oueees-202106 topic 08:

Network transports

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On the internet
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Lecture notes and reporting

- https://github.com/jj1bdx/oueees-202106-public/
- Check out the README.md file and the issues!
- Keyword at the end of the talk
- URL for submitting the report at the end of the talk

Topic of this video:

Network transports

IP address and the port number

- Each service has a 16-bit port number
- HTTPS = 443, DNS = 53, SSH = 22, etc.
- A pair of IP address and port number defines an endpoint of communication

UDP and TCP

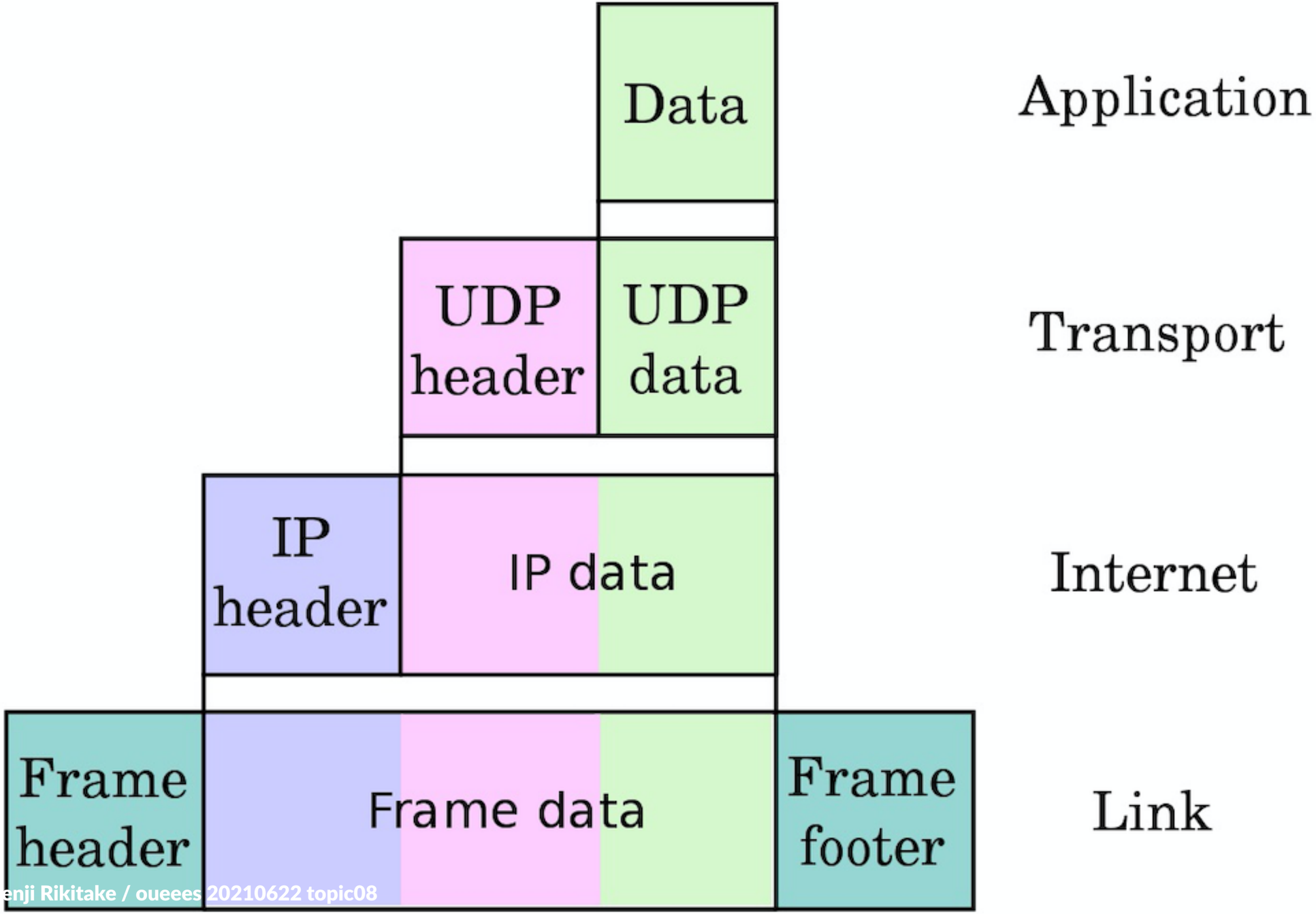
- Two major transport protocols on the internet
- User Datagram Protocol (UDP): connection-less
- Transport Control Protocol (TCP): connection-oriented

Packet exchange limitation

- Packets are not always delivered
- Sending sequence is not preserved
- The same packet may be received multiple times
- The content of the packet may get altered or damaged
- Packet size has the limitation

What UDP does

- Add a header with the port number
- Send it in an IP packet
- ... and that's it

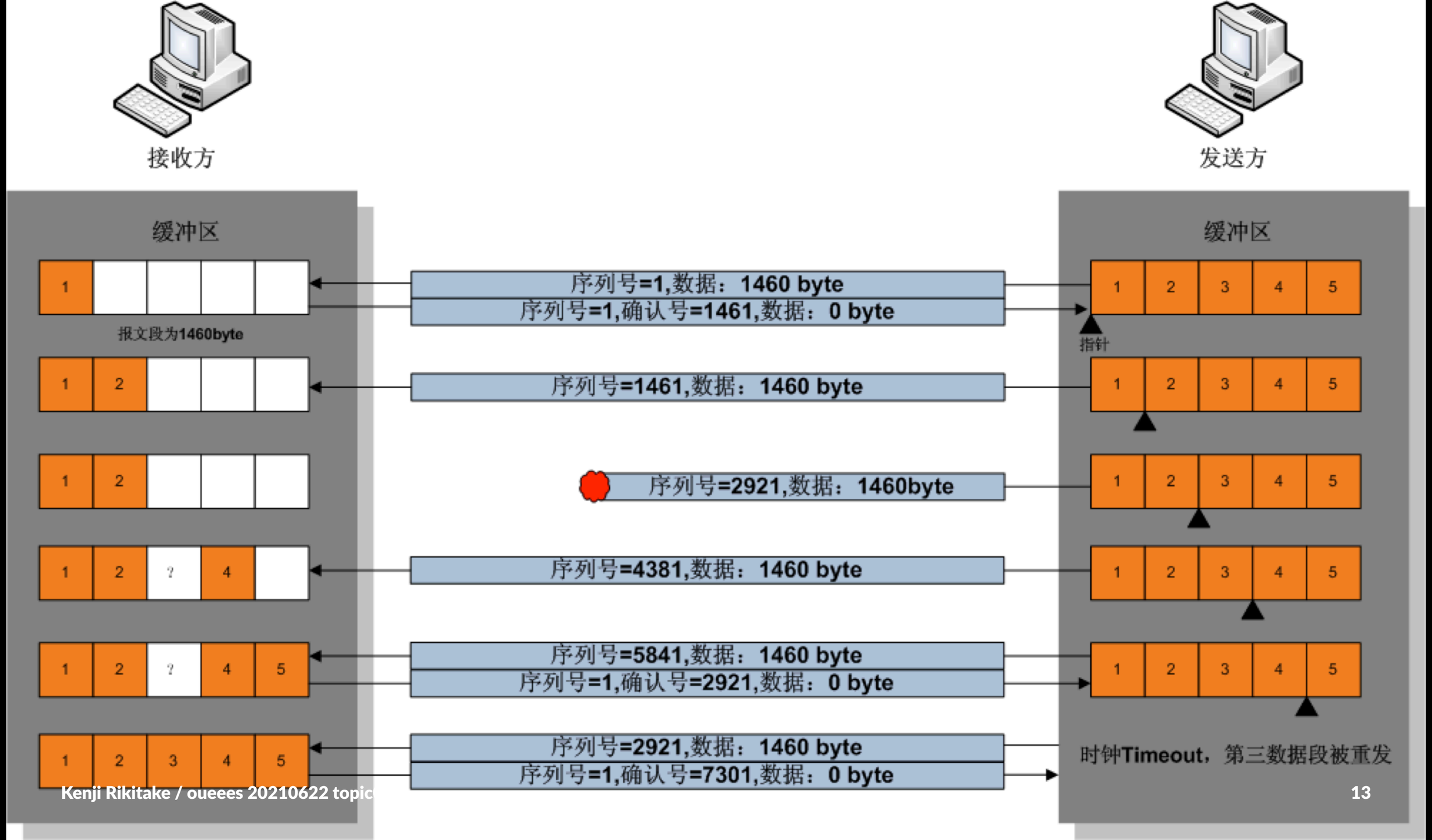

Data
Application
UDP header
UDP data
Transport
IP header
IP data
Internet
Frame header
Frame data
Frame footer
Link

UDP's pros and cons

- UDP datagrams are still not always delivered and may get lost
- Sequence is not preserved
- The same datagram may be received multiple times and may cause duplicate delivery
- The errors in the contents of UDP datagrams are detectable
- UDP datagram has the size limit: suitable for relatively small messages
- Very small additional latency

Transport control protocol (TCP)

- Detect packet loss by timeout
- Split stream into segments
- Put sequence numbers to the segments
- Reassemble segments to the stream
- Perform congestion control


接收方
发送方
缓冲区
缓冲区
1
报文段为1460byte
1 2
1 2
1 2 ? 4
1 2 ? 4 5
1 2 3 4 5
序列号=1,数据：1460 byte
序列号=1,确认号=1461,数据：0 byte
序列号=1461,数据：1460 byte
序列号=2921,数据：1460byte
序列号=4381,数据：1460 byte
序列号=5841,数据：1460 byte
序列号=1,确认号=2921,数据：0 byte
序列号=2921,数据：1460 byte
序列号=1,确认号=7301,数据：0 byte
1 2 3 4 5
指针
时钟Timeout，第三数据段被重发

TCP's pros and cons

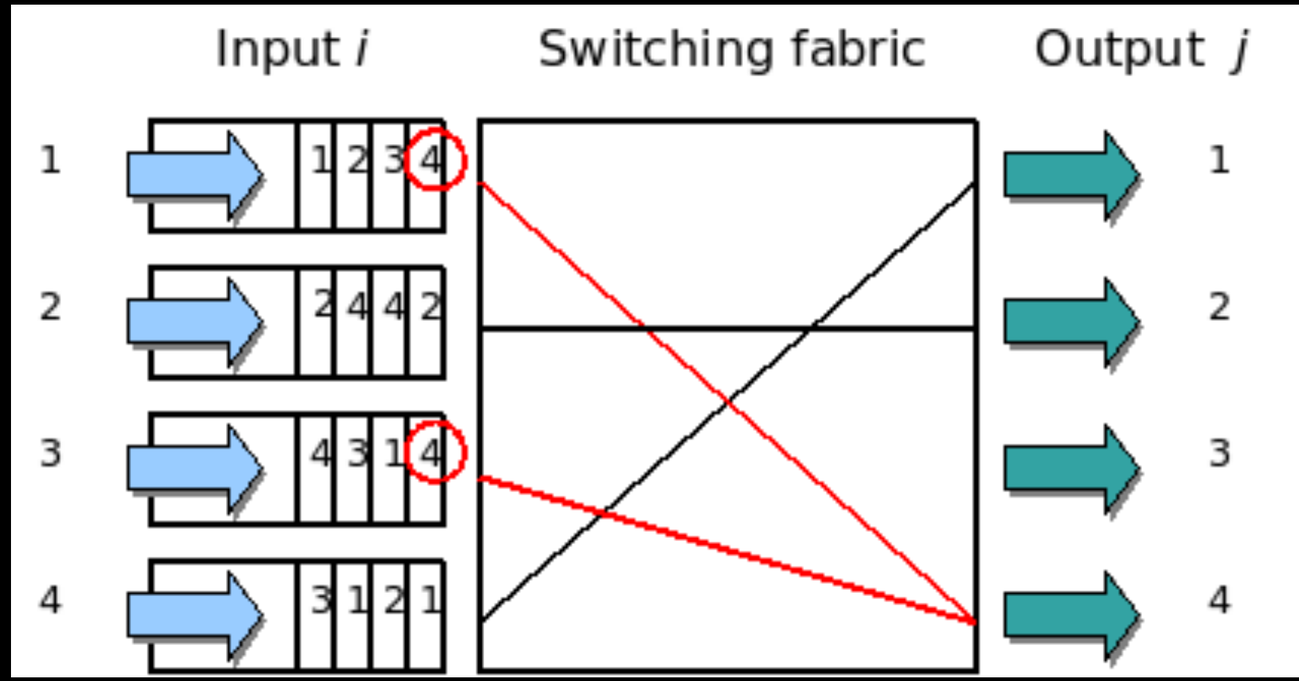
- Loss is detected and recovered so long as the connection is alive
- Sequence is preserved
- No content repetition
- Errors are detected and fixed by retransmission
- The stream will accept data so long as the connection is alive
- Data delivery may delay if retransmission occurs

Web: HTTP/2 (TCP) and QUIC (UDP) -> HTTP/3

- People wants *speed* and *smaller latency*
- HTTP/2 (RFC7540): TCP-bound, stream aggregation and content compression
- QUIC (RFC9000): UDP-based, tightly integrated to HTTP/2 and specific congestion control
- HTTP/2 had *head-of-line blocking problem* by TCP
- Major browsers have already supported HTTP/3 (= HTTP/2 over QUIC)

Buffering and head-of-line (HOL) blocking

- Buffering causes only the oldest packets to be forwarded
- Newer packets could be forwarded without HOL blocking
- In this example, moving buffers to output ports will avoid the delay for Output 3 at Input 1, blocked by the contention of Output 4



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