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at Northeastern University

EECE5155: Wireless Sensor Networks and the Internet of Things

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The Network Layer

The Protocol Stack

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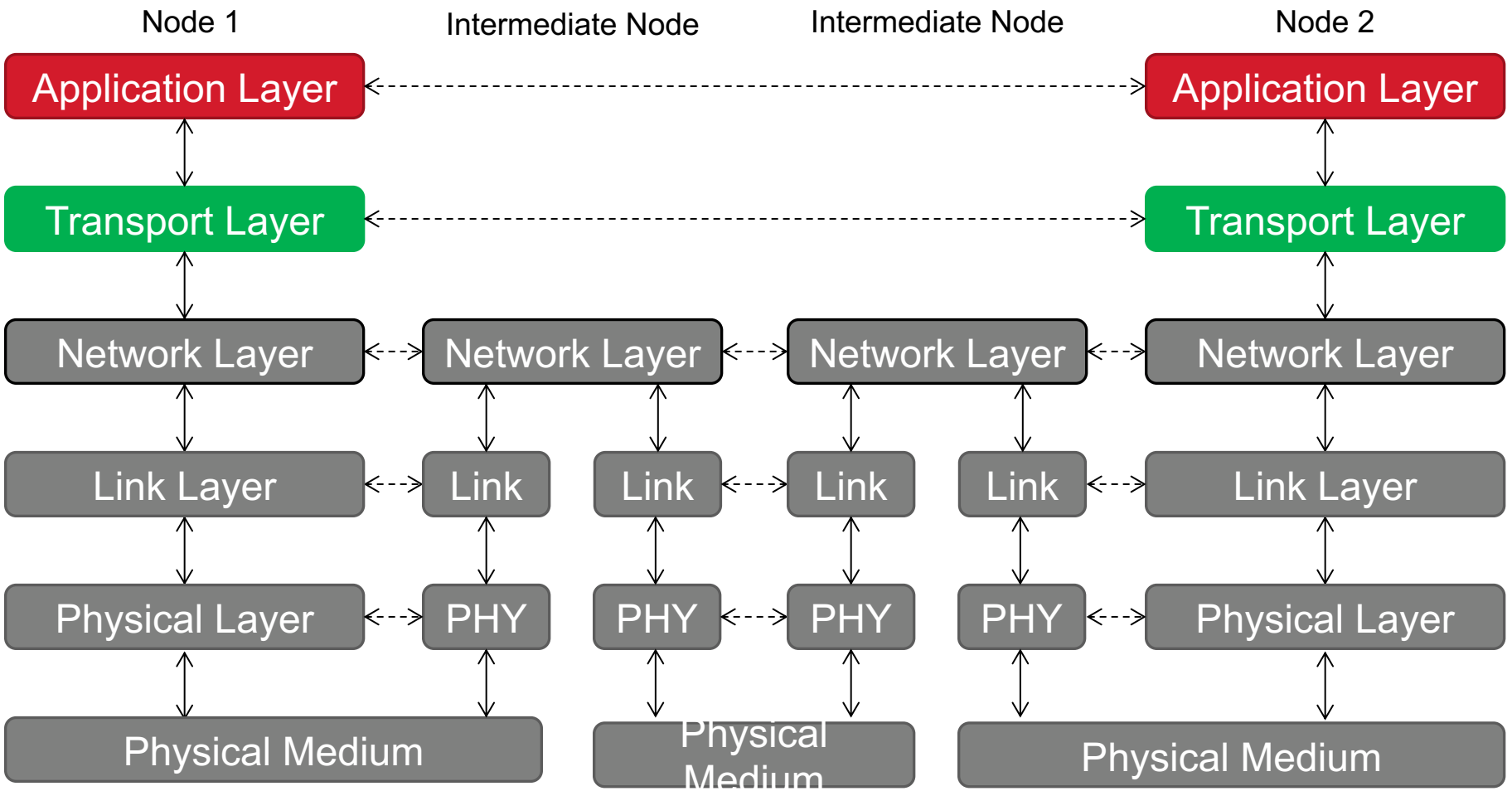


The Transport Layer

- **Objective:** To guarantee the correct end-to-end transmission of information over the network
  - This includes:
    - Reliable data transport
      - Deals with end-to-end packet losses
    - Flow Control
    - Congestion Control
      - Regulates transmission rates at the source to avoid losses due to congestion

Elements of Transport Protocols

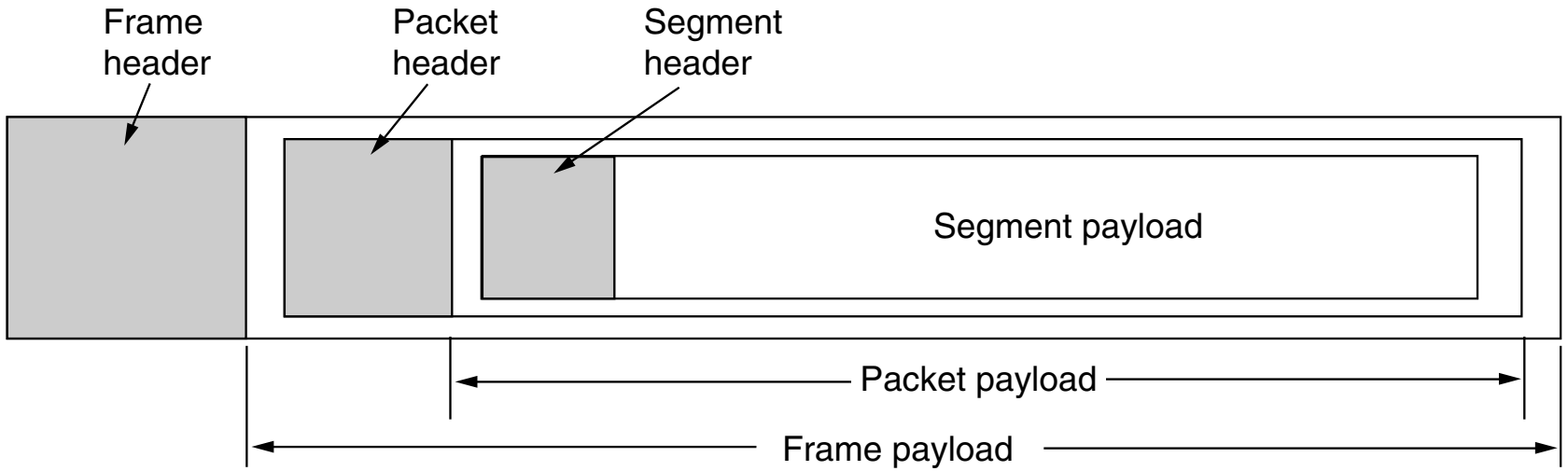
- The transport service is implemented by a transport protocol used between the two transport entities, normally at the two ends of the data path (e.g., **at the client and at the server**)

- In some ways, transport protocols resemble the link layer protocols:
  - Both have to deal with error control, sequencing and flow control, among other issues

- However, significant differences between the two also exist due to major dissimilarities between the environments in which the two protocols operate:
  - At the link layer, two routers communicate directly via a physical channel, whether wired or wireless
  - At the transport layer, this physical channel is replaced by the entire network

Segments

- Messages sent from transport entity to transport entity (compare to packets, frames)



Ports

- So far, we have talked about:
  - Link/MAC addresses at the link layer
    - Identify a node in a link
    - Are usually fixed/static (e.g., your WiFi card has a unique fixed MAC address)
  - Network/IP addresses at the network layer
    - Identify a node in a network
    - Are usually dynamic (e.g., your computer has a different IP address in different networks)
- New:
  - Ports are **addresses at the transport layer**
    - Identify a specific application executed in a node
    - Are generally different at the transmitter and the receiver

Ports

- For many common services, the port at which a server is listening is well-known:
  - **20: FTP (file transfer)**
  - **22: SSH, SFTP (secure file transfer)**
  - **25: SMTP (sending email)**
  - **80: HTTP (browsing)**
  - **110: POP 3 (retrieving email)**
  - **143: IMAP (email)**
  - **194: IRC (IRC chat)**
  - **443: HTTPS (secure HTTP)**
  - **…**

Common Transport Layer Protocols

- User Datagram Protocol (UDP)
- Transport Control Protocol (TCP)

User Datagram Protocol (UDP)

- **Objective:** To provide a connection-less transport service
  - Very little functionality beyond sending packets between applications
  - Lets applications build their own protocols on top as needed!

- Defined in RFC 768 (1981)

UDP: Segment Structure

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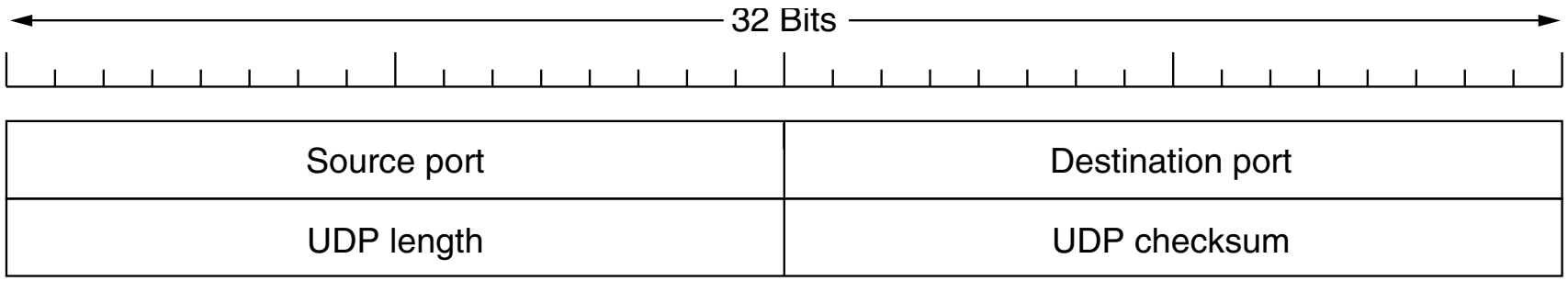
- Composed by:
  - 8-byte header:
    - Source Port
    - Destination Port
    - Length: from 8 bytes to 65,515 bytes
    - Checksum: 16-byte CRC
  - Payload



UDP: Ports

- The **main advantage of UDP over just IP** is the addition of the **source and destination ports**
  - Without the port fields, the transport layer would not know what to do with each incoming packet

- The receiver of a UDP segment must have a socket bound to a specific port
  - When a UDP packet arrives, its **payload is handed to the process attached to the destination port**

- The sender of a UDP segment must have a socket bound to another specific port too, mainly to be able to receive the reply from the destination

Transmission Control Protocol (TCP)

- Formally defined in RFC 793 (1981)
  - Plus several amendments, e.g., RFC 1122, RFC 1323, RFC 2018, …

- **Objective:** To provide a reliable end-to-end byte stream over an unreliable internetwork

TCP: Service Model

- All TCP connections are full duplex and point-to-point:
  - Remember:
    - Full duplex: traffic can go in both directions at the same time
    - Point-to-point: each connection has exactly two end points
- TCP does not support multicasting or broadcasting
- A TCP connection is a byte stream, not a message stream

TCP: Service Model

- When an application passes data to TCP, **TCP may send it immediately or buffer it** (in order to collect a larger amount to send at once)

- However, sometimes the application really wants the data to be sent immediately:
  - For example, suppose a user of an interactive game wants to send a stream of updates and it is essential that the updates be sent immediately

- To force data out, TCP has the notion of a **PUSH flag** that is carried on packets, whose original intent was to let applications tell TCP implementations not to delay the transmission

- However, applications cannot literally set the PUSH flag when they send data. Instead, different operating systems have evolved different options to expedite transmission (e.g., TCP NODELAY in Windows and Linux)

TCP: Protocol Overview

- A **TCP segment** consists of a fixed 20-byte header (plus an optional part) followed by zero or more data bytes

- The TCP software decides how big segments should be in order **to avoid packet fragmentation**:
  - It can either accumulate data into one segment or can split data from one write over multiple segments
  - Two limits need to be satisfied to avoid fragmentation:
    - On the one hand, each segment, including the TCP header, must fit in the 65,515-byte IP payload
    - On the other hand, each segment must fit in the MTU for all the links between the sender and the receiver
      - In practice, the MTU is generally 1500 bytes (the Ethernet payload size) and thus defines the upper bound on segment size
      - Path MTU discovery is used (Chapter 5 – Part 2, Slide 20)

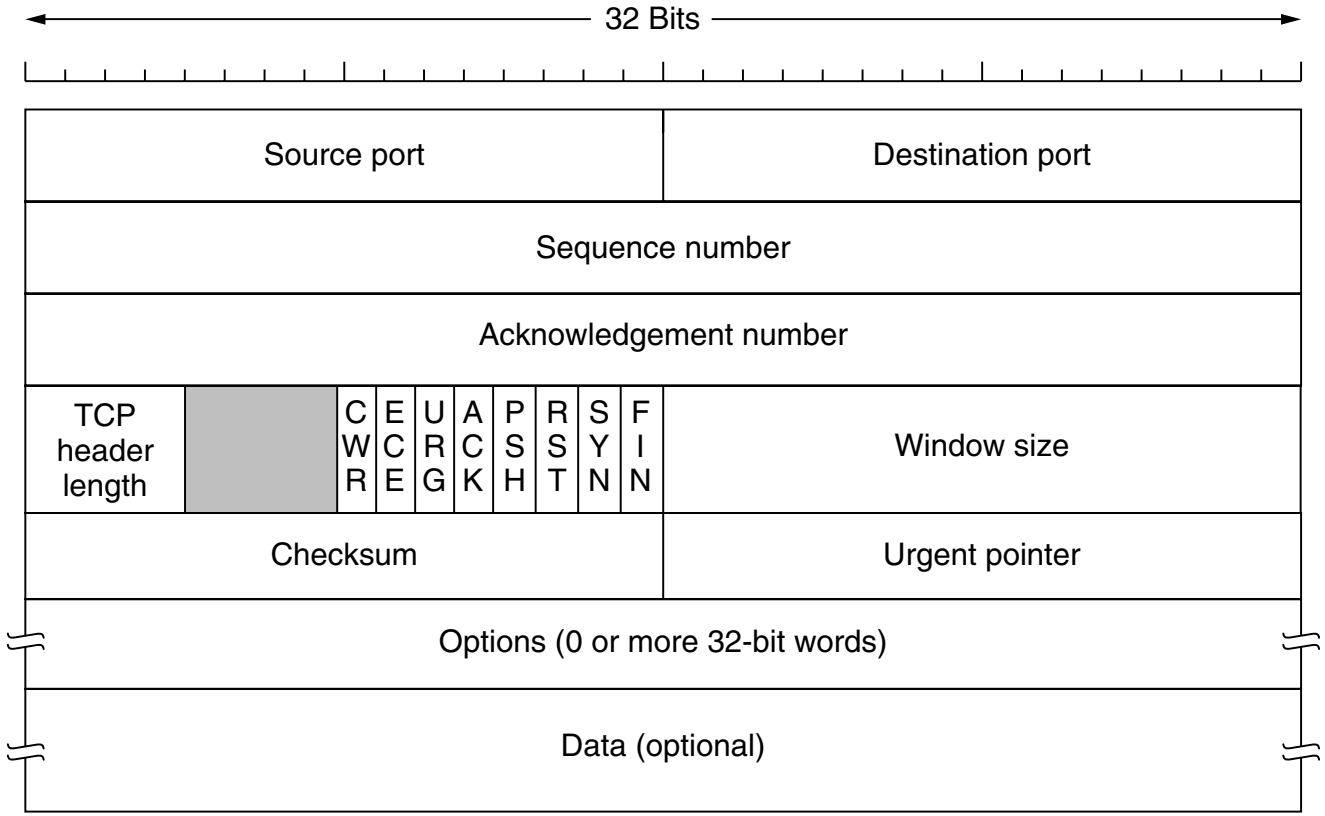
TCP: Protocol Overview

- The basic protocol used by TCP entities is the sliding window protocol with a dynamic window size:
  - When a sender transmits a segment, it also starts a timer
  - When the segment arrives at the destination, the receiving TCP entity sends back a segment (with data if any exist) and an acknowledgement number equal to the next sequence number it expects to receive and the remaining window size
  - If the sender's timer goes off before the acknowledgement is received, the sender transmits the segment again

- Some fine tuning is needed because
  - Segments can arrive out of order
  - Segments can also be delayed so long in transit that the sender times out and retransmits them
    - The retransmissions may include different byte ranges than the original transmission, requiring careful administration to keep track of the bytes!

TCP: Segment Header

- 20 byte long



TCP: Segment Header

- **Source Port**
- **Destination Port**
- **Sequence Number**
- **Acknowledgement Number:** Cumulative acknowledgement
- **TCP Header Length:** Indicates how many 32-bit words are contained in the TCP header
- 4 empty bits!
- 8 flags:
  - Explicit Congestion Notification (ECN) Echo (**ECE**): To notify the sender that it should slow down
  - *Congestion Window Reduced (**CWR**):* To notify the receiver that the sender has slowed down its transmission
  - *Urgent pointer (**URG**):* To indicate that the **urgent pointer** is active (this shows a byte offset from the current sequence number at which urgent data are to be found)
  - **ACK**: To indicate the acknowledgement number is valid

TCP: Segment Header

- Pushed Data (**PSH**): To indicate pushed data
- Reset (**RST**): To abruptly reset a connection due a host crash or similar conditions
- **SYN**: To establish connections
- **FIN**: To release a connection

- **Window size:** Indicates how many bytes may be sent starting at the byte acknowledged
  - Note that now acknowledgements and windows size are decoupled:
    - A node can say "I have properly received till byte XX, but I don't want any more data. Thanks."
- **Checksum**
- **Options:**
  - **Maximum Segment Size (MSS):** in case the receiver cannot handle large segments
  - **Selective Acknowledgement (SACK):** instead of the cumulative acknowledgement
  - **…**

TCP: Connection Establishment

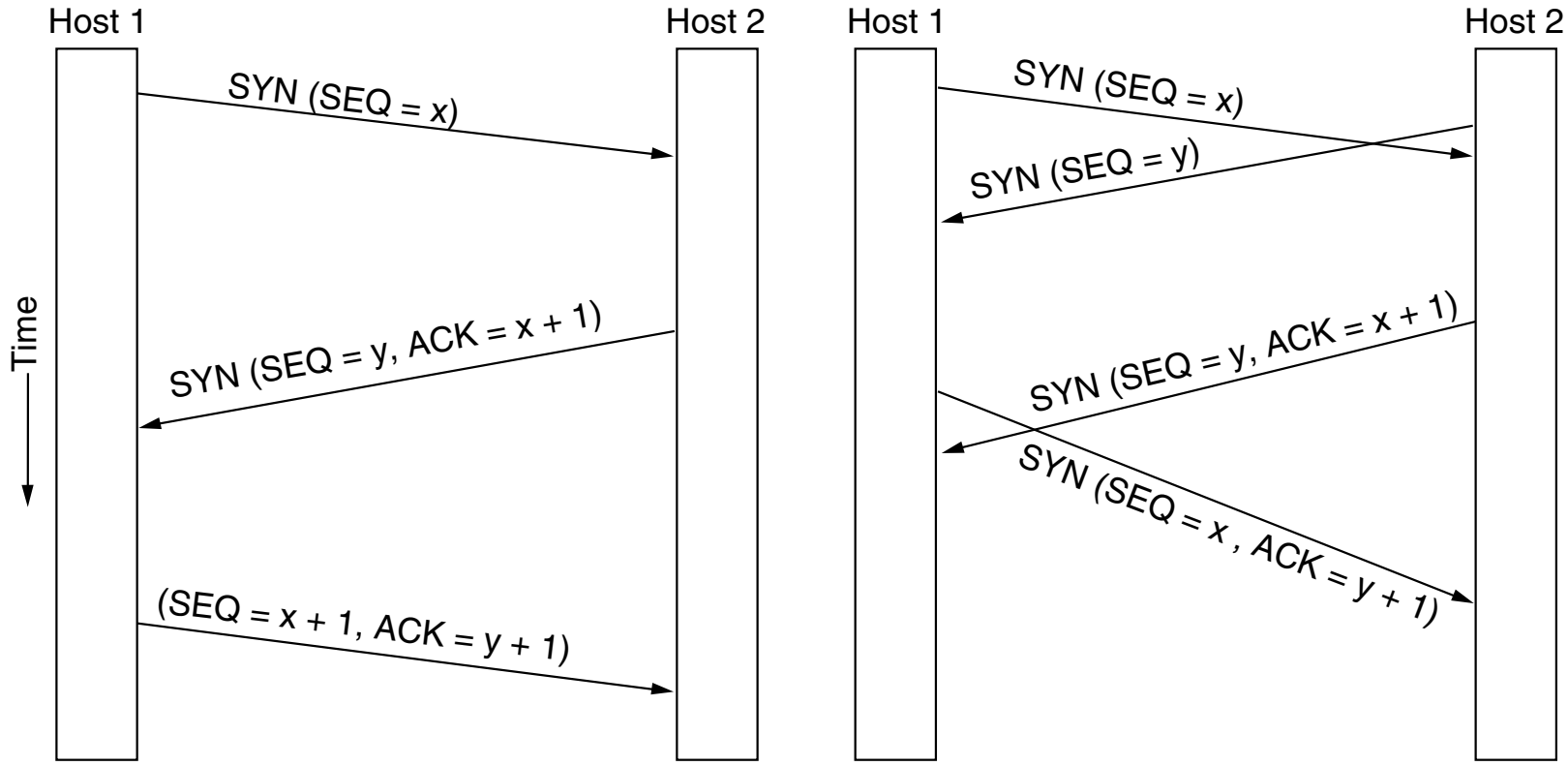
- Based on the three-way handshake explained before:
  - To establish a connection, one side, i.e., the server, passively waits for an incoming connection by executing the LISTEN and ACCEPT primitives
  - The other side, i.e., the client, executes a CONNECT primitive, specifying:
    - The IP address and port to which it wants to connect
    - The maximum TCP segment size it is willing to accept
    - Optionally some user data (e.g., a password)
  - The CONNECT primitive sends a TCP segment with the *SYN* bit on and *ACK* bit off and waits for a response

TCP: Connection Establishment

- When the CONNECT segment arrives at the destination, the TCP entity there checks to see if there is a process that has done a LISTEN on the port given in the Destination port field
  - If not, it sends a reply with the RST bit on to reject the connection
  - If some process is listening to the port, that process is given the incoming TCP segment:
    - It can either accept or reject the connection:
      - If it accepts, an acknowledgement segment is sent back

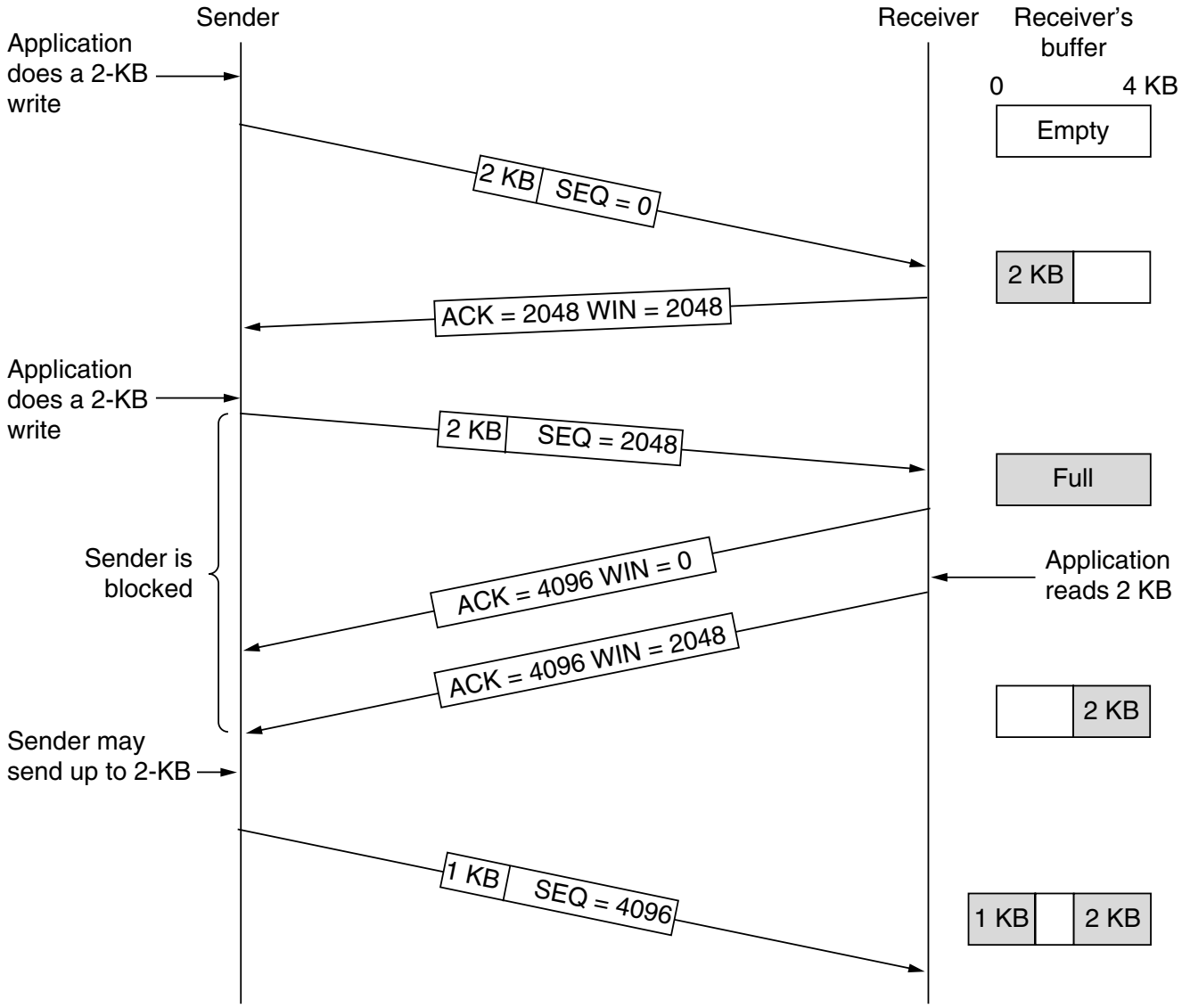
- In the event that two hosts simultaneously attempt to establish a connection between the same two sockets,
  - Just one connection is established, not two, because connections are identified by their end points
  - If the first setup results in a connection identified by (*x*, *y*) and the second one does too, only one table entry is made, namely, for (*x*, *y*).

TCP: Connection Establishment



TCP: Sliding Window

- There are two windows in TCP:
  - **Congestion window:** tries not to exceed the capacity of the network (congestion control)
  - **Receiver window:** tries not to exceed the capacity of the receiver to process data
- The details are beyond the scope of this course.



TCP: Connection Release

- Although TCP connections are full duplex, to understand how connections are released it is best to think of them as a pair of simplex connections
  - Each simplex connection is released independently of its sibling
- To release a connection,
  - Either party can send a TCP segment with the *FIN* bit set, which means that it has no more data to transmit
  - When the *FIN* is acknowledged, that direction is shut down for new data
- Data may continue to flow indefinitely in the other direction
- When both directions have been shut down, the connection is released
- Normally, four TCP segments are needed to release a connection: one *FIN* and one *ACK* for each direction
  - However, it is possible for the first *ACK* and the second *FIN* to be contained in the same segment, reducing the total count to three

Focus on IoT

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- "Things" have unique constraints for dependable data delivery that impact the transport layer
  - Limited energy (battery powered!)
  - Limited computational resources in a node
  - Limited memory
  - Limited dependability of individual nodes
  - Transmission errors over a wireless channel

- Standard solution to increase end-to-end reliability: **Redundancy**
  - Redundancy in nodes, transmission
  - Forward and backward error recovery
  - Combinations are necessary

Constraints

- Energy
  - Send as few packets as possible
  - Send with low power → high error rates
  - Avoid retransmissions
  - Short packets → weak FEC
  - Balance energy consumption in network
- Processing power
  - Only simple FEC schemes
  - No complicated algorithms (coding)
- Memory
  - Store as little data as briefly as possible

Delivering Single Packets

- What are the intended receivers?
  - A single receiver?
  - Multiple receivers?
    - In close vicinity? Spread out?
  - Mobile?

- Which routing structures are available?
  - Unicast routing along a single path?
  - Routing with multiple paths between source/destination pairs?
  - No routing structure at all – rely on flooding/gossiping?

Single Packet, Single Receiver, Single Path

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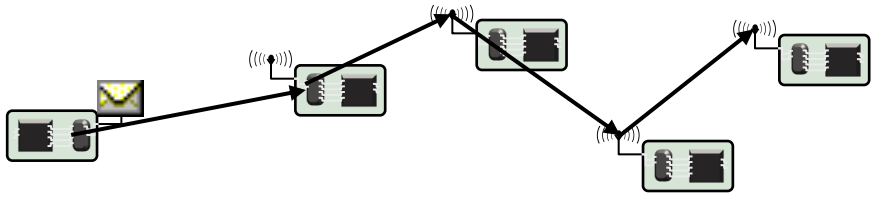
- Single, multi-hop path is given by routing protocol



- Issues: Which node
  - Detects losses (using which indicators)?
  - Requests retransmissions?
  - Carries out retransmissions?

Single Packet, Single Receiver, Single Path

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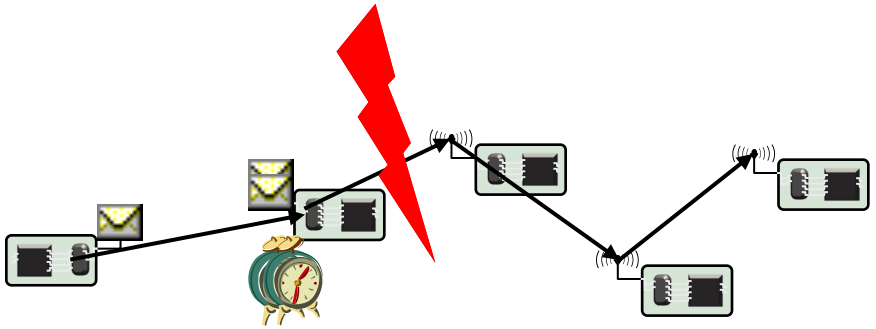
- Detecting loss of a single packet:
  Only positive acknowledgements (ACK) feasible
  - Negative acks (NACK) not an option
  - Receiver usually does not know a packet should have arrived, has no incentive to send a NACK

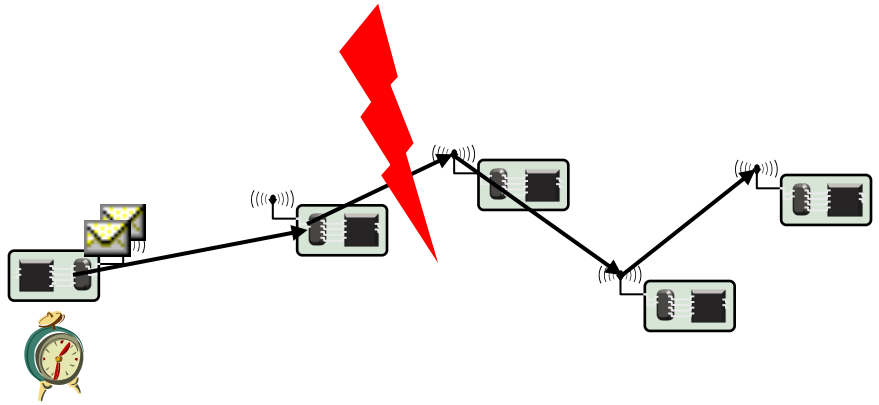
- Which node sends ACKs (avoiding retransmissions)?
  - At each intermediate node, at MAC/link level
    - Usually accompanied by link layer retransmissions
    - Usually, only a bounded number of attempts
  - At the destination node
    - Transport layer retransmissions
    - Problem: Timer selection

Retransmissions

- For link layer acknowledgements: Neighboring node



- For transport layer acknowledgements:
  - Source node → end-to-end retransmissions



Delivering Blocks of Packets

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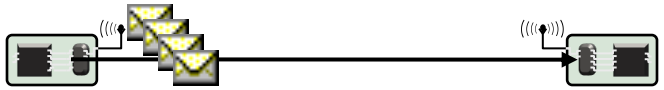
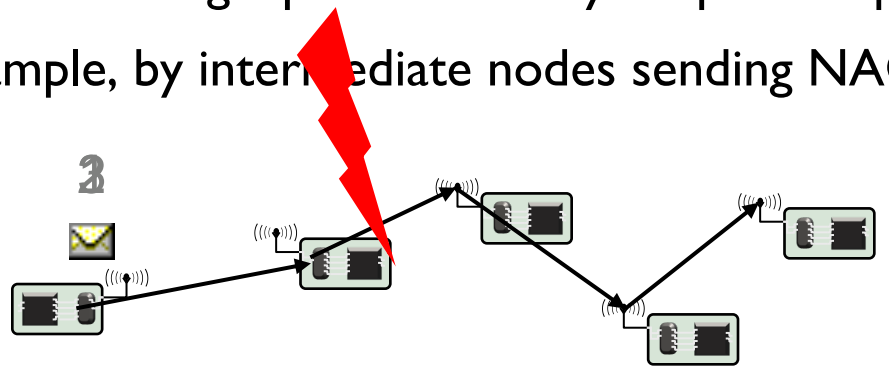
- Goal: Deliver large amounts of data
  - E.g., code update, large observations
  - Split data into several packets (reduce packet error rate)
  - Transfer this block of packets



- Main difference to single packet delivery: Gaps in sequence number can be detected and exploited
  - For example, by intermediate nodes sending NACKs



Where is packet 2?

- To answer NACK locally, intermediate nodes must cache packets
- Which packets? For how long?

Many Solutions

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- Once again, in the 90s and early 2000s, many solutions were developed for the Wireless Sensor Networks (the IoT ancestor), tackling different problems in different ways

- Examples:
  - Pump Slowly Fetch Quickly (PSFQ)
  - Reliable Multisegment Transport (RMST)
  - Event-to-Sink Reliability (ESRT)

- The reality:
  - With all the current IoT solutions, "Things" are usually one-hop away from an IP network
  - As we will see when in Module T5, almost everything goes on top of TCP or UDP

Next Steps

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- Research paper:
  - Due on November 20

- Coming soon:
  - Homework assignment (numerical)
  - Online (open-book) quiz (multiple choice questions)