P3

a) A circuit switched network would be more appropriate for this application because TDM can be used efficiently since the application is sending N bits every k time units. Therefore a slot of size N bits can be allocated in every frame, and the frame can be k time units.
b) Congestion control is not needed here because the sum of the application data rates is less than the capacity of each link. Since the applications are sending data a steady rate there will be no wait times for packets to be routed to their destination.

P4

a) 16 - 4 between A and B, 4 between B and C, 4 between C and D, and 4 between D and A.
b) 8 - 4 clockwise and 4 counter clockwise
c) No, if there are 4 existing connections between A and C then B would have no available path to send data to D.

P6

a) $d_{prop} = \frac{m}{s}$
b) $d_{trans} = \frac{L}{R}$
c) end-to-end $= d_{prop} + d_{trans}$
d) Still at Host A
e) In transit to Host B
f) At Host B
g) $(2.5 \cdot 10^8) \cdot \frac{120}{56 \cdot 10^3} = 535.7$km

P7

Since the voice data is 64 kbps, less than the transmission rate of 2 Mbps, there will be no queuing delay. There are no times given for how long it takes to generate the packets or decode the bits back into analog so I'll assume they are instant. Therefore the only delay is the propagation delay of 10 msec, so that is how much time elapses from the time a bit is created on Host A until the bit is decoded on Host B.

P10

Assuming no queuing delays, the total end to end delay of a packet is

$$2d_{proc} + \sum_{i=1}^{3}(\frac{L}{R_i} + \frac{d_i}{s_i})$$

L = 12kb, $\frac{12000}{2 \cdot 10^6} = .006$
$2 \cdot .003 + 3 \cdot .006 + \frac{5 \cdot 10^6}{2.5 \cdot 10^8} + \frac{4 \cdot 10^6}{2.5 \cdot 10^8} + \frac{1 \cdot 10^6}{2.5 \cdot 10^8} = 64$ ms is the end to end delay.

P13

a) If $\frac{NL}{R} > 1$ then the average queuing delay is $\frac{NL}{R} - 1$, otherwise it is negligible or 0.
b) In this case each group of N packets will be arriving when there are no packets currently being transmitted or queued, because it takes $\frac{LN}{R}$ seconds to transmit the N packets. Therefore the queuing delay is the same as part a.

P19

a) I performed trace routes to orange.fr and leboncoin.fr from my laptop using the traceroute program on my Mac. The first 6 links in each are the same, and the remaining 2 are different. I believe the jump from the 6th link to the 7th link is the transatlantic link. The North American side is the same but the European side is different between the 2 traces.
b) I performed a trace route to ebay-kleinanzeigen.de, compared to the French websites the first 6 links are the same. The 7th link is different from both the French routes, and this one has Amsterdam in the domain name so I believe the transatlantic jump is between the 6th and 7th links.
c) I performed trace routes to sina.cn and 10086.cn and the first 5 links are the same, and these are the only links in common between the two. I believe the routes leaves the US after those 5 links. sina.cn jumps to a series of cogentco.com servers and 10086.com jumps to some telia.net servers.

P25

a) $d_{prop} = \frac{20 \cdot 10^6}{2.5 \cdot 10^8} = .08sec$
$R \cdot d_{prop} = 160Kb$ for bandwidth-delay product
b) It takes $d_{prop} = .08sec$ to send a bit from Host A to Host B. $.08sec \cdot 2 \cdot 10^6 Mbps = 160000bits$. 160000 bits is the maximum number of bits that will

be in the link at any given time.
c) The bandwidth-delay product is the maximum number of bits in transit at any given time given the time it takes to traverse the link and the transmission rate of the link (the rate at which bits can be put into the link).
d) $\frac{20 \cdot 10^6}{160000} = 125$ meters. This is longer than a football field.
e) bit-width $= \frac{s}{R}$

P27

a) $d_{prop} = \frac{20 \cdot 10^6}{2.5 \cdot 10^8} = .08sec$
$10^9 \cdot d_{prop} = 80Mb$ for bandwidth-delay product.
b) All 800,000 bits will be in the link at some point in time.
c) bit-width $= \frac{s}{R} = .25$ meters.

P31

a) Without message segmentation 8 Mb is sent over each 2 Mbps link in 4 seconds. There are 3 links so the total time is 12 seconds.
b) The first packet takes $10000/2e6 = 5$ ms to reach the first switch. The second packet will reach the first switch in another 5 ms, so at time 10 ms.
c) The first packet reaches the destination at 15 ms, second at 20 ms, third at 25 ms, so the 800th packet will reach the destination at $799 \cdot 5 + 15 = 4.01$ seconds. This is 3 times as fast as sending the file without message segmentation. This is because message segmentation is almost always using all 3 links while non message segmentation only uses 1 link at a time.
d) Errors are easier to recover from because a transmission error on a link will only affect a packet rather than the entire file.
e) The drawbacks of message segmentation are the processing time required to split the data into packets at the source and the time required to reconstruct the data at the destination.

DARPA Internet Protocols

1) The top 4 goals of the Internet architecture are to develop an effective technique for multiplexed utilization of existing interconnected networks (using packet switching), to continue communication despite loss of networks or gateways, to support multiple types of services, and to accommodate a variety of networks.
2) The goals that led to the success of the Internet are supporting multiple types of communications services and accommodating a variety of networks. By creating a general IP protocol this allowed for a variety of protocols underneath to suit different applications. Also accommodating a variety of networks allowed

the Internet to grow because it did not impose strict constraints on networks.
3) The datagram model was selected because the datagram eliminate the need for connections state in intermediate switching nodes, the datagram provides a basic building block for a variety of of types of services, and it represents a minimum network service assumption.
4) If I were designing the Internet today I would make it easier to use distributed management tools for routing and in the context of multiple administrations. Also I would allow for easier transmission of lost packets.

End to End Arguments in System Design

1) End-to-end check and retry in which the destination generates a checksum on the file and verifies it with the checksum from the source.
Use the communication system to perform checksums and validations on the individual packets of the file.
2) Multicast - this can be done at the application layer or in the networks/communications layer. You would probably acheive some performance gain if it were implemented in the lower network layer but it would result in overhead for every application, many of which don't need multicasting. It should be in the application layer.
Routing - This should be in the network/communications layer. This is fundamental to the Internet and something that every application needs. Efficiently implementing it in the network layer benefits all applications.
Quality of Service - This should be in the application layer. A metric such as this can be measured on the hosts running the application rather than on the lower level switches in the network layer. It would also add unnecessary overhead in the network layer.
Name Resolution - This should be in the network/communications layer. This is a common service that all applications require, and it would be better to implement this very efficiently in the communications layer to benefit all applications.
Web caches - This should be at the application layer. This can speed up client server communication in web applications but would add unwanted overhead to the communications layer that would be undesirable for many applications. It could probably be implemented just as efficiently in the application layer.