1. Layering partitions related communications functions into groups that are manageable. It simplifies design, implementation, and testing by partitioning overall communications process into parts. Protocol in each layer can be designed separately from those in other layers. Layering provides flexibility for modifying and evolving protocols and services without having to change layers below.

Generally, we follow layering in our design, but sometimes we want to do cross-layer design to have better performance. Note that this is only directed to certain small regions not the whole computer network. Generally, layering is good.

1. C = W\_c log\_2(1+SNR) bps

W\_c is channel bandwidth, i.e. range of frequencies passed by channel.

SNR is Signal-to-Noise Ratio.

Since the transmission rate must be smaller than C, a channel coding scheme tries to maximize the channel bandwidth and the Signal-to-Noise Ratio in order to improve transmission.

1. a) Yes. Because the communication rate is a few Mbps and a MAC data frame is usually about 1500 bytes, then the efficiency is acceptable to use Stop-and-Wait.

b) No. Because the communication rate is tens Gbps and Stop-and-Wait ARQ does not work well for very high speeds or long propagation delays. If Bandwidth-Delay Product of the system is high, don't use Stop-and-Wait.

c) Yes. Selective Repeat ARQ is not sensitive to large Bandwidth-Delay Product. Although it still is sensitive to error rate, it can be used in (b).

1. a) In CSMA, collisions result in wastage of X seconds spent transmitting an entire frame. CSMA-CD reduces wastage of time to detect collision and abort transmission. For small a(where a=t\_{prop}/X), CSMA-DC has large throughput, which suits the case of Ethernet (a is small).

b) No. In CSMA-CD, the collision is detected by observing the energy change. However, in wireless LAN, since it is wireless, the energy change is too weak to be detected when there is a collision. The signal sending out by the terminal itself is much stronger than the signal coming in, so it cannot detect others are sending. But if we use some high resolution Analog-to-Digital Converter or add a block of self interference cancellation to cancel out the interference of self-sent signals， then CD can be used.

1. X
2. Because physical carrier sensing analyzes all detected frames and monitors relative signal strength from other sources. We want to know if channel is still idle after DIFS period, only then can we transmit. Physical and virtual carrier sensing are both performed to doublely guarantee that the channel is idle. If either one senses busy, then the channel is busy.
3. X
4. We can impose different interframe times. For users who need higher priorities, give them high priority traffic that senses channel for time t1. For users with low priorities, they need to sense channel for time t2(>t1). Thus, higher priority users seizes channel first and therefore have access earlier than low priority users.
5. Carry the routing path at first and when arriving at each hop, discard the corresponding already-used route information. This way, the overhead is reduces.

Or, store tables at each hop and when packet arrives at each hop, look up the table for next hop.

1. A Byte-Stream protocol transfers a contiguous stream of bytes across the network, with no indication of boundaries. It groups bytes into segments and transmits segments as convenient. TCP should be reliable and connection-oriented. Using byte-stream, receivers can receive packet with order and also control the rate at which sender transmits data to prevent buffer overflow and packet loss. Meanwhile, UDP is best-effort and connectionless and so it can use datagram and don’t care about datagram lost or out-of-order.
2. Receiver controls rate at which sender transmits to prevent buffer overflow through sending back ACKs and informing sender with the Round Trip Time. Similarly, for congestion control, transmitter dynamically adjusts transmission rate according to network congestion as indicated by RTT (round trip time) & ACKs. So basically, ACK and RTT feedbacks give the sender both flow and congestion control.
3. a) Because slow start is a method for source to adapt dynamically to available Bandwidth when it does not know what its fair share of available bandwidth should be. Slow start directs the source to an appropriate sending rate by probing the network.

b) It is slow because there are many low rate sections after each time it reaches the threshold and so the average rate is approximately halved and therefore very slow.

c) It can use fast retransmit and fast recovery. When three duplicate ACKs arrive, retransmit lost segment immediately. Reset congestion threshold to ½ cwnd. Reset cwnd to congestion threshold + 3 to account for the three segments that triggered duplicate ACKs. Remain in congestion avoidance phase. If timeout expires, reset cwnd to 1. In absence of timeouts, cwnd will oscillate around optimal value.