

the one's complement of the following two numbers: 10010111 + 10010000 => 00100111

For demultiplexing, a TCP socket is identified by: <src addr, src port, dest addr, dest port>

The UDP protocol proves error detection. True

It is acceptable to create two TCP connections on the same server/port doublet for the same client/port doublet. False

Server X gets IP + port of Client A and Client A gets IP + port of Server X.

TCP header size: Max -> 60 bytes Min -> 20 bytes

TCP header fields: Header Length/Data Offset, Checksum, Destination Port, Sequence Number, Source Port, Window Size

The TCP countdown timer is used to implement reliable data transmission.

Pipelining is intended primarily to increase network utilization.

Event: Arrival of segment that partially or completely fills in gap in received data.

TCP Receiver Action: Immediately send ACK, provided that segment starts at the lower end of the gap.

A Go-back-N type retransmission protocol will retransmit all un-ACK'd segments upon a countdown timer interrupt.

UDP header size: Max -> 8 bytes Min -> 8 bytes

UDP header fields: Length, Source Port, Checksum, Destination Port

The TCP sequence numbers are used to implement reliable data transmission.

Flow control is intended primarily to keep a TCP sender from overwhelming a receiver's buffer.

HostB has already received and acknowledged everything sent by HostA's application up to and including byte #3642. HostA now sends segments of the same application data stream in order: P-287 bytes, Q-427 bytes, R-493 bytes. What is the sequence number on segment Q?  $3642 + 1 + 287 = 3930$

3930 would also be the acknowledgment number for P.

Assume a TCP sender is continuously sending 1005 byte segment. If a TCP receiver advertises a window size of 5729 bytes, and with a link transmission rate of 32 Mbps, an end-to-end propagation delay of 32 ms, what is the utilization?  $M = \text{rnd\_dn}(5729/1005) \text{ U} = \lfloor M \cdot L^8 R / (L^8 R + 2^{\text{prop\_delay}}) \rfloor \cdot 1.9\text{ms}$

The UDP protocol provides reliable, connectionless service. False

If I were going to implement a lossy VoIP connection, I would use UDP protocol.

A Selective Repeat type retransmission protocol will retransmit one segment at a time upon a countdown timer interrupt.

In a Cumulative acknowledgement scheme, a received ACK indicates all segments prior to the ACK'd segment were received.

The TCP three-way handshake is used to implement a connection.

Event: Arrival of in-order segments with expected sequence number. One other in-order segment waiting for ACK transmission. TCP Receiver Action: Immediately send single cumulative ACK, ACKing both in-order segments.

TCP Connection Initialization: Client sends segment with SYN set to 1. Server sends segment with SYN set to 1 and ACK set to the Client's SeqNum + 1. Client sends segment with SYN set to 0 and ACK set to the Server's SeqNum + 1.

The TCP sliding window is used to implement flow control.

Retransmitting a missing segment before the segment's countdown timer expires is called fast retransmission.

DevRTTn = (1 - B) \* DevRTTn-1 + B \* [SampleRTTnew - Estimated RTTn-1]

UDP uses an additive-increase/multiplicative-decrease (AIMD) system to manage flows. False

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.

In host-inferred congestion control, congestion is detected based on delayed and/or dropped packets.

TCP implements network fairness directly. False

What are some causes of network congestion? Dropped TCP Packets, Parallel TCP Connections, High Utilization, Reliable Data Transfer Schemes, Typical Internet Usage.

UDP has a congestion control mechanism. False

Select the appropriate new CongWin sizes for the following TCP Reno congestion scenario. Assume ssthresh is initially set to 4 MSS ->

1) Connection Established with new server host: CongWin = 1 MSS

2) ACK(s) received from first segment set: CongWin = 2 MSS

3) ACK(s) received from next segment set: CongWin = 4 MSS

4) ACK(s) received from next segment set: CongWin = 5 MSS

5) ACK(s) received from next segment set: CongWin = 6 MSS

6) Timeout occurs: CongWin = 1 MM. ssthresh = 3 MSS

7) ACK(s) received from next segment set: CongWin = 2 MSS

Given an effective delay of 90ms when network usage is 79%, what is the effective delay when network usage = 15%

Effective Delay =  $(90\text{ms}) * ((1 - 0.79) / ((1 - 0.15)))$

EstimatedRTTn =  $((1 - A) * \text{EstimatedRTTn-1} + A * \text{SampleRTTnew})$

A host starts a TCP transmission with an EstimatedRTT of 47.8ms (from the handshake). The host then sends 3 packets and records the RTT for each: SampleRTT1 = 25.7 ms, SampleRTT2 = 45.1 ms, SampleRTT3 = 26.8 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?

EstimatedRTT2 =  $((1 - 0.4) * 47.8 + 0.4 * 25.7)$

EstimatedRTT3 =  $((1 - 0.4) * \text{EstimatedRTT2} + 0.4 * 45.1)$

EstimatedRTT4 =  $((1 - 0.4) * \text{EstimatedRTT3} + 0.4 * 26.8) \rightarrow 35.6\text{ ms}$

Imagine a mythical set of protocols with the following details ->

Maximum Link-Layer data frame: 1431 bytes

Network-Layer header size: 19 bytes

Transport-Layer header size: 29 bytes

What is the size, in bytes, of the MSS? ->  $1431 - 19 - 29 = 1383$

Given a 4 Gbps link with TCP applications A, B, and C. Application A has 47 TCP connections to a remote web server. Application B has 4 TCP connections to a mail server. Application C has 3 TCP connections to a remote web server.

According to TCP "fairness", during times when all connections are transmitting, how much bandwidth should Application C have?  $(3/54) * 4\text{Gbps} = 222.2\text{ Mbps}$

TCP has a congestion control mechanism. True

In a datagram network, the responsibilities of the network layer include: host-to-host communication, packet forwarding, packet routing.

Which of the following are benefits of a datagram network? Faster delivery, Less overhead than a VC network.

The IP header may be 21 bytes long. False

The process of moving a datagram from a router's input port to output port is handled by the switching fabric.

What can cause packet queuing at a router's output port? Slow outbound link transmission rate, Multiple data flows requiring the same outbound link.

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.

The IP implements data reliability services. False

The IP implements timing controls. False

A router's routing table is output by a routing algorithm.

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.

The IP implements flow control. False

What can cause queuing at a router's input ports? Head of line blocking, slow outbound link transmission rate, output port contention.

The process of moving a datagram from a router's input port to output port is called forwarding.

Given a router with 5 input ports and 5 output ports. If the switching fabric is 5 times as fast as the input/output line speed, queuing can occur at both an input port and output port.

Upon encountering a router with the following routing table:

Prefix Match	Port
10011110 00011110 10001111	0
10011110 00011110 10001111 000	1
10011110 00011110 10001111 01	2
10011110 00011110 10001111 0001	3
Default	4

A datagram with the destination IP address 158.30.142.90 would be routed to Port 4.

A datagram with the destination IP address 158.30.143.10 would be routed to Port 1.

A datagram with the destination IP address 158.30.142.30 would be routed to Port 3.

For the IPv4 CIDR address 153.10.22.56/22 what is the Netmask: 22 ones, the rest zeros

11111111 11111111 11111100 00000000 -> to dotted-decimal -> 255.255.252.0

Network Address: turn 153.10.22.56 to binary -> 10011001 00001010 00010110 00111000

11111111 11111111 11111100 00000000 AND 10011001 00001010 00010110 00000000 -> to dotted-decimal -> 153.10.20.0

Host Mask: 22 zeros and the rest ones 00000000 00000000 00000011 11111111 -> to dotted-decimal -> 0.0.3.255

Broadcast Address: turn 153.10.22.56 to binary -> 10011001 00001010 00010110 00111000

00000000 00000000 00000011 11111111 OR 10011001 00001010 00010110 00111000 -> to dotted-decimal -> 153.10.23.255

Number of possible hosts:  $2^{(32-22)} - 2 = 1022$

Host Number: 10011001 00001010 00010110 00111000

00000000 00000000 00000011 11111111 AND 00000000 00000000 00000010 00111000 -> to decimal -> 568

Put the following steps in the correct order for new host "Jetpack" joining a network with a DHCP-enabled server "Rhino".

Jetpack sends DHCP Discover to MAC broadcast address.

Rhino sends DHCP Offer to Jetpack's MAC address.

Jetpack sends DHCP Request to MAC broadcast address.

Rhino sends DHCP Acknowledgement to Jetpack's MAC address.

UDP implements network fairness. False

The process of determining a path through the internet is handled by the routing algorithm.

Routing would be more complicated if we used hardware addresses as network addresses. True

A network with a connectionless network layer is called a datagram network.

The network layer manages communications from host to host.

The transport layer manages communications from process to process.

The data-link layer provides logical communications between adjacent node and adjacent node.

Which of the following are benefits of a virtual circuit network? Connection states are preserved, Guaranteed bandwidth, Guaranteed timing.

A network with a connection-oriented network layer is called a virtual circuit network.

A group of hosts sharing a common address prefix, behind a router, is called a/an subnet.

In a subnet, there are 2 reserved IP address.

It is the responsibility of a routing algorithm to correlate MAC addresses with IP addresses. False

When a destination host's IP fragment timer expires, it drops all accumulated fragments corresponding to that timer. True

If hosting a server inside a NATed network, how do clients outside the NATP router connect to the server? Using Universal Plug and Play (UPnP). By using the NATP devices IP address, and a port number pre-configured to correspond to the server. Through a connection relay service.

Network address translation is strictly a Layer-3 protocol. False

The path MTU is the largest MTU on a path from sender to receiver. False

IP datagrams fragments cannot be fragmented again. False

It is the responsibility of a routing algorithm to find a datagram's path through a network. True

In a fragmented IP datagram, the "offset" IP header field value is equal to the number of bytes of fragmented data preceding this fragment. False

The path MTU is the smallest MTU on a path from sender to receiver. True

For a TCP/IP datagram coming into a home network through a NATP device, which of the following header fields (IP and/or TCP) are altered? Destination IP address, Destination port, Header checksum.

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)

The "ping" application (on Windows) uses ICMP echo request/reply. True

The IP header is encapsulated in IP datagram fragments. False

The "Identification" header field is unchanged by IP datagram fragmentation. True

The transport-layer header is encapsulated in every IP datagram fragment. False

Suppose that a 1600-byte datagram (identification #20) must transit a network which has a 740-byte MTU. Assume the minimum IP and TCP header sizes, i.e., the IP header is 20 bytes and the TCP header is 20 bytes.

1) How many fragments are created?  $(1600 - 20) / (740 - 20) = 3$

2) How many bytes of application data are carried in the first fragment?  $740 - 20 = 720$

3) How many bytes of application data are carried in the second fragment?  $740 - 20 = 720$

4) How many bytes of application data are carried in the last fragment?  $1600 - 20 - 720 - 720 = 140$

5) What is the identification number of the second fragment? 20

6) What is the fragment offset in the last fragment?  $720 / 8 = 90 \quad 0 \rightarrow 90 \rightarrow 180$

Using the version of Dijkstra's Algorithm discussed in the lectures, and the network configuration in the graph, to calculate the shortest path from node H to node B.

What is the 3rd node to be eliminated from the set  $S = \{A, B, C, D, E, F, G\}$ ?

What is the full shortest path from node H to node B? H-G-E-A-C-B

What is the cost of the shortest path from node H to node B? 8

Fill in the complete routing table for node H, as it would be calculated by Dijkstra's algorithm and stored inside router H.

Destination: A B C D E F G

First Hop: G G G D G G G

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).

Where do network-layer protocols run? PCs, Laptops, Routers, Mobile devices.

The "traceroute" application (on Windows) sends UDP messages by default. False

The "traceroute" application (on Windows) sends ICMP messages by default. True

The largest amount of data, in bytes, which can be accommodated by a particular network, link, or physical-layer is called the Maximum Transmission Unit (MTU).

In a prefix-matching network, a routing table stores: Next-Hop link information and IP Prefixes.

NAPT devices translate IP address and port numbers. True

Network address translation alters IP to add new IP addresses. False

For a TCP/IP datagram leaving a home network through a NATP device, which of the following header fields (IP and/or TCP) are altered? Source IP address, Source Port, Header Checksum

The transport-layer header is encapsulated in the first fragmented IP datagram. True

Network address translation has ameliorated the IP address shortage problem. True

When a destination host's IP fragment timer expires, it drops all accumulated fragments corresponding to that timer. True

It is the responsibility of a routing algorithm to determine a datagram's next hop information. True

The IP header may be 28 bytes long. True

It is the responsibility of a routing algorithm to forward packets to the appropriate output link. False

Re-assembly of fragmented IP datagrams is handled by the destination host.

The "time to live" field in a modern IPv4 datagram header specifies the number of remaining hops before the datagram is dropped.

The "Hop Limit" IPv6 header field indicates how many remaining hops to the destination. False

IPv6 datagrams cannot be converted to IPv4 datagrams without losing any information. True

The IPv6 header does not have a checksum. True

Select all features explicit in IPv6 which are not explicitly available in IPv4. Flow labeling, 128-bit addresses, Extension Headers, Explicit Payload Length

In IPv6, there is no datagram fragmentation performed in the network core. True

