, ACK, SYN
    00.. .... = Reserved: 0x0
    ..0. .... = URG: Not used
    ...1 .... = ACK: Used
    .... 0... = PSH: Not used
    .... .0.. = RST: Not used
    .... ..1. = SYN: Used
    .... ...0 = FIN: Not used
  Source Mac Address: 0c:ad:95:fb:00:00
  Destination Mac Address: 0c:27:76:de:00:00
  InputInt: 1
  Classification Engine ID: PANA-L7 (13)
  Selector ID: 000001
  OutputInt: 0
  Direction: Ingress (0)
  Octets: 84
  Packets: 2
```

TCP Header

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

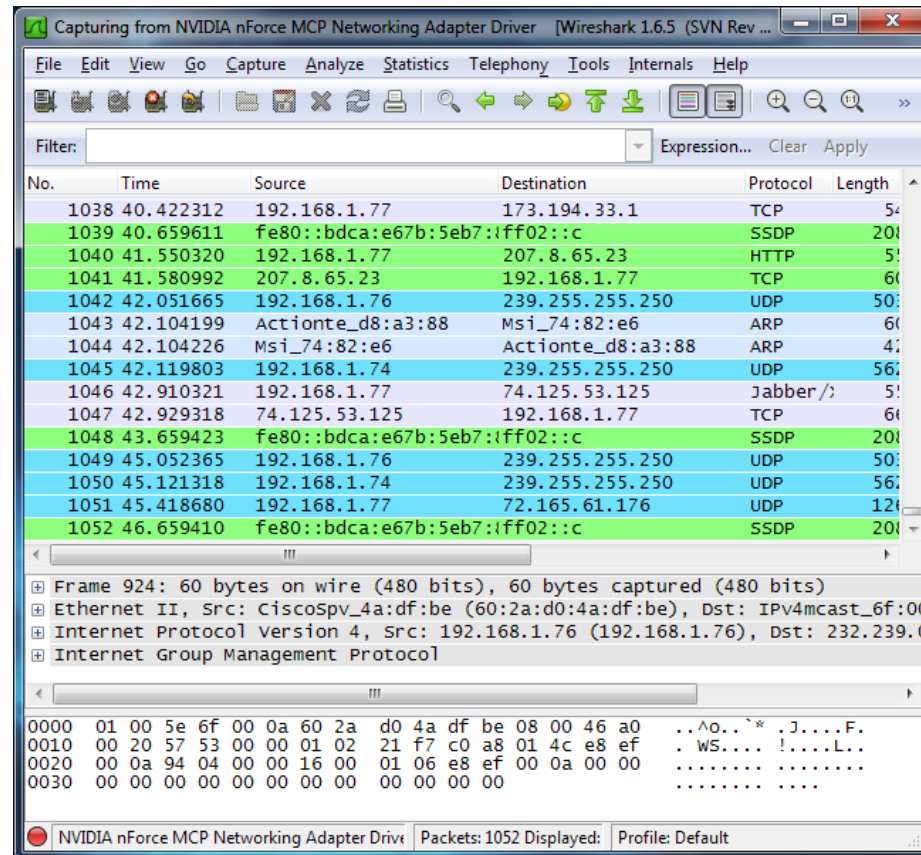


The SYN and FIN flags are set.

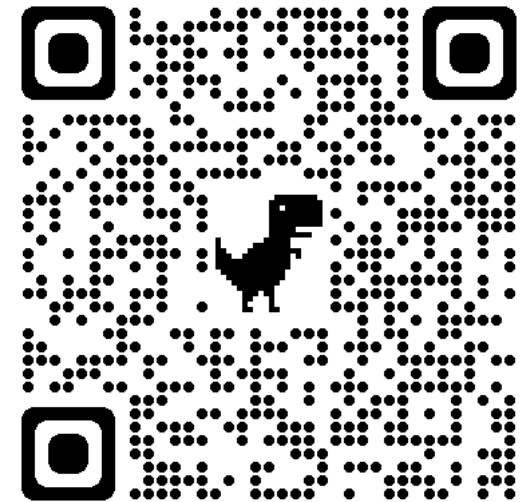
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Packet Analysis in Wireshark

- A free and open-source **packet analyzer** for network troubleshooting, analysis, software and communications protocol development, and education.



<https://www.wireshark.org/>



Wireshark

- Allow us to check TCP/UDP packet in details
- Statistical data
 - ✓ Packet flow (src → dest)
 - ✓ Endpoints
 - ✓ Packet lengths
 - ✓ I/O graphs

Time	Source	Destination	Protocol	Length	Info
125.580331	192.168.3.153	146.66.71.198	TCP	66	33572 → 80 [SYN] Seq=0 Win=64240 Len=0 MSS=1460 WS=256 SACK
154.645496	146.66.71.198	192.168.3.153	TCP	66	80 → 33572 [SYN, ACK] Seq=0 Ack=1 Win=29200 Len=0 MSS=1460
155.645569	192.168.3.153	146.66.71.198	TCP	54	33572 → 80 [ACK] Seq=1 Ack=1 Win=65536 Len=0
386.6563605	192.168.3.153	146.66.71.198	HTTP	635	GET / HTTP/1.1
418.6626732	146.66.71.198	192.168.3.153	TCP	54	80 → 33572 [ACK] Seq=1 Ack=582 Win=30464 Len=0
429.7036925	146.66.71.198	192.168.3.153	TCP	1514	80 → 33572 [ACK] Seq=1 Ack=582 Win=30464 Len=1460 [TCP segm
430.7036935	146.66.71.198	192.168.3.153	TCP	1514	80 → 33572 [ACK] Seq=1461 Ack=582 Win=30464 Len=1460 [TCP s
431.7037267	192.168.3.153	146.66.71.198	TCP	54	33572 → 80 [ACK] Seq=582 Ack=2921 Win=65536 Len=0
432.7037726	146.66.71.198	192.168.3.153	TCP	1514	80 → 33572 [ACK] Seq=2921 Ack=582 Win=30464 Len=1460 [TCP s
433.7037734	146.66.71.198	192.168.3.153	TCP	1514	80 → 33572 [ACK] Seq=4381 Ack=582 Win=30464 Len=1460 [TCP s
434.7037736	146.66.71.198	192.168.3.153	TCP	1514	80 → 33572 [ACK] Seq=5841 Ack=582 Win=30464 Len=1460 [TCP s
435.7037739	146.66.71.198	192.168.3.153	TCP	1514	80 → 33572 [ACK] Seq=7301 Ack=582 Win=30464 Len=1460 [TCP s
436.7037741	146.66.71.198	192.168.3.153	TCP	1514	80 → 33572 [ACK] Seq=8761 Ack=582 Win=30464 Len=1460 [TCP s
437.7037744	146.66.71.198	192.168.3.153	TCP	1514	80 → 33572 [ACK] Seq=10221 Ack=582 Win=30464 Len=1460 [TCP
438.7037747	146.66.71.198	192.168.3.153	TCP	1514	80 → 33572 [ACK] Seq=11681 Ack=582 Win=30464 Len=1460 [TCP
439.7037750	146.66.71.198	192.168.3.153	TCP	1514	80 → 33572 [ACK] Seq=13141 Ack=582 Win=30464 Len=1460 [TCP
440.7038214	192.168.3.153	146.66.71.198	TCP	54	33572 → 80 [ACK] Seq=582 Ack=2921 Win=65536 Len=0
450.7098733	146.66.71.198	192.168.3.153	TCP	1514	80 → 33572 [ACK] Seq=2921 Ack=582 Win=30464 Len=1460 [TCP

Packet flows

<

> Frame 431: 54 bytes on wire (432 bits), 54 bytes captured (432 bits) on interface 0

> Ethernet II, Src: IntelCor_42:70:89 (48:f1:7f:42:70:89), Dst: Rosewill_12:2b:0f (68:1c:a2:12:2b:0f)

> Internet Protocol Version 4, Src: 192.168.3.153, Dst: 146.66.71.198

▼ Transmission Control Protocol, Src Port: 33572, Dst Port: 80, Seq: 582, Ack: 2921, Len: 0

Source Port: 33572

Destination Port: 80

[Stream index: 12]

[TCP Segment Len: 0]

Sequence number: 582 (relative sequence number)

[Next sequence number: 582 (relative sequence number)]

Acknowledgment number: 2921 (relative ack number)

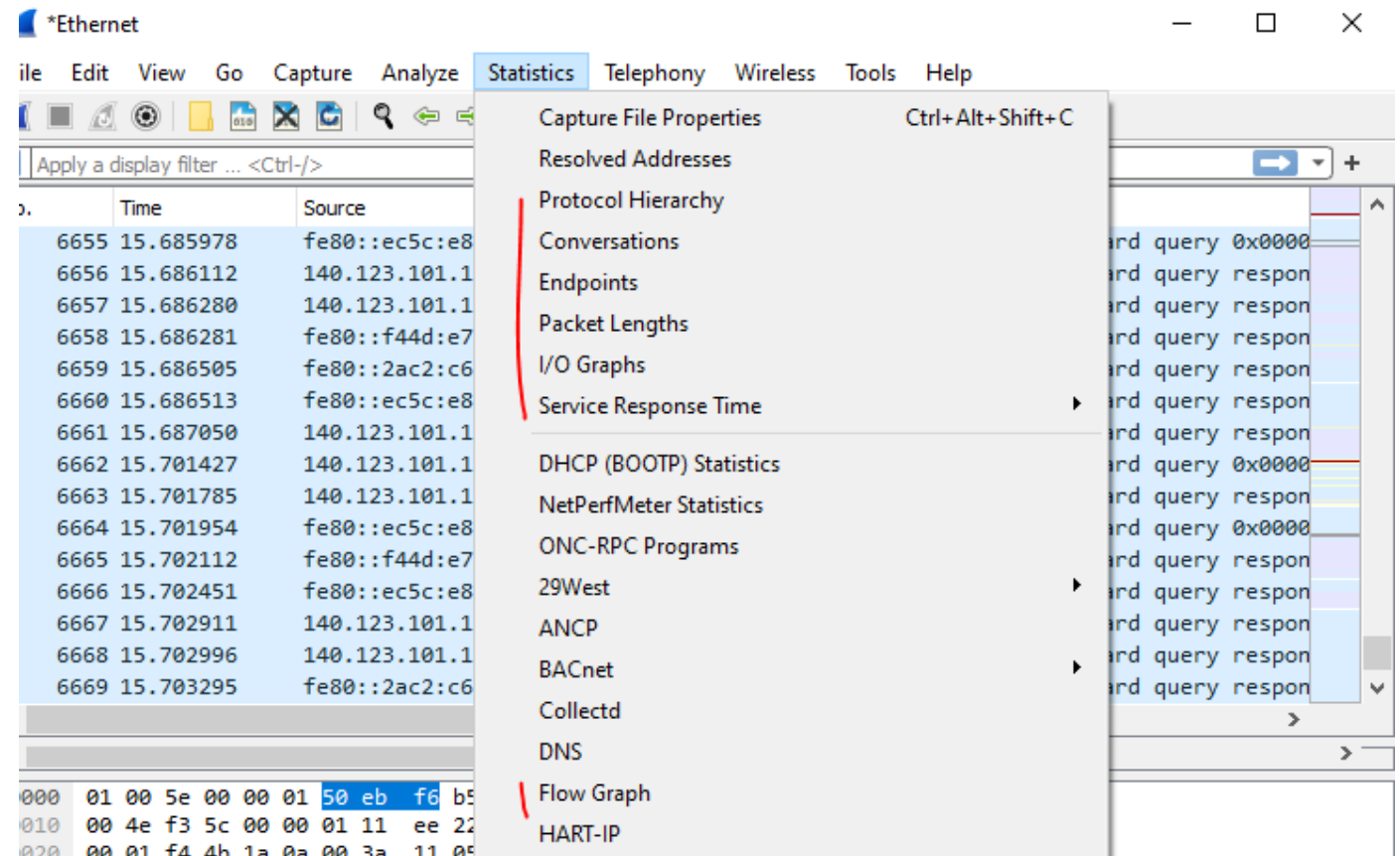
Packet details

Wireshark

- Allow us to check TCP/UDP packet in details

- **Statistic feature**

- ✓ Packet flow (src → dest)
- ✓ Endpoints
- ✓ Packet lengths
- ✓ I/O graphs



Wireshark

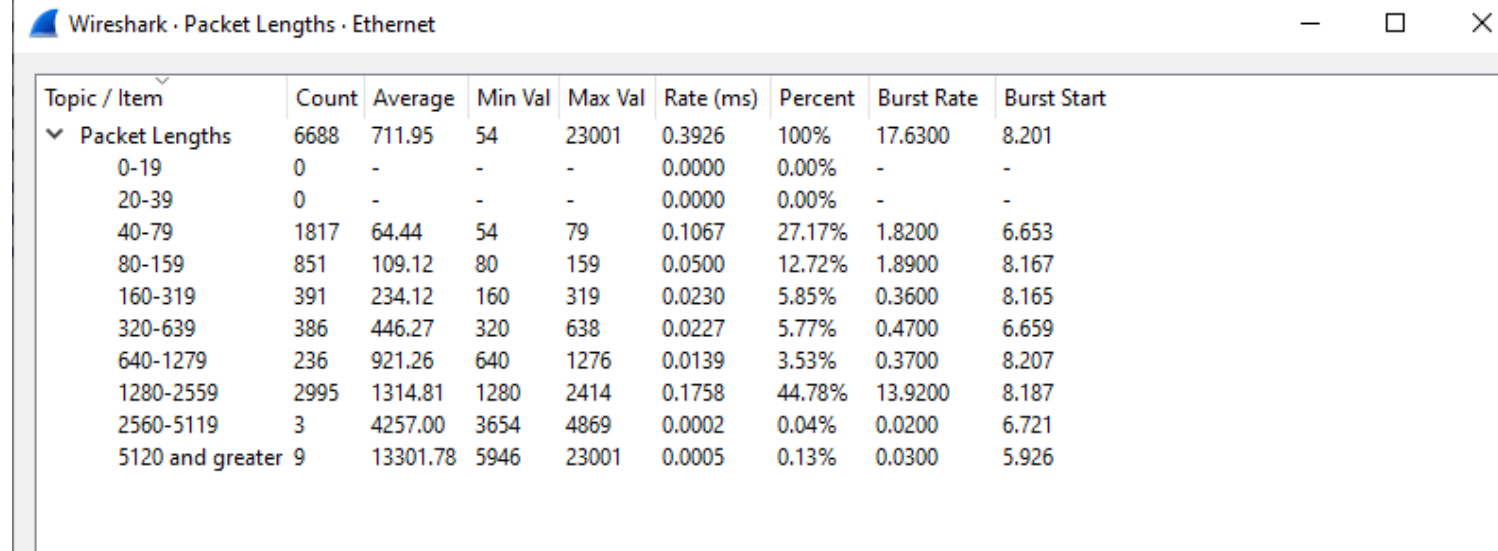
- Allow us to check TCP/UDP packet in details
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Wireshark · Endpoints · Ethernet

Ethernet · 43	IPv4 · 92	IPv6 · 13	TCP · 167	UDP · 203		
Address	Packets	Bytes	Tx Packets	Tx Bytes	Rx Packets	Rx Bytes
20.78.118.165	2	114	1	60	1	54
20.89.149.168	5	328	3	220	2	108
20.108.229.21	2	115	1	60	1	55
20.110.103.72	1	60	1	60	0	0
20.126.223.223	1	60	1	60	0	0
20.189.173.1	1	60	1	60	0	0
20.189.173.23	1	60	1	60	0	0
20.190.166.66	1	60	1	60	0	0
20.210.223.40	1	60	1	60	0	0
34.36.124.104	24	8670	12	6205	12	2465
34.64.34.68	6	402	3	216	3	186
34.200.122.61	27	13 k	15	10 k	12	3521
35.168.49.207	31	16 k	19	8857	12	7774
44.218.58.190	51	20 k	27	14 k	24	5603
51.104.15.253	2	120	2	120	0	0
52.72.226.68	60	20 k	28	11 k	32	9189
52.86.181.185	2	114	1	60	1	54
52.92.243.160	48	24 k	28	22 k	20	2374
52.111.234.0	3	260	2	160	1	100
52.112.104.122	3	180	3	180	0	0

Wireshark

- Allow us to check TCP/UDP packet in details
- Statistic feature
 - ✓ Packet flow (src→ dest)
 - ✓ Endpoints
 - ✓ **Packet lengths**
 - ✓ I/O graphs

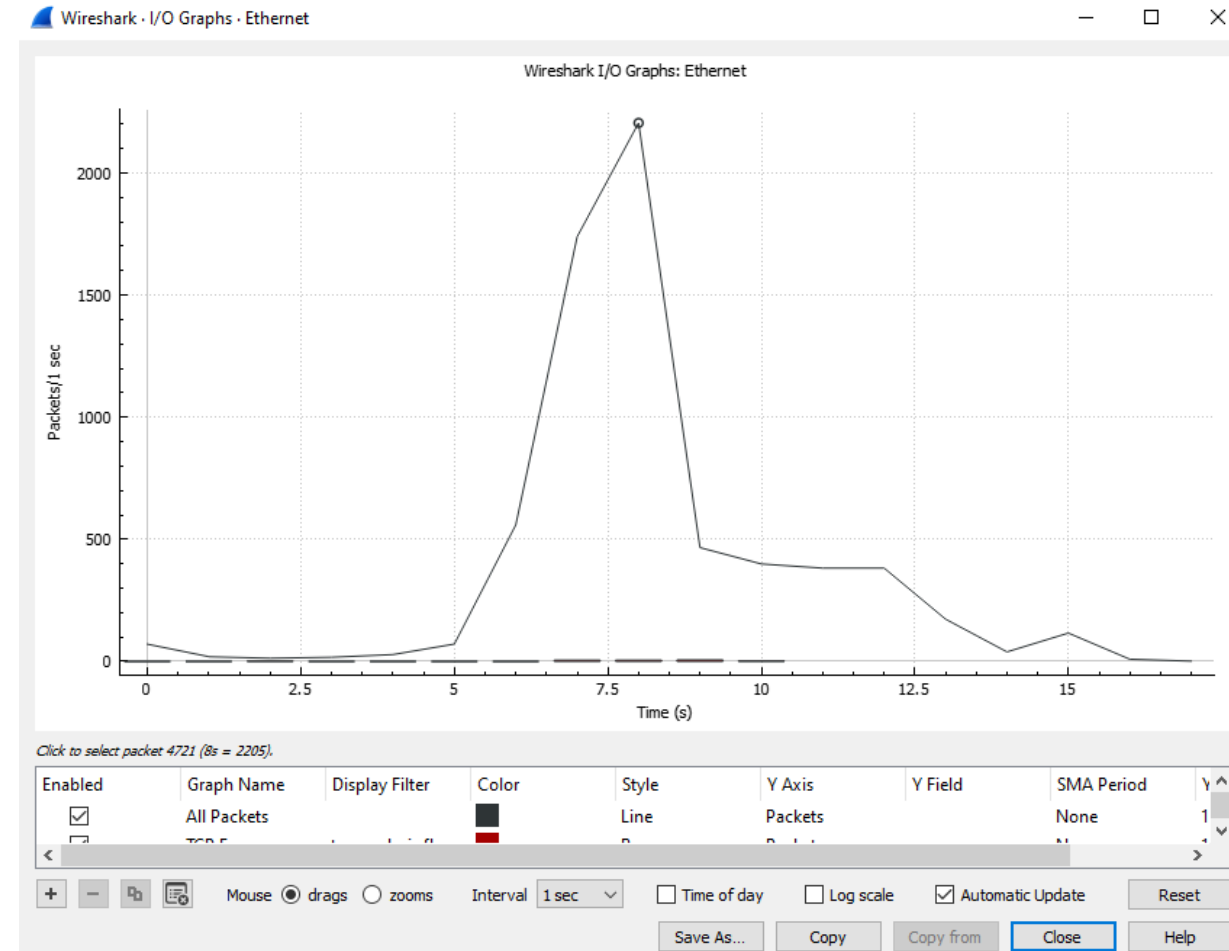
A screenshot of the Wireshark 'Packet Lengths' statistics window for the Ethernet interface. The window title is 'Wireshark · Packet Lengths · Ethernet'. It contains a table with 9 columns: 'Topic / Item', 'Count', 'Average', 'Min Val', 'Max Val', 'Rate (ms)', 'Percent', 'Burst Rate', and 'Burst Start'. The 'Packet Lengths' category is expanded, showing a list of packet size ranges and their corresponding statistics. The data is as follows:

Topic / Item	Count	Average	Min Val	Max Val	Rate (ms)	Percent	Burst Rate	Burst Start
Packet Lengths	6688	711.95	54	23001	0.3926	100%	17.6300	8.201
0-19	0	-	-	-	0.0000	0.00%	-	-
20-39	0	-	-	-	0.0000	0.00%	-	-
40-79	1817	64.44	54	79	0.1067	27.17%	1.8200	6.653
80-159	851	109.12	80	159	0.0500	12.72%	1.8900	8.167
160-319	391	234.12	160	319	0.0230	5.85%	0.3600	8.165
320-639	386	446.27	320	638	0.0227	5.77%	0.4700	6.659
640-1279	236	921.26	640	1276	0.0139	3.53%	0.3700	8.207
1280-2559	2995	1314.81	1280	2414	0.1758	44.78%	13.9200	8.187
2560-5119	3	4257.00	3654	4869	0.0002	0.04%	0.0200	6.721
5120 and greater	9	13301.78	5946	23001	0.0005	0.13%	0.0300	5.926

Wireshark

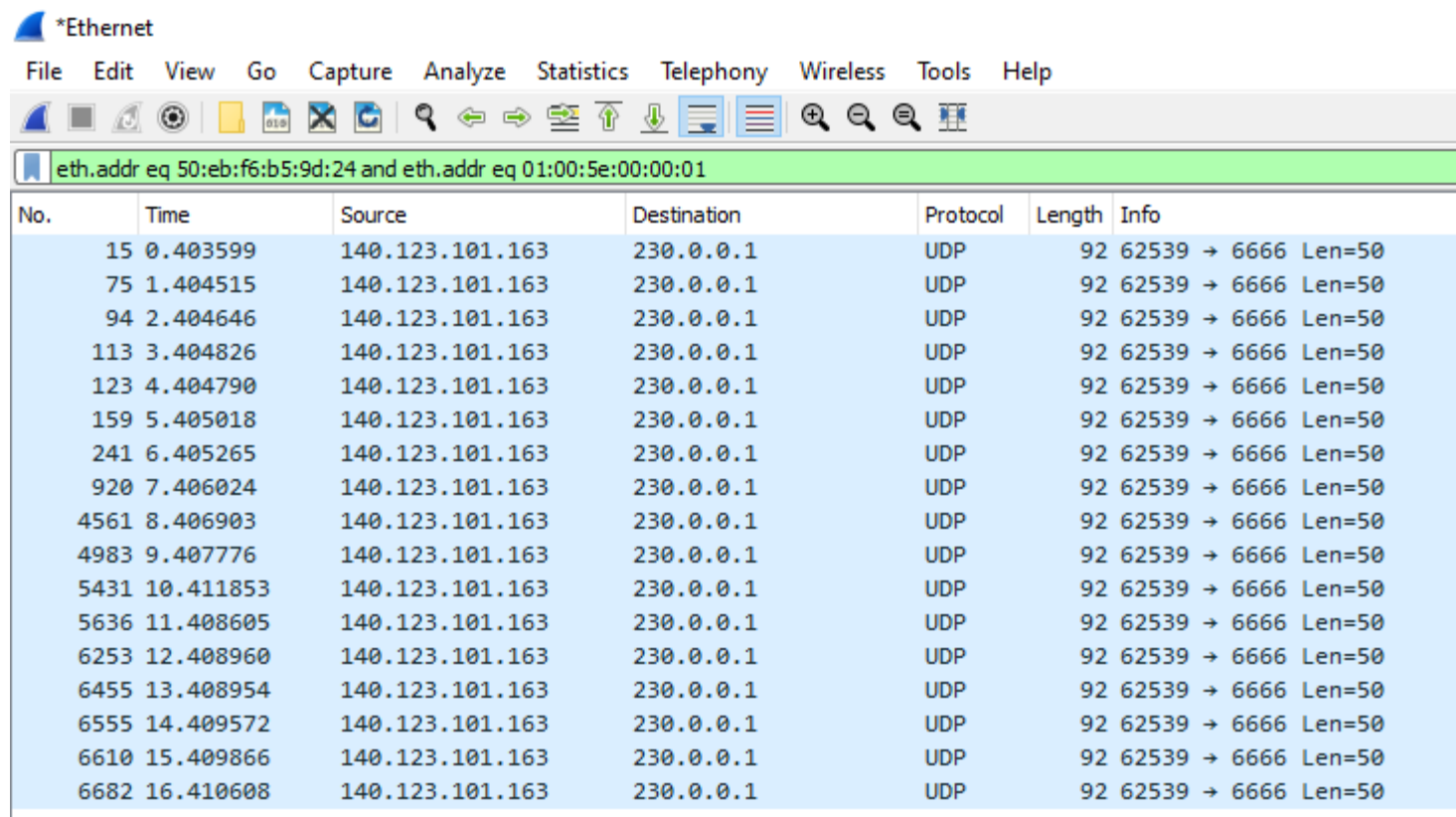
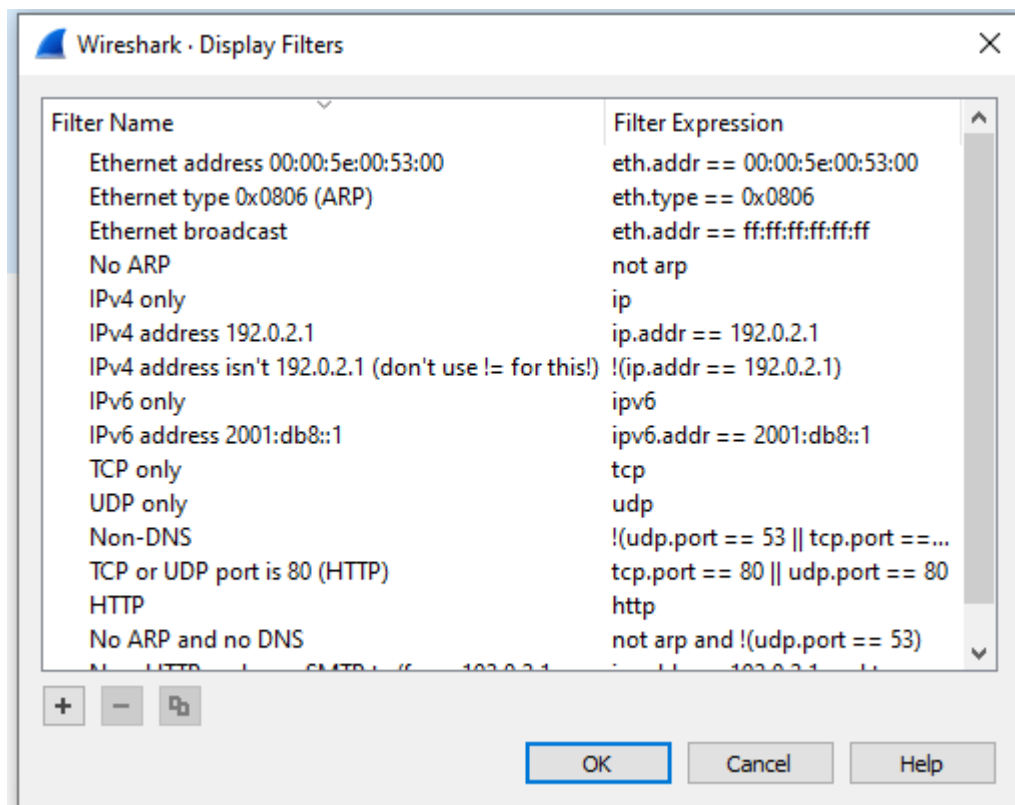
- Allow us to check TCP/UDP packet in details
- Statistic data
 - ✓ Packet flow (src→ dest)
 - ✓ Endpoints
 - ✓ Packet lengths
 - ✓ I/O graphs

I/O graphs



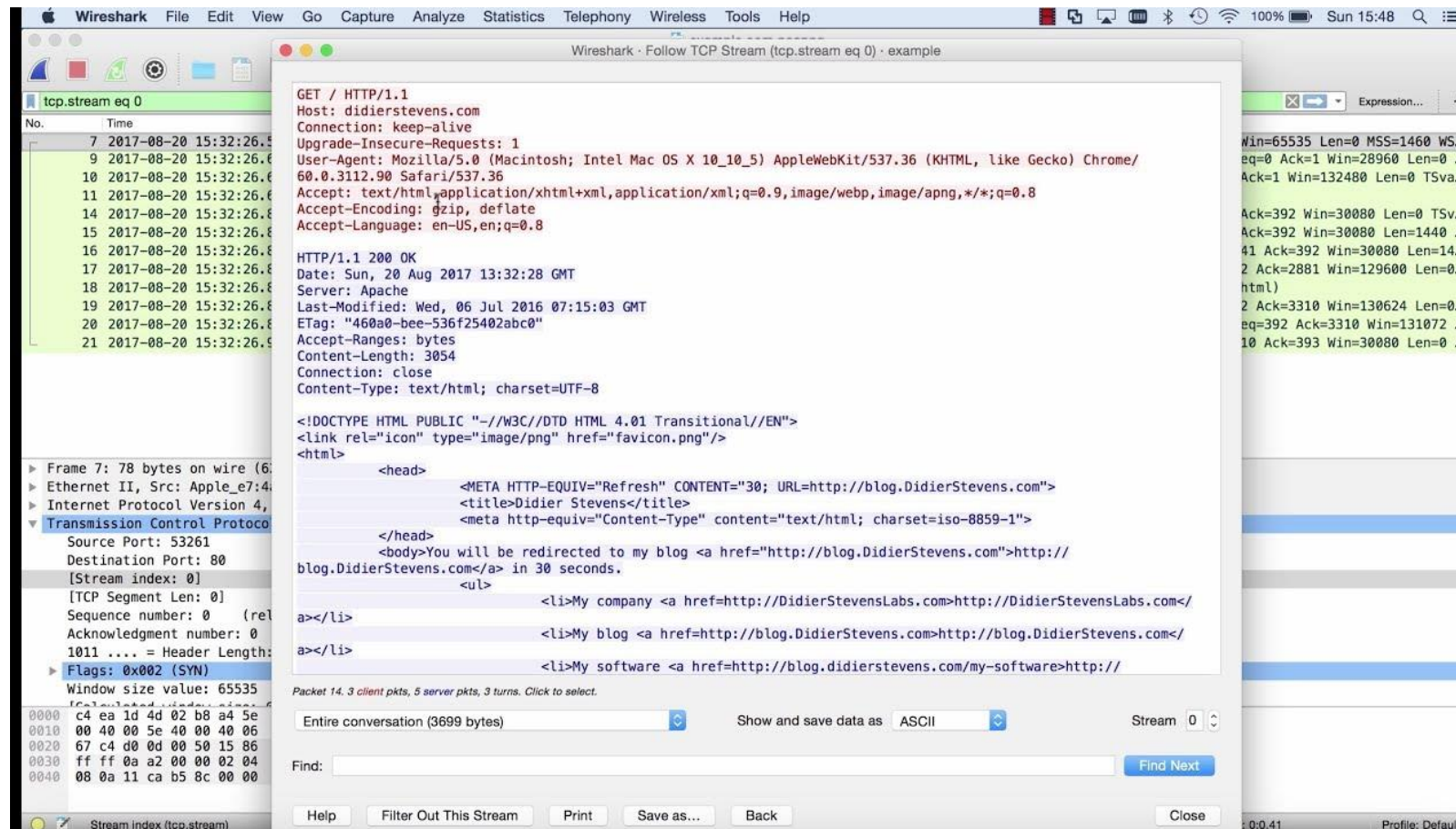
Packet flow filter in Wireshark

- Find all packets of a flow



TCP stream follow in Wireshark

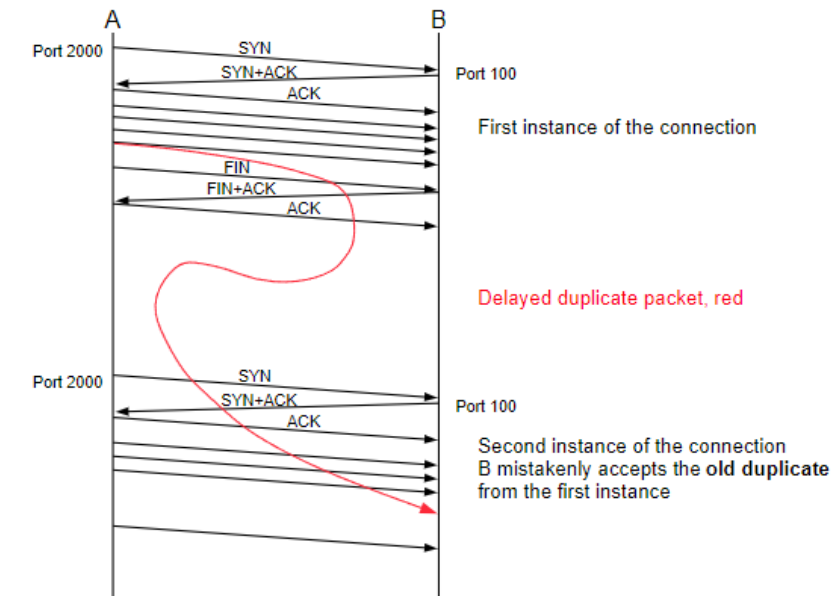
- Read all packets of http requests



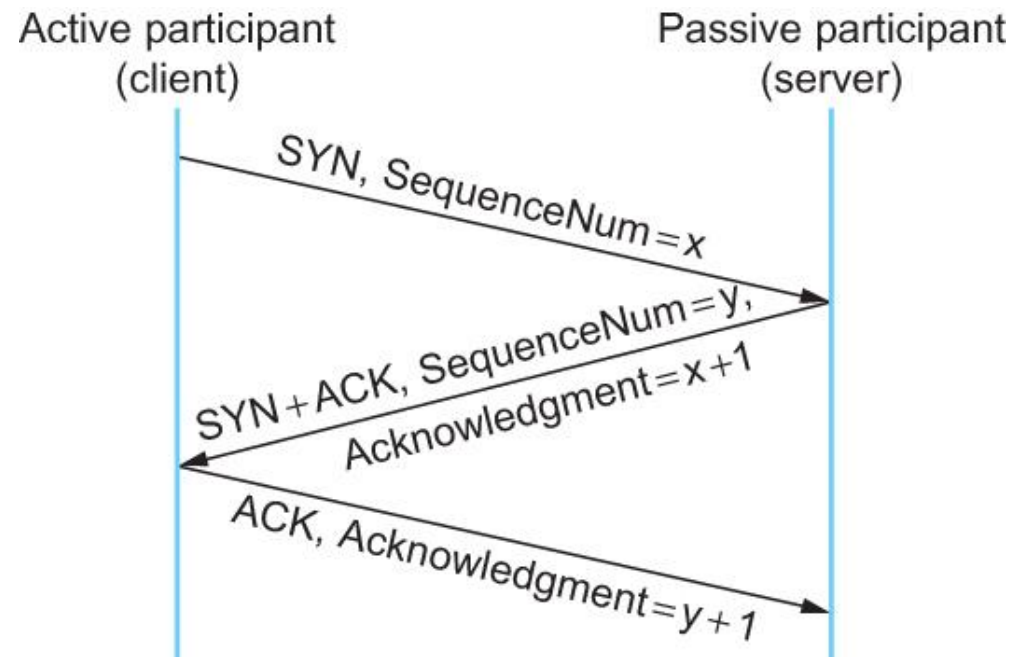
Network issues

TCP Issues

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



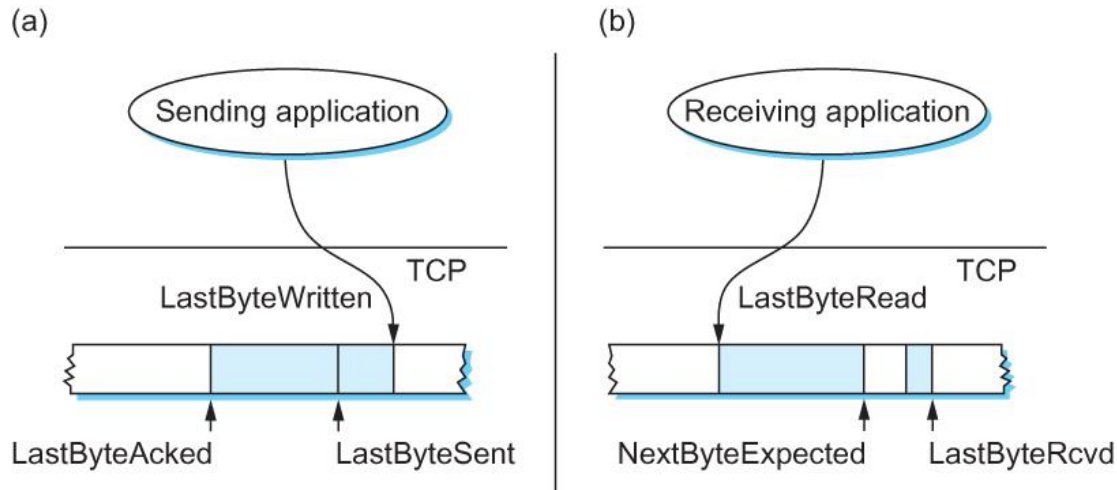
Connection Establishment/Termination in TCP



Timeline for three-way handshake algorithm

Sliding Window Revisited

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



- Sending Side
 - $\text{LastByteAcked} \leq \text{LastByteSent}$
 - $\text{LastByteSent} \leq \text{LastByteWritten}$
- Receiving Side
 - $\text{LastByteRead} < \text{NextByteExpected}$
 - $\text{NextByteExpected} \leq \text{LastByteRcvd} + 1$

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

End-to-end Issues

- TCP is a **byte-oriented protocol**, which means that the sender writes bytes into a TCP connection and the receiver reads bytes out of the TCP connection.
- Although “byte stream” describes the service TCP offers to application processes, TCP **does not, itself, transmit individual bytes over the Internet**.
- TCP needs **a mechanism using which each side of a connection will learn what resources the other side** is able to apply to the connection → TCP flow/congestion control
- TCP needs a mechanism using which the sending side will learn the capacity of the network

Flow control VS Congestion control

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

- **Flow Control:** Algorithms to prevent that the sender overruns the receiver with information
- **Error Control:** Algorithms to recover or conceal the effects from packet losses
- **Congestion Control:** Algorithms to prevent that the sender overloads the network

→ The goal of each of the control mechanisms are different.

→ In TCP, the implementation of these algorithms is combined

bandwidth limited

Opposite objectives

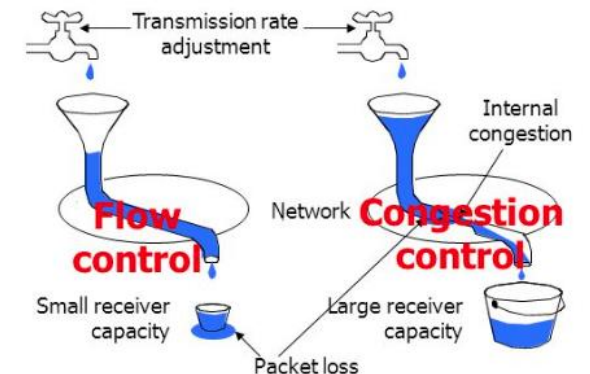
- End-system
 - Optimize its own throughput
 - Possibly at the expense of other end-systems

Opposite objectives

- Network
 - Optimize overall throughput

Two different problems

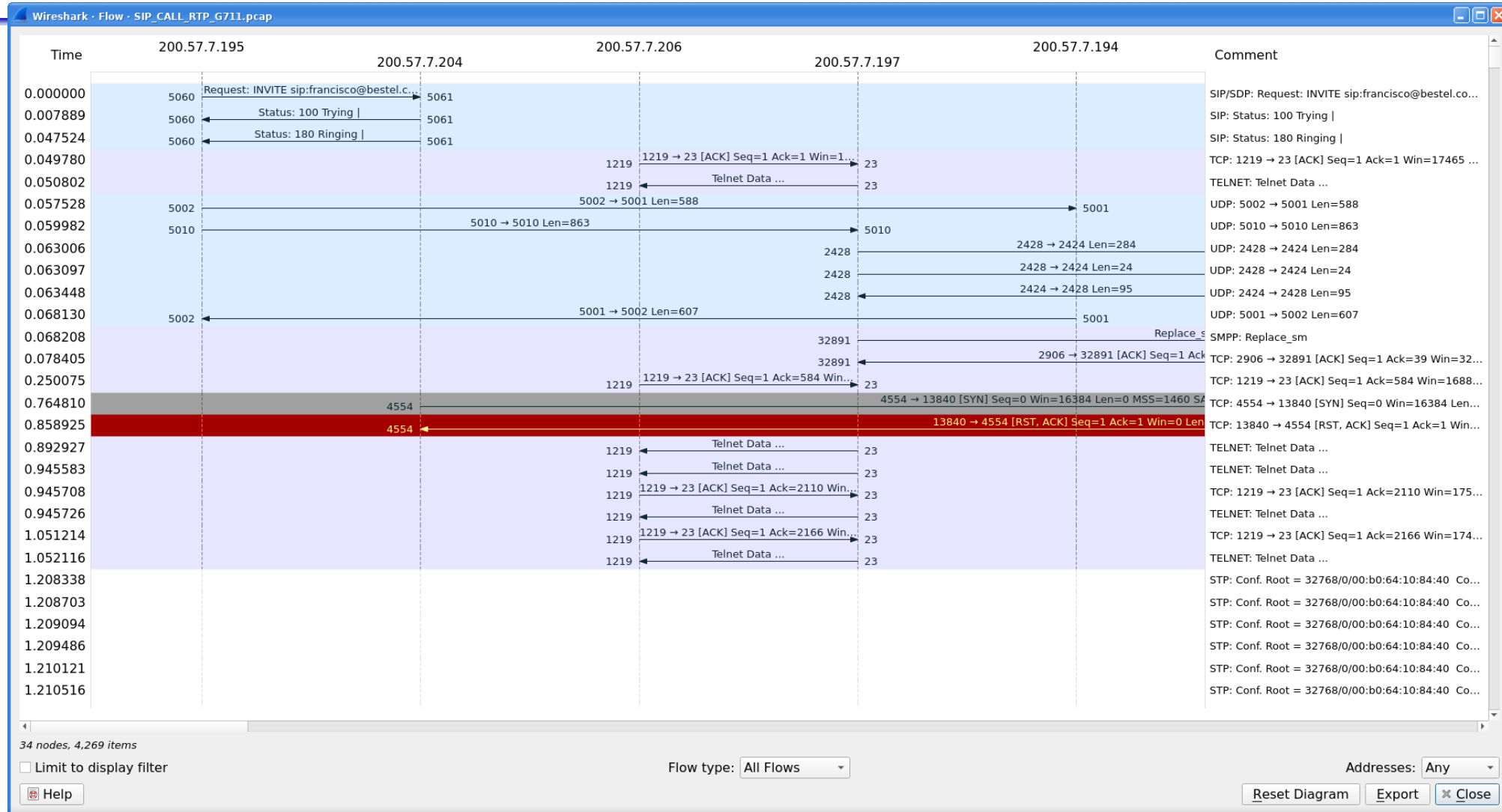
- Receiver capacity
- Network capacity
- Cannot be distinguished easily at all places
- Should be differentiated



TCP Flow Control

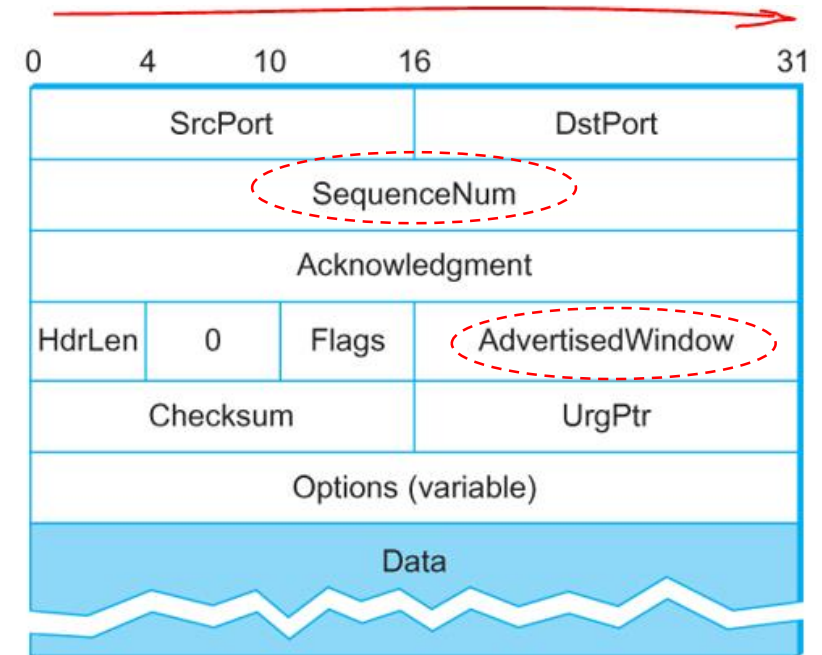
- $\text{LastByteRcvd} - \text{LastByteRead} \leq \text{MaxRcvBuffer}$
- $\text{AdvertisedWindow} = \text{MaxRcvBuffer} - ((\text{NextByteExpected} - 1) - \text{LastByteRead})$
- $\text{LastByteSent} - \text{LastByteAcked} \leq \text{AdvertisedWindow}$
- $\text{EffectiveWindow} = \text{AdvertisedWindow} - (\text{LastByteSent} - \text{LastByteAcked})$
- $\text{LastByteWritten} - \text{LastByteAcked} \leq \text{MaxSendBuffer}$
- If the sending process tries to write y bytes to TCP, but $(\text{LastByteWritten} - \text{LastByteAcked}) + y > \text{MaxSendBuffer}$ then TCP blocks the sending process and does not allow it to generate more data.

TCP flow control in Wireshark



Consequence number's limit: Wraparound

- **SequenceNum**: 32 bits longs $2^{32} - 1 = 4\text{GB}$
- Every packet's sequence number must be unique → the Sequence Number will be exhausted ($> 4\text{GB}$).
- When accessible, the sequence numbers that were previously utilized can be **reused** as needed
- Reusing of sequence numbers is known as the **Wraparound**
- **AdvertisedWindow**: 16 bits long
 - TCP has satisfied the requirement of the sliding
 - window algorithm that is the sequence number
 - space be twice as big as the window size
 - $2^{32} \gg 2 \times 2^{16}$



TCP Header Format

Protecting against Wraparound

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

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

Time until 32-bit sequence number space wraps around.

Keeping the Pipe Full

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

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

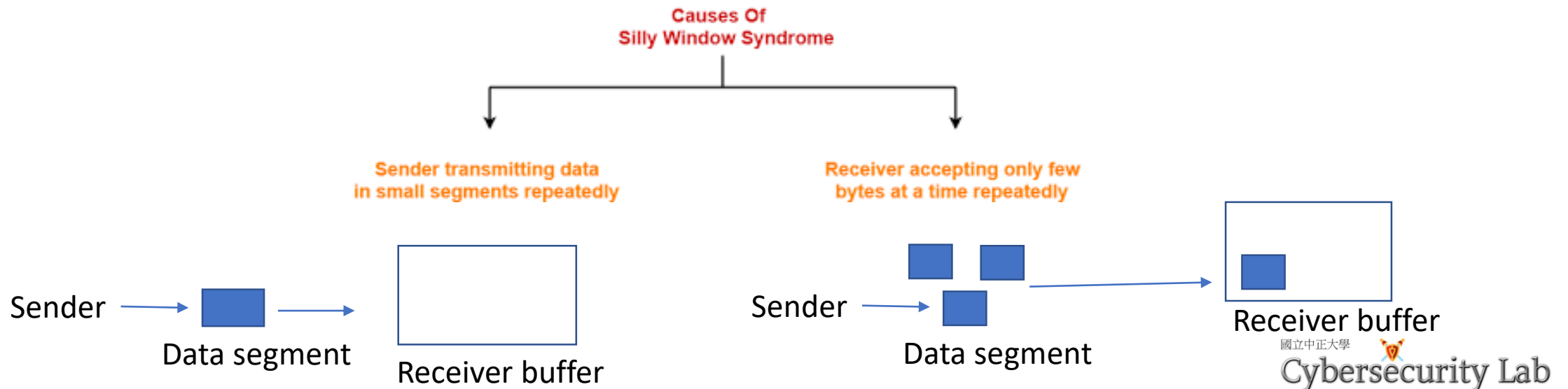
Required window size for 100-ms RTT.

Triggering Transmission

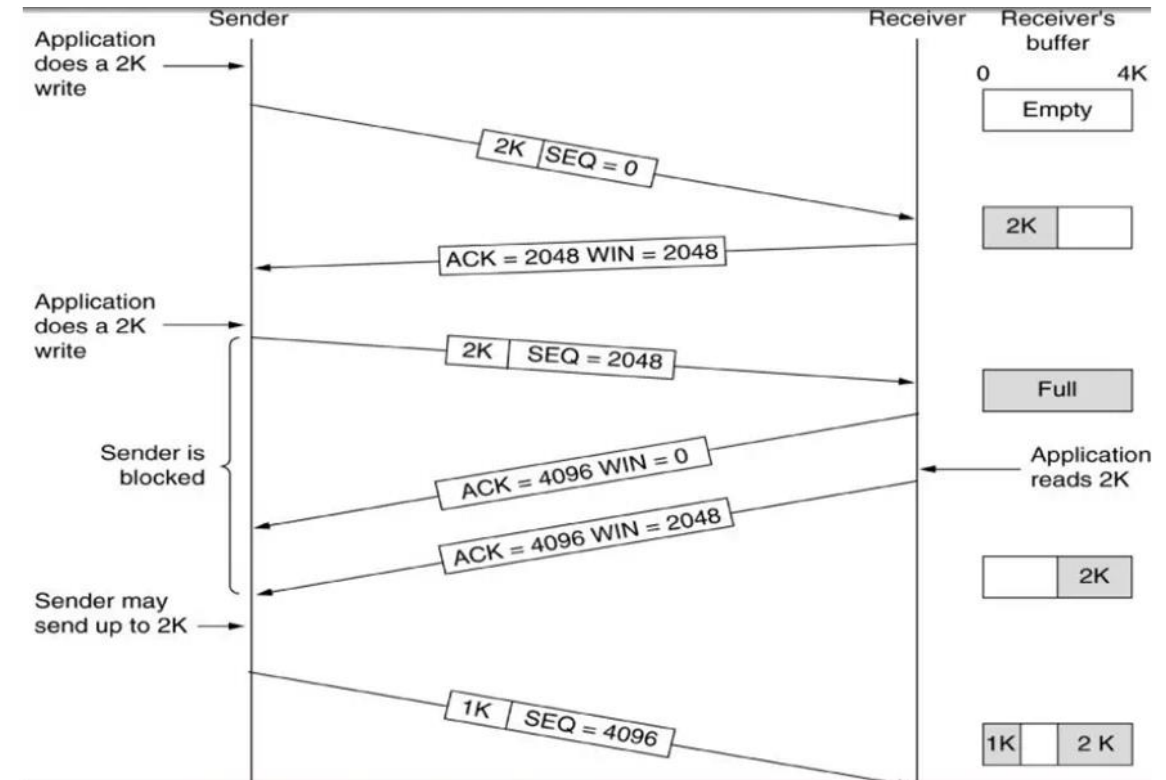
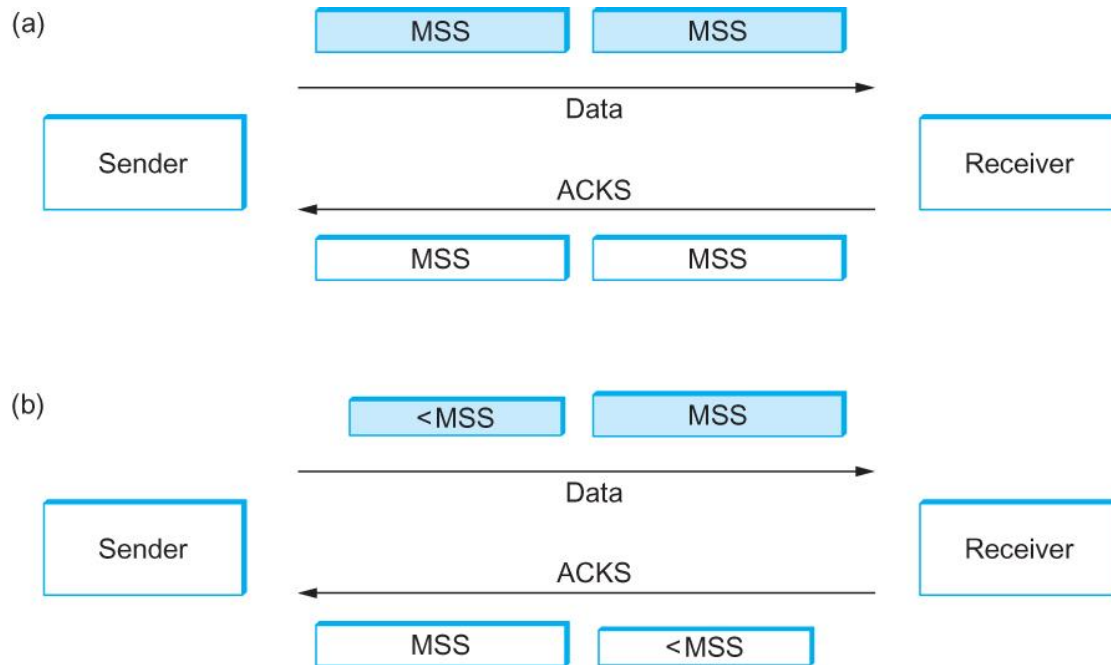
- How does TCP decide to transmit a segment?
 - TCP supports a byte stream abstraction
 - Application programs write bytes into streams
 - It is up to TCP to decide that it has enough bytes to send a segment
- What factors governs this decision
 - Ignore flow control: window is wide open, as would be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

Silly Window Syndrome

- A problem in computer networking caused by poorly implemented TCP flow control
- This problem occurs when the sending application program creates **data slowly**, the receiving application program consumes **data slowly**, or both



Silly Window Syndrome

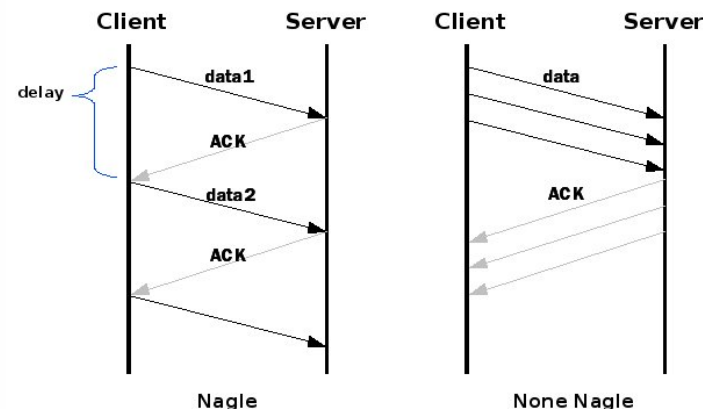


Silly Window Syndrome

Source: Gateoverflow

Nagle's Algorithm

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



Nagle's Algorithm

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

Nagle's Algorithm

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

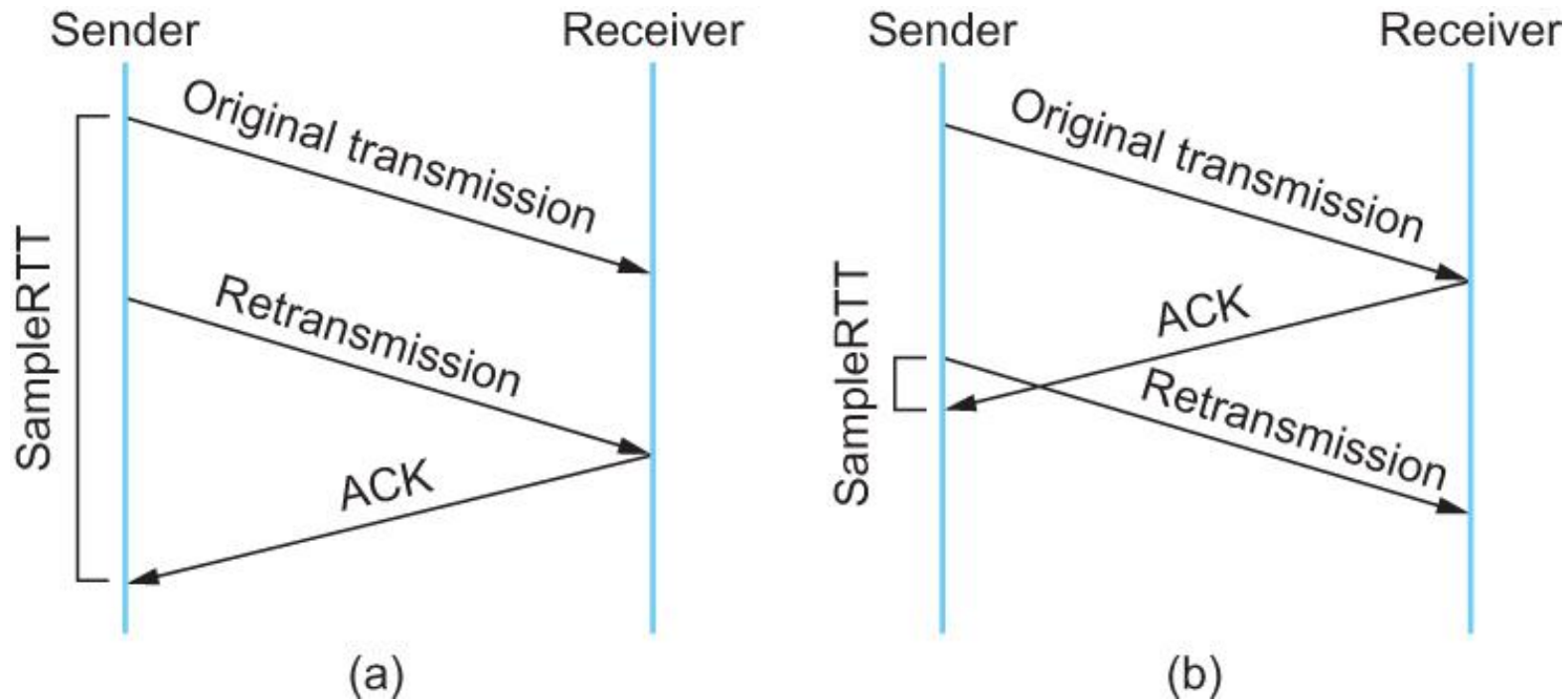
Adaptive Retransmission

- Original Algorithm
 - Measure **SampleRTT** for each segment/ ACK pair
 - Compute weighted average of RTT
 - $\text{EstRTT} = \alpha \times \text{EstRTT} + (1 - \alpha) \times \text{SampleRTT}$
 - α between 0.8 and 0.9
 - Set timeout based on **EstRTT**
 - $\text{TimeOut} = 2 \times \text{EstRTT}$

Original Algorithm

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

Karn/Partridge Algorithm



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

Karn/Partridge Algorithm

- Do not sample RTT when retransmitting
- Double timeout after each retransmission
- Karn-Partridge algorithm was an improvement over the original approach, but it does not eliminate congestion
- We need to understand how timeout is related to congestion
 - If you timeout too soon, you may unnecessarily retransmit a segment which adds load to the network

Karn/Partridge Algorithm

- Main problem with the original computation is that it does not take variance of Sample RTTs into consideration.
- If the variance among Sample RTTs is small
 - Then the Estimated RTT can be better trusted
 - There is no need to multiply this by 2 to compute the timeout
- On the other hand, a large variance in the samples suggest that timeout value should not be tightly coupled to the Estimated RTT
- Jacobson/Karels proposed a new scheme for TCP retransmission

Jacobson/Karels Algorithm

- $\text{Difference} = \text{SampleRTT} - \text{EstimatedRTT}$
- $\text{EstimatedRTT} = \text{EstimatedRTT} + (\alpha \times \text{Difference})$
- $\text{Deviation} = \text{Deviation} + (|\text{Difference}| - \text{Deviation}) \times \beta$
- $\text{TimeOut} = \mu \times \text{EstimatedRTT} + \gamma \times \text{Deviation}$
 - where based on experience, μ is typically set to 1 and γ is set to 4.
Thus, when the variance is small, TimeOut is close to EstimatedRTT; a large variance causes the deviation term to dominate the calculation.

TCP congestion control

- The mechanism that **prevents congestion** from happening or removes it after congestion takes place
- Congestion window state of TCP that **limits the amount of data to be sent** by the sender into the network **even before receiving the acknowledgment**

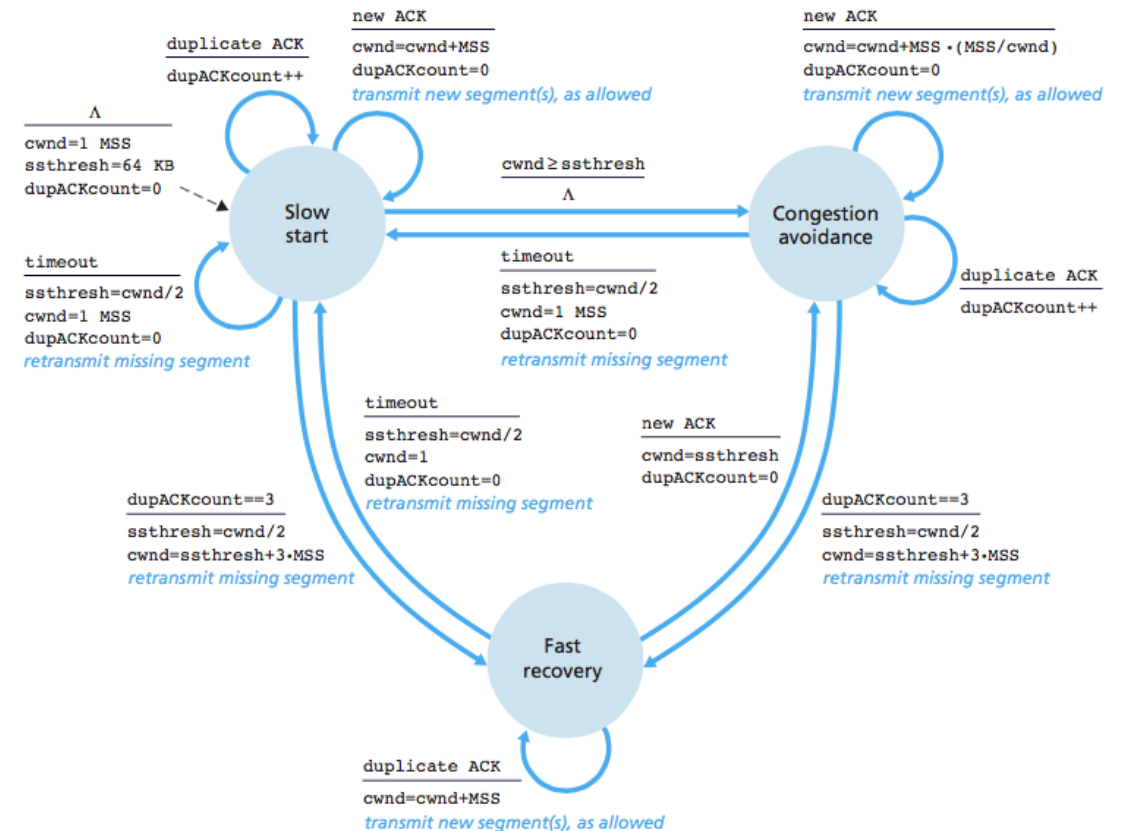
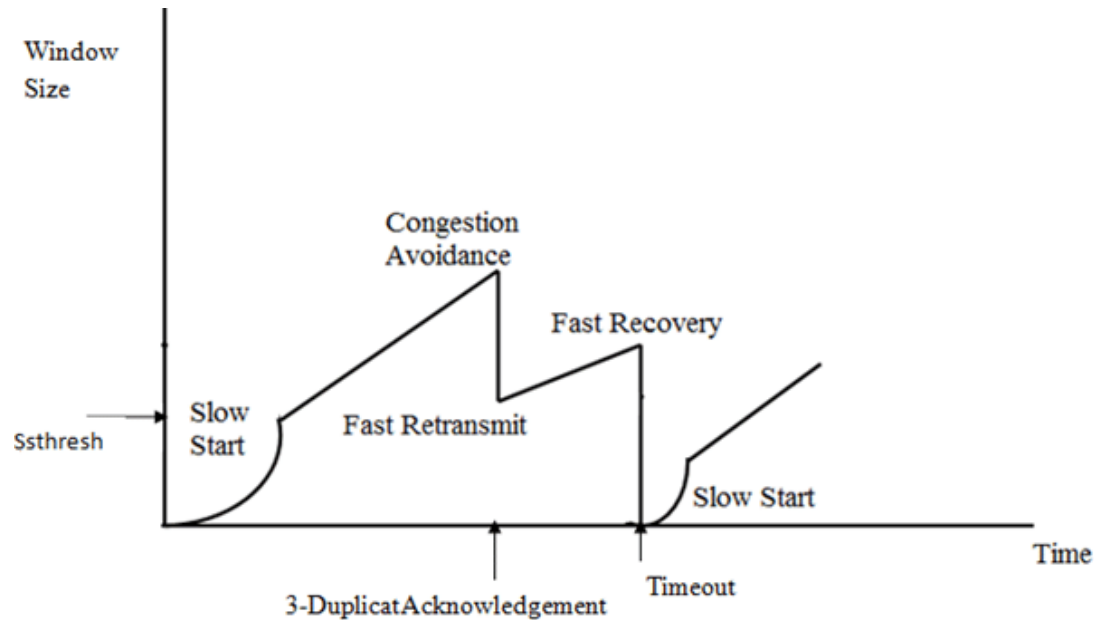
TCP congestion control list

- The naming convention for congestion control algorithms (CCAs) may have originated in a 1996 paper by Kevin Fall and Sally Floyd
- There are many variants of congestion algorithms: TCP Reno/TCP Vegas....
- Efficient TCP congestion control is still a open problem for research

Variant ↕	Feedback ↕	Required changes ↕	Benefits ↕	Fairness ↕
(New) Reno	Loss	—	—	Delay
Vegas	Delay	Sender	Less loss	Proportional
High Speed	Loss	Sender	High bandwidth	
BIC	Loss	Sender	High bandwidth	
CUBIC	Loss	Sender	High bandwidth	
C2TCP ^{[11][12]}	Loss/Delay	Sender	Ultra-low latency and high bandwidth	
NATCP ^[13]	Multi-bit signal	Sender	Near Optimal Performance	
Elastic-TCP	Loss/Delay	Sender	High bandwidth/short & long-distance	
Agile-TCP	Loss	Sender	High bandwidth/short-distance	
H-TCP	Loss	Sender	High bandwidth	
FAST	Delay	Sender	High bandwidth	Proportional
Compound TCP	Loss/Delay	Sender	High bandwidth	Proportional
Westwood	Loss/Delay	Sender	Lossy links	
Jersey	Loss/Delay	Sender	Lossy links	
BBR ^[14]	Delay	Sender	BLVC, Bufferbloat	
CLAMP	Multi-bit signal	Receiver, Router	Variable-rate links	Max-min
TFRC	Loss	Sender, Receiver	No Retransmission	Minimum delay
XCP	Multi-bit signal	Sender, Receiver, Router	BLFC	Max-min
VCP	2-bit signal	Sender, Receiver, Router	BLF	Proportional
MaxNet	Multi-bit signal	Sender, Receiver, Router	BLFSC	Max-min
JetMax	Multi-bit signal	Sender, Receiver, Router	High bandwidth	Max-min
RED	Loss	Router	Reduced delay	
ECN	Single-bit signal	Sender, Receiver, Router	Reduced loss	

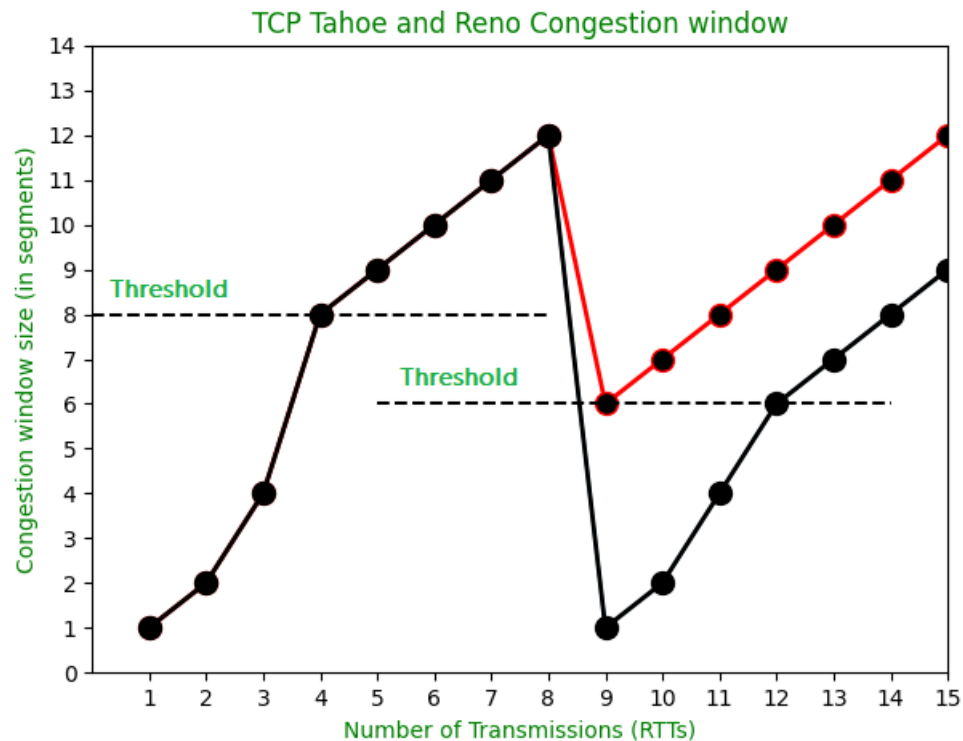
TCP congestion control

- Some famous congestion control mechanisms



TCP congestion control

TCP Reno may be the best choice due to its ability to adapt quickly and avoid congestion



TCP Congestion Control

Algorithms	condition	Design	action
Slow Start	$cwnd \leq ssthres$;	$cwnd$ doubles per RTT	$cwnd += 1MSS$ per ACK
Congestion Avoidance	$cwnd > ssthres$	$cwnd++$ per RTT (additive increase)	$cwnd += 1/cwnd * MSS$ per ACK
fast retx	3 duplicate ACK	reduce the $cwnd$ by half (multiplicative decreasing)	$ssthres = \max(cwnd/2, 2)$ $cwnd = ssthres + 3 MSS$; retx the lost packet
fast recovery	receiving a new ACK after fast retx	finish the 1/2 reduction of $cwnd$ in fast retx/fast recovery	$cwnd = ssthres$; tx if allowed by $cwnd$
	upon a dup ACK after fast retx before fast recovery	("transition phrase")	$cwnd += 1MSS$; Note: it is different from slow start.
RTO timeout	time out	reset	$ssthres = \max(cwnd/2, 2)$ $cwnd = 1$; retx the lost packet

Summary

- We have discussed how to transmit data via transport layer.
- We have discussed UDP
- We have discussed TCP/TCP flow control/congestion control
- We have discussed to use Wireshark to analyze packets
- We have discussed several algorithms to optimize TCP transmission