Consider the following design problem concerning implementation of virtual-circuit service. If virtual circuits are used internal to the subnet, each data packet must have a 3-byte header and each router must tie up 8 bytes of storage for circuit identification. If datagrams are used internally, 12-byte headers are needed but no router table space is required. Transmission capacity costs 1 cent per 10^6 bytes, per hop. Very fast router memory can be purchased for 1 cent per byte and is depreciated over two years, assuming a 40-hour business week. The statistically average session runs for 1000 sec, in which time 200 packets are transmitted. The mean packet requires four hops. Which implementation is cheaper, and by how much?

1. 15 points

2 years = 2*52*40*3600 / 1000 = 14976 sessions

VC: Storage: 5*8*1cent = 40 cents in 2 years;

transmission: 14976 * 3 * 200 * 4 / 10^6 = 35.9424 cents in 2 years;

Datagram: transmission: 14976 * 12 * 200 * 4 / 10^6 = 143.7696

VC is cheaper in two years by 67.8272 cents.

(c.) Explain how loops in paths can be detected in BGP.

BGP advertisements contain complete paths showing the AS's the path passes through, and so a router can easily identify a loop because an AS will appear two or more times.

Pure ALOHA efficiency

P(success by given node) = P(node transmits) ·

P(no other node transmits in $[t_0-1,t_0]$ ·

P(no other node transmits in $[t_0-1,t_0]$

$= p \cdot (1-p)^{N-1} \cdot (1-p)^{N-1}$

$= p \cdot (1-p)^{2(N-1)}$

… choosing optimum p and then letting $n \longrightarrow \infty$

$= 1/(2e) = .18$

6) Suppose datagrams are limited to 1,500 bytes (including header) between source Host A and destination Host B. Assuming a 20-byte IP header and a 20-byte TCP header, how many datagrams would be required to send an MP3 consisting of 4 million bytes.

Answer: MP3 file size = 4 million bytes. Assume the data is carried in TCP segments, with each TCP segment also having 20 bytes of header. Then each datagram can carry 1500-40=1460 bytes of the MP3 file.

Number of datagrams required = $\left\lceil \frac{4\times10^6}{1460} \right\rceil = 2740$. All but the last datagram will be 1,500 bytes; the last datagram will be 1060+40 = 1100 bytes. Note that here there is not fragmentation – the source host does not create datagrams larger than 1500 bytes, and these datagrams are smaller than the MTUs of the links.

Suppose two nodes, A and B, are attached to opposite ends of an 1200m cable, and that they each have one frame of 1,500 bits (including all headers and preambles) to send to each other. Both nodes attempt to transmit at time t=0. Suppose there are four repeaters between A and B, each inserting a 40-bit delay. Assume the transmission rate is 100 Mbp, and CSMA/CD with backoff intervals of multiples of 512 bits times is used. After the collision, A draws K=0 and B draws K=1 in the exponential backoff protocol. Ignore the jam signal in this case.

a. What is the one-way propagation delay (including repeater delays) between A and B in seconds? Assume the signal propagation speed is $2*10^8$ m/sec.
b. At what time (in seconds) is A's packet completely delivered at B?
c. Now suppose that only A has a packet to send and that the repeaters are replaced with switches. Suppose that each switch has a 20-bit processing delay in addition to a store-and-forward delay. At what time, in seconds, is A's packet delivered at B?

Solution:

a)

$$\frac{\mathbf{1200m}}{\mathbf{2*10^8 m/sec}} + \mathbf{4*}\frac{\mathbf{40bits}}{\mathbf{100*10^6bps}} \text{Type equation here.}$$

$$= (\mathbf{6*10^{-6}+1.6*10^{-6}})\mathbf{sec}$$
$$= \mathbf{7.6\mu sec}$$

b)

First note, the transmission time of a single frame is given by 1500/(100Mbps)=15 micro sec, longer than the propagation delay of a bit.

- At time $t=0$, both A and B transmit.
- At time $t=7.6\mu$sec, both A and B detect a collision, and then abort.
- At time $t=15.2\mu$sec last bit of B's aborted transmission arrives at A.
- At time $t=22.8\mu$sec first bit of A's retransmission frame arrives at B.
- At time $t=22.8\mu\text{sec}+\frac{1500bits}{100\times10^6 bps}=37.8\mu\text{sec}$ A's packet is completely delivered at B.

c) The line is divided into 5 segments by the switches, so the propagation delay between switches or between a switch and a host is given by $\frac{\mathbf{1200m/5}}{\mathbf{2*10^8 m/sec}} = \mathbf{1.2microsec}$.
The delay from Host A to the first switch is given by 15micosec (transmission delay), longer than propagation delay. Thus, the first switch will wait 16.4=15+1.2+0.2 (note, 0.2 is processing delay) till it is ready to send the frame to the second switch. Note that the store-and-forward delay at a switch is 15 microsec. Similarly each of the other 3 switches will wait for 16.4 microsec before ready for transmitting the frame.
The total delay is:
16.4*4 + 15+0.8=81.4 micro sec.

A router is blasting out IP packets whose total length (data plus header) is 1024 bytes. Assuming that packets live for 10 seconds, what is the maximum line speed the router can operate at without danger of cycling through the IP datagram ID number space?

(2) If the bit rate of the line is b, the number of packets/sec that the router can emit is *b*/8192, so the number of seconds it takes to emit a packet is 8192/*b*. To put out 65,536 packets takes (2^29) /*b* sec. Equating this to the maximum packet lifetime, we get (2^29) /*b* =10. Then, *b* is about 53,687,091 bps.

4. (20%) Suppose an ~~IP implementation~~ adheres literally to the following algorithm on receipt of a packet P destined to IP address D:

```
if (Ethernet address for D is in ARP cache)
        send P;
else {
        send out an ARP query for D;
        put P into a queue until the response comes back.
}
```

MAC

(a) If the IP layer receives a burst of packets destined to D, how might this algorithm waste resources unnecessarily? If D isn't in ARP Cache, All packet will send

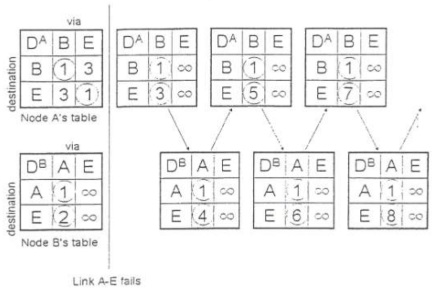
(b) Suppose we simply drop P after sending out a query, when cache lookup fails. How would this behave? and buffer to store

(a) If multiple packets after the first arrive at the IP layer for outbound delivery, but before the first ARP response comes back, then we send out multiple unnecessary ARP packets. Not only do these consume bandwidth, but, because they are broadcast, they interrupt every host and propagate across bridges.

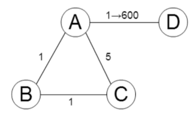
(b) This might, among other things, lead to frequent and excessive packet loss at the beginning of new connections.

4 (10%) Consider a simple network below, in which A and B exchange distance vector routing information. All links have cost 1. Suppose the A-E link fails. Give a sequence of routing table updates that leads to a routing loop between A and B.

4. A necessary and sufficient condition for the routing loop to form is that B reports to A the networks B believes it can currently reach E, after A discovers the problem with the A—E link, but before A has communicated to B that A no longer can reach E.



No, for example



For A, AD = 1 (BD = ∞) CD = 3
After AD→600, A 發現 CD 只要 3
此時 BD = 9, CD = 10, AD = 15
則 BD = 16, CD = 17, AD = 22
則 BD = 23, CD = 24, AD = 29…. Count to infinity

(15%) Suppose a TCP message that contains 1024 bytes of data and 20 bytes of TCP header is passed to IP for delivery across several subnets (i.e. from source host connected to router A to destination host connected to router F) in the network you determined for Prob. 3(a). Assume all IP headers are 20 bytes. The source subnet has a maximum frame size of 1024 bytes including a 14-byte header; the destination subnet has a maximum frame size of 576 bytes including an 8-byte header; the rest of the subnets have a maximum frame size of 512 bytes including a 12-byte header. Please give the fields of *length*, *offset*, and *flags* (*More Fragment* bit only) of the sequence of fragments delivered to the network layer at the destination host. Note that MTU stands for the maximum transmission unit, which is the maximum amount of data that a link-layer frame can carry. The IP datagram format is shown in Fig.3.

Length (Data + IP header)	Offset	Flag
508	0	1
508	61	1
44	122	1
64	125	0

1044 → 20 + 1000 → 20 + 488
20 + 488
20 + 24
→ 20 + 44

(20%, 10 points each) Suppose there are two active nodes A and B on the same 10 Mbps broadcast channel, and the propagation delay between nodes A and B is 312 bit times.

(a) Suppose the two nodes competing for an access to the channel using slotted ALOHA. Assume each node has an infinite number of frames to send and each node attempts to transmit in each slot with probability p. The first slot is numbered slot 1, the second slot is numbered slot 2, and so on. What is probability that node A succeeds for the first time in slot 5? What is the efficiency of this two-node system?

Suppose CSMA/CD and Ethernet frames are used for this channel. Suppose node A begins transmitting a frame and before it finishes, node B starts its transmission. Can node A finish transmitting before it detects that node B has transmitted? Why or why not? The format of an Ethernet frame is

(a) Probability = $[1-p(1-p)]^4 p(1-p)$
Efficiency = $2p(1-p)$ or $Np(1-p)$

(b) No, 311 + 312 = 623 > 576 (Ethernet Min. Packet Size)