Units: Speed - K = 10^3, M=10^6, G=10^9; Size – K =2^{10}, M = 2^{20}, G = 2^{30}; 8 bits in a byte; b for bits and B for bytes; 1 Mib = 128KiB, 1 Mib = 2^{20} = 1048576 b = /2^{10} = 1024 Kib = 128 KiB
1. Application: Supporting network applications. Determines destination IP, supports network apps, decides which data will be sent. Types: **HTTP:** Used for communication of hosted info on web; **FTP:** Used to transfer files without web interface; **SSH**; **Telnet; SMTP/IMAP/POP3**: Email; **DNS:** Domain name resolution (google.com to ip); **SCCP:** VoIP. Must specify message fields & structures, message sending rules, types of messages exchanged, message response rules, message semantics.
2. Transport: Process-Process data transfer (TCP, UDP)
3. Network: Routing of datagrams from source to destination (IP, Routing protocols)
4. Link: Data transfer between neighboring network elements (PPP, Ethernet)
5. Physical: Carries actual signals between devices (cable, wireless)
Encapsulation: As a packet is being constructed and passed down to the next layer of the internet protocol stack, a new header is added.
Application Data: The payload (non-header portion) of transport layer.
Transport layer: Logical communication between processes: Relies on, enhances, network layer services, The transport-layer unit is called a segment
Network layer: Logical communication between hosts:
- Relies on, enhances, link layer services, The network-layer unit is called a datagram network-layer protocols run: Laptops, mobile devices, PCs, routers, end systems
Sender side: Get segments from transport layer, Encapsulates segments into datagrams
Routers examine header fields in all IP datagrams
Receiver side: Delivers segments to transport layer
<u>**Two Major Network-Layer Functions**</u>
Routing: Determine route taken by packets from source to destination,Routing algorithms
Forwarding: Move packets from router's input to appropriate router output
Routing and forwarding. - 1) Routing algorithm determines path through network 2) Forwarding table determines local forwarding at this router 3) Link to first hop in path from this router to final destination
<u>**What are the causes/costs of network congestion?**</u>
As utilization approaches maximum, delay increases
Causes: End systems are sending to much data too fast for network/routers to handle.
Costs: Delayed/dropped packets (buffer overflow at routers) 1. Long delays from queuing in router buffers (causing queues to fill up which causes incoming packets to be dropped) 2. Dropped packets (everything used for those packets are wasted) 3. Slow acknowledgements cause unnecessary retransmission (waste of resources and tretransmission compounds the problem) 4. Eventual congestion collapse
<u>**Optimizing Utilization**</u> Smarter timeouts, Better sliding window size management, More efficient acknowledgements
<u>**Smarter timeouts: TCP Round Trip Time (RTT) and TImeout**</u>
TCP countdown timer: RTT: Can only be estimated and varies from one packet to another, Timer is too short:, Premature timeout, Unnecessary retransmissions, Timer too long: Slow reaction to segment loss, Just right timer can improve efficiency
Estimating the RTT: SampleRTT: Measured time from end of segment transmission until receipt of ACK, Ignore retransmissions, Can use handshake timing, Can vary widely
For smother estimated RTT: Average several recent measurements, not just current SampleRTT
<u>**Setting the RCP Timeout**</u>
- Best to add a safety margin to EstimatedTT (arge variation in EstimatedRTT -> larger safety margin, See how much SampleRT deviates from EstimatedRTT
DevRTT(n) = (1-Beta)DevRTT(n-1) + Beta(SampleRTT-EstimatedRTT) Typically Beta = 0.25
Then set the timeout interval: TimeoutInterval = EstimatedRTT + 4*DevRTT
Approaches to detecting congestion 1. Inferred by end to end systems: Delay/loss & Used by TCP; 2. Network core assistance: Routers provide feedback to senders, More later about ICMP (Internet Control Messaging Protocol)
<u>**Summary of TCP Congestion control**</u>
When CongWin is below Limit, sender is in slow-start phase; + CongWin grows exponentially. Whn CongWin is above limit, sender is in congestion-avoidance phase + CongWin grows linearly. When a triple duplicate ACK occurs, Limit set to CongWin/2 and CongWin set to Limit. + Back to slow-start or congestion-avoidance. When timeout occurs, Limit set to CongWin/2 and CongWin is set to 1 MSS. + Back to slowStart
Steps in TCP 3-way handshake
1. Client listen: Choose initial sequence number (x) and send SYN message
2. Server Listen: Choose initial sequenc enumber (y). Send SYNACK message
3. Client receives SYNACK (x + 1) which indicates server is live. Send ACK. Segment may contain client to server data.
4. Server receives ACK (y + 1) which indicates the client is live.
Steps in TCP closing connection 1. Client ESTAB: clientSocket.close(, 2. Server CLOSE_WAIT: Send ACK message. Can still send Data, 3. Client FIN_WAIT_1: Can no longer send, but can receive data, 4. Client FIN_WAIT_2: Wait for server clos, 5. Server LAST_ACK: Can no longer send data, 6. Client TIMED_WAIT: Timed wait (in case ACK is lost), 7. Server CLOSED, 8. Client CLOSED
<u>**Compare UDP and TCP**</u> UDP provides: Connectionless service, End to end best effort (unreliable) delivery, simple and fast (8-byte header)
TCP provides: Connection-oriented service, End to end reliable byte stream delivery, Complicated (20-byte header), Adds a huge amount of overhead to achieve reliablility Depends on extremely fast and reiable hardware in the network core
Both: Use IP (network layer) for delivery to destination, Send to and receive from multiple applications, De-multiplex to destination application protocol ports

<u>**Weekly Summary Qs From W6-W10**</u>
A TCP fast-retransmit will occur after *Three duplicate ACKS for the same segment.*
TCP Receiver Action: Arrival of segment that partially or completely fills in gap in received ata. = *Immediately send ACK, provided that segment starts at the lower end of gap.*
TCP Connection initialization – 1) Client sends segment with SYN set to 1, 2) Server sends segment with SYN set to 1 and ACK set to the Client's SeqNum+1, 3) Client sends segment with SYN set to 0 and ACK set to the Server's SeqNum+1
The TCP countdown timer is used to **implement reliable data transmission**. | Flow control primarily intends to **keep a TCP sender from overwhelming a receiver's buffer.**
In a **Selective** acknowledgement scheme, a received ACK indicates only that the ACK'd segment was received.| A **Selective Repeat** -type retransmission protocol will retransmit one segment at a time upon a countdown timer interrupt.| The rate of CongWin size increase (in terms of MSS) while in TCP's Congestion Avoidance phase is **Linear**.
In **network-assisted** congestion control, flags may be set during transit which indicate the presence and/or level of congestion in certain portions of the network.
<u>What are some possible consequences of network-core congestion? (Check all that apply)</u>
Network collapse, Dropped Packets, Out-of-order packet arrival, Increased network congestion, Delayed packets| Less overhead than a VC network, faster delivery (below)
Which of the following are benefits of a datagram network? (see above line 2nd half)
Which of the following are benefits of a virtual circuit network?
Guaranteed timing, guaranteed bandwidth, Connection states are preserved.
The process of moving a datagram from a router's input port to output port is called **forwarding**. | A router's routing table is output by a **routing algorithm.|** The process of determining a path through the internet is called **routing**, handled by **routing algorithm**.
True: T TCP implements network fairness indirectly. TCP has cc mechanism. | The Internet Protocol (IP) header may be **28** bytes long.| Routing would be more complicated if we used hardware addresses as network addresses.| The "traceroute" application (on Windows) receives ICMP messages.| The "traceroute" application (on Windows) sends ICMP messages by default.| The "Identification" header field is unchanged by IP datagram fragmentation.| It is the responsibility of a routing algorithm to find a datagram's path through a network.| When a destination host's IP fragment timer expires, it drops all accumulated fragments corresponding to that timer.| Network address translation has ameliorated the IP address shortage problem.| The transport-layer header is encapsulated in the first fragmented IP datagram.| When a destination host's IP fragment timer expires, it drops all accumulated fragments corresponding to that timer.| The path MTU is the smallest MTU on a path from sender to receiver.| It is the responsibility of a routing algorithm to determine a datagram's next hop information.| The IPv6 address size is 128 bits.| In IPv6, there is no datagram fragmentation performed in the network core.| The IPv6 header does not have a checksum.| The IPv6 header does not have a checksum.| 1234:aac:a03::abcd or 1234::a03:abcd is a valid preferred-format IPv6 address.| A MAC address was originally designed to be permanent and unique.| Given the following received byte on an odd-parity machine, there is definitely at least one error. 01001101| It is fairly easy to detect collisions in wired networks.| Bus Ethernet uses a random access scheme.| IPv6 datagrams cannot be converted to IPv4 datagrams without losing any information.| There are reserved MAC addresses unusable for devices.| For a 10Mbps link, 1000 bit times is 0.1
False: UDP has congestion control mechanism| TCP implements network fairness directly. UDP implements network fairness.| UDP uses an additive-increase multiplicative-decrease (AIMD) system to manage flows.| The Internet Protocol implements congestion control.| The Internet Protocol (IP) implements timing controls. The Internet Protocol (IP) implements flow control.| It is the responsibility of a routing algorithm to determine the cost of an output link.| The path MTU is the largest MTU on a path from sender to receiver. IP datagrams fragments cannot be fragmented again.| If an IP datagram is fragmented into 1000-byte fragments, and later encounters a link with an 800-byte MTU, a special procedure (other than standard IP fragmentation) must be used.| It is the responsibility of a routing algorithm to correlate MAC addresses with IP addresses.| The transport-layer header is encapsulated in every IP datagram fragment.| Network address translation is strictly a Layer-3 protocol.| Network address translation alters IP to add new IP addresses.| The "traceroute" application (on Windows) sends UDP messages by default.| It is the responsibility of a routing algorithm to forward packets to the appropriate output link.| When encountering an IPv4-only router, an IPv6 datagram is dropped.| 1234::a03::abcd is a valid preferred-format IPv6 address.| A MAC address is permanent and unique.| When sending a message to all devices on a link, you would send it to the broadcast MAC address: 00-00-00-00-00-00| The IPv6 address size is 120 bits.| A switch is a network-layer device.| Star Ethernet uses the same multiple access control as Bus Ethernet.| It is fairly easy to detect collisions in wireless networks.| The default multiple access scheme of 802.11g is RTS/CTS.
Next: Given a router with 5 input ports + 5 output ports. If the switching fabric is 5 times as fast as the input/output line speed, queueing **can** occur at an input or output port.
In a datagram network, the responsibilities of the network layer include: *Packet routing, host-to-host communication, packet forwarding*
What can cause packet queueing at a router's output port? Slow outbound link transmission rate. Multiple data flows requiring the same outbound link.
cause queueing at a router's input ports: Output port contention, Slow outbound link transmission rate, Head of Line blocking.
In a subnet, the reserved addresses are the **subnet address** (with a lowest subnet IP address) and the **broadcast address** (with a highest subnet IP address).
A network with a connectionless network layer is called a **datagram network**.

In addition to a "default" entry, routing tables in an internet store **the "first hop" in a path to each of the networks known to the router**.| Subnet has 2 reserved IP addresses. When a host in a network needs to obtain a valid IP address for itself, it broadcasts a "discover" message that can be handled by a **Dynamic Host Configuration Protocol (DHCP)** server, which will "offer" an IP address within the correct domain.
A group of hosts sharing a common address prefix, behind a router, is called a/an **subnet**. The process of moving a datagram from a router's input port to output port is handled by **switching fabric**.| The largest amount of data, in bytes, which can be accomodated by a particilar network, link, or physical-layer is called the **Maximum Transmission Unit (MTU).|** The largest amount of data, in bytes, which can be accommodated throughout a datagram's route from sender to receiver is called the Path Maximum Transmission Unit (Path MTU)| For a TCP/IP datagram coming into a home network through a NAPT device, which of the following header fields (IP and/or TCP) are altered? Header Checksum, Destination IP address & port| The "time to live" field in a modern IPv4 datagram header specifies: the number of remaining hops before the datagram is dropped.
If hosting a server inside a NATed network, how do clients outside the NAPT router connect to the server? *Using Universal Plug and Play (UPnP), thorugh a connection relay service, By using the NAPT devices IP address, and a port number pre-configured to correspond to the server.*| ICMP can carry messages from: Destination Host to Source Host, Source Host to Destination Host, Router to Router, Router to Sender Host
In network graph terminology, **weights** represent costs. In network graph terminology, **nodes** represent routers. In network graph terminology, a **shortest path** from A to B is the set of edges to traverse to reach B from A for the lowest total cost.
In network graph terminology, **edges** represent direct connections between routers.
Given an internet represented as a weighted undirected graph, the shortest path between node *X* and node *Y* is the path that: **has the smallest sum of edge weights.**
Re-assembly of fragmented IP datagrams is handled by: the destination host
In IPv6, datagram fragmentation is **handled at the network edge**.
Select all features explicit in IPv6 which are not explicitly available in IPv4: extension headers, Explicit Payload Length, 128-bit addresses, flow labelling
A multiple access scheme which listens to the channel to make sure it is empty, prior to transmitting: carrier sense protocol | A network with a **bus** topology must terminate the endpoints, but in with a **ring** topology they are connected so there is no endpoint.
A multiple access scheme which divides the usable medium into "chunks" and allows each device sole acces to some number of "chunks": channel partitioning protocol
Which are functions of the Ethernet preamble? Circuit wake-up, Clock synchronization, start signal | The method by which a MAC protocol coordinates access to a broadcast medium to prevent and/or reduce collisions is most commonly called **multiple access**.
Channel Partitioning schemes: TDMA, FDMA, WDMA | Random Access schemes: CSMA, ALOHA| "Taking Turns" schemes:Polling Multiple Access, Token Ring Multiple Access
In a CSMA/CD system, when a collision is detected, **the sender will cut off transmission and wait some time before retransmitting. |** A "collision" is best described as: **when a node receives two or more frames at the same time. |** Collision Detection (CD), Carrier Sense Multi-Access (CSMA), Exponential back-off/retry for collision resolution
Which of the following are used in a wired Ethernet network? (above line)
protocol designed control access to a medium is most commonly:multiple access protocol
On the sending or receiving host, most of the protocol tasks "below" the application layer of the protocol stack (data encapsulation, IP addressing) are handled by
the network interface controller (NIC) | The transition from IPv4 to IPv6 requires that **IPv4 routers still in use must "tunnel" IPv6 datagrams, by fragmenting/encapsulating them in IPv4 datagrams**. | To retrieve an adjacent node's MAC address, **ARP** is used.
A multiple access scheme which uses a master node to poll each slave node, and control who has 'permission' to transmit at any given time is called: "taking turns" protocol
The link-layer device at the center of an ethernet star is a **switch**.
A link-layer link between only two adjacent nodes is called a/an **point to point** link.
The data-link layer provides logical communications between **adjacent nodes**
Most modern Ethernet LANs use a **star** topology.
Ethernet provides: error detection via CRC check | In an Ethernet network, after 8 collisions, the range of wait times will be: [0, 1, ... 255] * 512 bit times
When a mobile unit moves from a home or foreign agent to another (foreign) agent, the new agent must assign: **a new "care-of" address** to the mobile unit. Also:In direct routing, after the initial contact with the home network, the correspondent sends packets here
In one type of wireless network, hosts communicate through a central "base station" access point, which is typically connected to a wired network. This communication model is called an **infrastructure network.**
An organization typically implements its firewall security by using: packet filtering
S represents a <u>source host</u> and *D* represents a <u>destination host</u>. Which of the following is the most typical use of public key encryption, when *S* sends an encrypted message to *D*? Answers: *S* encrypts a msg using *D*'s public key, and *D* decrypts the msg using *D*'s private key. | *S* encrypts a sig using *S*'s private key, and *D* decrypts the sig using *S*'s public key.
S represents a <u>source host</u> and *D* represents a <u>destination host</u>. Which of the following is the most typical use of public key encryption, when *S* sends an authenticated (digitally signed) message to *D*? (see above) | In a link between Host A, and Host B, we have three intermediary routers: Host A ----- Router Snucky ------ Router Jumpy ------ Router Po ------- Host B. Answer: Host A's first hop router is Router **Snucky** . | In a prefix-matching network, a routing table stores: IP Prefixes, Next-Hop link information
<u>Technical Problems</u>
HostA has established a TCP connection with HostB in a remote network. HostA is sending packets to HostB. Assume we have configured TCP, somehow, to ACK every segment (no ACKing every other segment). Assume that the timeout is the same for all packets. HostB's "window size" is 20000 bytes. HostB has already received and acknowledged everything sent by HostA's application up to and including byte #1630. HostA now sends segments of the same application data stream in order: P: 173 bytes Q: 408 bytes R: 251 bytes What is the sequence number on segment Q? Answer: **1804.0**
<u>Acknowledgement Number</u> - The AWK would be one higher because it's asking for the first byte of Q not the last byte of P.
CongWin Reno Sizes from 4 MSS: 1, 2, 4, 5, 6, 1, 3, 2
CongWin Reno Sizes from 8 MSS: 1, 2, 4, 8, 9, 10, 8, 5, 9
Given a nodal delay of 83.4ms when there is no traffic on the network (i.e. usage = 0%), what is the effective delay when network usage = 78.5% ? **D = 83.4/(1-0.785) = *<u>387.9</u>***
When utilization is 1 (100% of capacity), congestion collapse is guaranteed.
Given a 4 Gbps link with TCP applications A (33), B (2), and C (16). According to TCP "fairness", during times when all connections are transmitting, how much bandwidth should Application C have? [16/(33+2+16)]*(4*1000) = ***1254.9 Mbps***
Imagine a mythical set of protocols with the following details. Maximum Link-Layer data frame: 1323 bytes| Network-Layer header size: 8 bytes| Transport-Layer header size: 17 bytes| What is the size, in bytes, of the MSS? 1323-8-17 = ***1298 bytes***
A host starts a TCP transmission with an EstimatedRTT of 17.9ms (from the "handshake"). The host then sends 3 packets and records the RTT for each:
SampleRTT1 = 48.1 ms, SampleRTT2 = 15.7 ms, SampleRTT3 = 48.7 ms
Using an exponential weighted moving average with a weight of 0.4 given to the most recent sample, what is the EstimatedRTT for packet #4? 0.6 * 17.9 + 0.4 * 48.1 = *29.98*| 0.6 * 29.98 + 0.4 * 15.7 = *24.268*| 0.6 * 24.268 + 0.4 * 48.7 = *34.0408*
Suppose that a 2200-byte datagram (identification #40) must transit a network which has a 660-byte MTU. Assume the minimum IP and TCP header sizes, i.e., the IP header and the TCP header are both 20 bytes. 1) How many fragments are created? 4
2) How many bytes of application data are carried in the first fragment? 620 bytes
3) How many bytes of application data are carried in the second fragment? 640 bytes
4) How many bytes of application data are carried in the last fragment? 260
5) What is the identification number of the second fragment? #40
6) What is the fragment offset in the last fragment? 240
For the following binary IP address, give the dotted-decimal representation:
11011110 01110011 01100110 01100110 = **222.115.102.102**
What is the longest-common-prefix for the following address range?
10011110 10111001 10011100 10000000 -- 10011110 10111001 10011101 00000000
Answer: **10011110 10111001 1001110**
<u>Sending a message from Host A -> Host D Diagram</u>
1) A finds that D belongs to a different subnet by checking *D's IP address*. 2) A looks up *RouterA's NIC#1 IP address* in its routing table. 3) A uses ARP to get *RouterA's NIC#1 MAC address*. 4) A creates frame with *RouterA's NIC#1 MAC address* as destination. Frame contains IP datagram with *D's IP address* as destination. 5) A's NIC sends frame and RouterA's NIC receives it. 6) RouterA removes IP datagram from frame, learns that its destination is *D's IP address*. 7) RouterA uses ARP to get *D's MAC address*. 8) RouterA creates frame with *D's MAC Address* as destination. Frame contains IP datagram with *D's IP address.* 9) RouterA's NIC sends frame and D's NIC receives it.
Convert the following IPv4 address to its corresponding IPv6-mapped address, with proper formatting. 192.123.33.1 -> ::ffff:c07b:2101

A	B	C	D	Data	A

Diagram of typical Ethernet hardware frame, what is proper portion of the data encapsulation? A = hardware framing character(s), B = hardware frame header(s), C = IP header(s), D = TCP/UDP header(s) | **<u>For the IPv4 CIDR address 153.10.22.56 /22</u>**
Netmask: 255.255.252.0; Network address: 153.10.20.0; Host Mask: 0.0.3.255; Broadcast Address: 153.10.23.255; Number of possible hosts: 1022; Host Number: 568
Steps for new host "Jetpack" joining a network with a DHCP-enabled server "Rhino". 1) *Jetpack* send *DHCP Discover* to *IP broadcast address* 2) *Rhino* sends *DHCP Offer* to *IP broadcast address* 3) *Jetpack* sends *DHCP Request* to *IP broadcast address* 4) *Rhino* sends *DHCP Acknowledgement* to *IP broadcast address*
Q: Assume a TCP sender is continuously sending 1185-byte segment. If a TCP receiver advertises a window size of 7236 bytes, and with a link transmission rate 29 Mbps an end-to-end propagation delay of 13.2 ms, what is the utilization? **7236/1185 = 6 packets.**
L/R = 1185 * 8) **/** (29*10^6) = .000326897 = .326897ms
L/R * 6 packets = 1.961382ms to send 6 packets
Find RTT = propagation delay x 2 = 13.2x 2 = 26.4 ms
Total Time before sending packet 7: 26.4 + **0.326897** = 26.726897
% of time actually transmitting: (L/R*number of packets) / total time = 1.961382 / 26.726897 = 0.073386 = 7.34%

Character in data	Characters sent
soh	esc x
eot	esc y
esc	esc z

Character	Hex code
soh	01h
eot	04h
esc	1Bh
'x'	78h
'y'	79h
'z'	7Ah

DATA: **78h 79h 01h 04h** | Note: soh and eot are the framing characters.
If byte stuffing is used to transmit Data, what is the byte sequence of the frame (including framing characters)? **Answer: 01h 78h 79h 1Bh 78h 1Bh 79h 04h**