

## Chapter 1 Introduction Problems

**2. An alternative to a LAN is simply a big timesharing system with terminals for all users. Give two advantages of a client-server system using a LAN.(M)**

使用局域网模型可以容易地增加节点。

如果局域网只是一条长的电缆，且不会因个别的失效而崩溃(例如采用镜像服务器)的情况下，使用局域网模型会更便宜。

使用局域网可提供更多的计算能力和更好交互式接口。

**3. The performance of a client-server system is influenced by two network factors: the bandwidth of the network (how many bits/sec it can transport) and the latency (how many seconds it takes for the first bit to get from the client to the server). Give an example of a network that exhibits high bandwidth and high latency. Then give an example of one with low bandwidth and low latency.(E)**

横贯大陆的光纤连接可以有很多千兆位/秒带宽，但是由于光速度传送要越过数千公里，时延将也高。

相反，使用 56 kbps 调制解调器呼叫在同一大楼内的计算机则有低带宽和较低的时延。

**4. Besides bandwidth and latency, what other parameter is needed to give a good characterization of the quality of service offered by a network used for digitized voice traffic?(M)**

声音的传输需要相应的固定时间，因此网络时隙数量是很重要的。传输时间可以用标准偏差方式表示。实际上，短延迟但是大变化性比更长的延迟和低变化性更糟。

**6. A client-server system uses a satellite network, with the satellite at a height of 40,000 km. What is the best-case delay in response to a request?(E)**

由于请求和应答都必须通过卫星，因此传输总路径长度为 160,000 千米。在空气和真空中的光速为 300,000 公里/秒，因此最佳的传播延迟为 160,000/300,000 秒，约 533 msec。

**8. A collection of five routers is to be connected in a point-to-point subnet. Between each pair of routers, the designers may put a high-speed line, a**

**medium-speed line, a low-speed line, or no line. If it takes 100 ms of computer time to generate and inspect each topology, how long will it take to inspect all of them?(E)**

将路由器称为 A, B, C, D 和 E。

则有 10 条可能的线路; AB, AC, AD, AE, BC, BD, BE, CD, CE, 和 DE

每条线路有 4 种可能性(3 速度或者不是线路)，拓扑的总数为  $4^{10} = 1,048,576$ 。

检查每个拓扑需要 100 ms，全部检查总共需要 104,857.6 秒，或者稍微超过 29 个小时。

**9. A group of  $2^n - 1$  routers are interconnected in a centralized binary tree, with a router at each tree node. Router  $i$  communicates with router  $j$  by sending a message to the root of the tree. The root then sends the message back down to  $j$ . Derive an approximate expression for the mean number of hops per message for large  $n$ , assuming that all router pairs are equally likely.(H)**

这意味着，从路由器到路由器的路径长度相当于路由器到根的两倍。若在树中，根深度为 1，深度为  $n$ ，从根到第  $n$  层需要  $n-1$  跳，在该层的路由器为 0.50。

从根到  $n-1$  层的路径有 router 的 0.25 和  $n-2$  跳步。因此，路径长度  $l$  为：

$$l = 0.5 * (n-1) + 0.25 * (n-2) + 0.125 * (n-3) + \dots$$

结果化简为  $l = n - 2$ ，平均路由路径为  $2n - 4$ 。

**10. A disadvantage of a broadcast subnet is the capacity wasted when multiple hosts attempt to access the channel at the same time. As a simplistic example, suppose that time is divided into discrete slots, with each of the  $n$  hosts attempting to use the channel with probability  $p$  during each slot. What fraction of the slots are wasted due to collisions?(H)**

区分  $n-2$  事件。事件 1 到  $n$  由主机成功地、没有冲突地使用这条信道的事件组成。这些可能性的事件的概率为  $p(1-p)^{n-1}$ 。事件  $n+1$  是一个空闲的信道，其概率为  $(1-p)^n$ 。事件  $n+2$  是一个冲突。由于事件  $n+2$  互斥，它们可能发生的事件必须统一合计。冲突的可能性等于那些小部分的槽的浪费，只是

$$1 - np(1-p)^{n-1} - (1-p)^n$$

**11. What are two reasons for using layered protocols?(M)**

通过协议分层可以把设计问题划分成较小的易于处理的片段  
分层意味着某一层的协议的改变不会影响高层或低层的协议

**13. What is the principal difference between connectionless communication and connection-oriented communication?(E)**

主要的区别有两条。

其一：面向连接通信分为三个阶段，第一是建立连接，在此阶段，发出一个建立连接的请求。第二阶段，只有在连接成功建立之后，保持连接状态，才能开始数据传输。第三阶段，当数据传输完毕，必须释放连接。而无连接通信没有这么多阶段，它直接进行数据传输。

其二：面向连接的通信具有数据的保序性，而无连接的通信不能保证接收数据的顺序与发送数据的顺序一致。

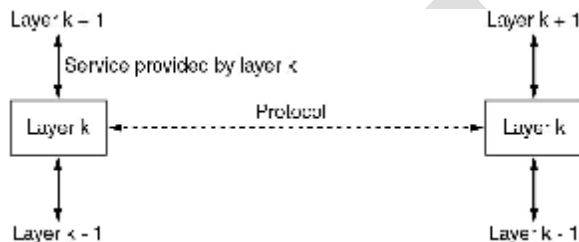
**14. Two networks each provide reliable connection-oriented service. One of them offers a reliable byte stream and the other offers a reliable message stream. Are these identical? If so, why is the distinction made? If not, give an example of how they differ.(E)**

不相同。在报文流中，网络保持对报文边界的跟踪；而在字节流中，网络不做这样的跟踪。例如，一个进程向一条连接写了 1024 字节，稍后又写了另外 1024 字节。那么接收方共读了 2048 字节。对于报文流，接受方将得到两个报文。每个报文 1024 字节。而对于字节流，报文边界不被识别。接收方把全部的 2048 个字节当作一个整体，在此已经体现不出原先有两个报文的事实。

**15. What does "negotiation" mean when discussing network protocols? Give an example.(E)**

协商就是要让双方就在通信期间将使用的某些参数或数值达成一致。最大分组长度就是一个例子。

**16. In Fig. 1-19, a service is shown. Are any other services implicit in this figure? If so, where? If not, why not?(E)**



服务是由 k 层向 k+1 层提供的。

服务必须由下层 k 提供，即，对层 k 的服务是由 k-1 层提供的。

**17. In some networks, the data link layer handles transmission errors by requesting damaged frames to be retransmitted. If the probability of a frame's being**

**damaged is p, what is the mean number of transmissions required to send a frame?**

**Assume that acknowledgements are never lost.(M)**

假设某帧传到第 k 次才传输成功，起初 k-1 次传输皆尝试失败，概率为  $p^{k-1}$ ，第 k 次传输成功，概率为  $(1-p)$ ，则发送一帧成功的平均传输次数为：

$$\sum_{k=1}^{\infty} k P_k = \sum_{k=1}^{\infty} k (1-p) p^{k-1} = \frac{1}{1-p}$$

1. Which of the OSI layers handles each of the following:

a. (a) Dividing the transmitted bit stream into frames.

b. (b) Determining which route through the subnet to use. (E)

把传输的比特流划分为帧——数据链路层

决定使用哪条路径通过子网——网络层。

**19. If the unit exchanged at the data link level is called a frame and the unit exchanged at the network level is called a packet, do frames encapsulate packets or do packets encapsulate frames? Explain your answer. (E)**

帧封装包。当一个包到达数据链路层时，整个数据包，包括包头、数据及全部内容，都用作帧的数据区。或者说，将整个包放进一个信封(帧)里面，(如果能装的话)。

**21. List two ways in which the OSI reference model and the TCP/IP reference model are the same. Now list two ways in which they differ. (M)**

相似点：都是独立的协议栈的概念；层的功能也大体相似。

不同点：OSI 更好的区分了服务、接口和协议的概念，因此比 TCP/IP 具有更好的隐藏性，能够比较容易的进行替换；OSI 是先有的模型的概念，然后再进行协议的实现，而 TCP/IP 是先有协议，然后建立描述该协议的模型；层次数量有差别；TCP/IP 没有会话层和表示层，OSI 不支持网络互连。OSI 在网络层支持无连接和面向连接的通信，而在传输层仅有面向连接的通信，而 TCP/IP 在网络层仅有一种通信模式(无连接)，但在传输层支持两种模式。

**22. What is the main difference between TCP and UDP? (E)**

TCP 是面向连接的，而 UDP 是一种数据报服务。

**25. When a file is transferred between two computers, two acknowledgement strategies are possible. In the first one, the file is chopped up into packets, which are individually acknowledged by the receiver, but the file transfer as a whole is not acknowledged. In the second one, the packets are not acknowledged individually, but the entire file is acknowledged when it arrives. Discuss these two approaches. (E)**

如果网络容易丢失分组，那么对每一个分组逐一进行确认较好，此时仅重传丢失的分组。

如果网络高度可靠，那么在不发差错的情况下，仅在整个文件传送的结尾发送一次确认，从而减少了确认的次数，节省了带宽；不过，即使有单个分组丢失，也需要重传整个文件。

## 26. Why does ATM use small, fixed-length cells? (E)

因为这样可以迅速地经由交换机转发，并且这是在硬件上完成的。这样的设计使得制造可以同时并行处理多个 CELLS 的硬件设备更加容易。另外，它们不会阻碍传输线路很久，更加容易保证提供出高质量的服务。

**28. An image is 1024 x 768 pixels with 3 bytes/pixel. Assume the image is uncompressed. How long does it take to transmit it over a 56-kbps modem channel? Over a 1-Mbps cable modem? Over a 10-Mbps Ethernet? Over 100-Mbps Ethernet?(E)**

该图像大小为  $1024 * 768 * 3 * 8 = 18,874,368$  bits.

传输速率为 56Kbits/sec, 需要  $18,874,368 / 56,000 = 337.042$  sec.

传输速率为 1Mbits/sec, 需要  $18,874,368 / 10^6 = 18.874$  sec.

传输速率为 10Mbits/sec, 需要  $18,874,368 / 10^7 = 1.887$  sec.

传输速率为 100Mbits/sec, 需要  $18,874,368 / 10^8 = 0.189$  sec.

**29. Ethernet and wireless networks have some similarities and some differences. One property of Ethernet is that only one frame at a time can be transmitted on an Ethernet. Does 802.11 share this property with Ethernet? Discuss your answer.(E)**

想象一下隐藏终端的问题。假设一个无线网络里有五台终端，从 A 至 E，使它们每一台都只可以联系到与其相邻的两个邻居之一，那么 A 在与 B 通讯的同时 D 可以与 E 进行通讯。因此无线网络有潜在的并行性，这与以太网上不同的。

**30. Wireless networks are easy to install, which makes them inexpensive since installation costs usually far overshadow equipment costs. Nevertheless, they also have some disadvantages. Name two of them. (E)**

无线网络的缺点：一是安全性，偶然出现在无线网络内的人都能监听到网络上传递的消息；再有就是可靠性，无线网络在传输过程中会出现很多错误；另外，因为许多无线设备需要移动，电池使用寿命不长也是其缺点之一。

## Chapter 2 The Physical Problems

**2. A noiseless 4-kHz channel is sampled every 1 msec. What is the maximum data rate? (E)**

由尼奎斯特定理，无噪声信道最大数据传输率= $2H\log_2 V$  b/s。依题有带宽  $H = 4\text{kHz}$ ，因此最大数据传输率决定于每次采样所产生的比特数 ( $\log_2 V$ )。

如果每次采样产生 16bits，那么数据传输率可达 128kbps；

如果每次采样产生 1024bits，那么可达 8.2Mbps。

**3. Television channels are 6 MHz wide. How many bits/sec can be sent if four-level digital signals are used? Assume a noiseless channel. (E)**

依题有带宽  $H = 6\text{MHz}$ ，每次采样  $\log_2 V = 2\text{bit}$

由尼奎斯特定理，可发送的最大数据传输率为  $2H\log_2 V = 24\text{Mbps}$ 。

**4. If a binary signal is sent over a 3-kHz channel whose signal-to-noise ratio is 20 dB, what is the maximum achievable data rate? (M)**

由香农定理信道比为  $S/N$  的有噪声信道的最大数据传输率 =  $H\log_2(1+S/N)$ 。

依题知带宽  $H = 3\text{kHz}$ ，信噪比为  $10\lg S/N = 20\text{ dB}$ ，知  $S/N = 100$

由于  $\log_2 101 \approx 6.658$ ，该信道的信道容量为  $3\log_2(1+100) = 19.98\text{kbps}$

再根据尼奎斯特定理，发送二进制信号的 3kHz 信道的最大数据传输速率为

$2H\log_2 V = 2 * 3 \log_2 2 = 6\text{kbps}$  综上，可以取得的最大数据传输速率为 6kbps。

**5. What signal-to-noise ratio is needed to put a T1 carrier on a 50-kHz line? (M)**

T1 信号的带宽 =  $1.544 * 10^6\text{Hz}$ ，为发送 T1 信号，由香农定理，最大数据传输率 =  $H\log_2(1+S/N) = 1.544 * 10^6\text{Hz}$ ，依题知带宽  $H = 50\text{ kHz}$ ，解得  $S/N = 2^{31} - 1$

再由尼奎斯特定理  $2H\log_2 V = 2H\log_2 S/N = 93\text{ dB}$

所以，在 50kHz 线路上使用 T1 载波需要 93dB 的信噪比。

**7. How much bandwidth is there in 0.1 micron of spectrum at a wavelength of 1 micron?(M)**

依题知频段为  $0.1\mu\text{m}$ ，波长为  $1\mu\text{m}$

$$\frac{f - \frac{c}{\lambda}}{\frac{df}{d\lambda} - \frac{c}{\lambda^2} d\lambda} = \frac{\frac{df}{d\lambda} - \frac{c}{\lambda^2}}{\Delta f - \frac{c}{\lambda^2} \Delta \lambda}$$

$$c = 3 \times 10^8 \quad \lambda = 10^{-6} \text{ m}$$

$$\Delta \lambda = 0.1 \times 10^{-6} = 10^{-7} \text{ m}$$

$$\Delta f = \frac{3 \times 10^8}{(10^{-6})^2} \times 10^{-7} = 30 \times 10^{12} \text{ Hz} = 30 \text{ THz}$$

因此，在  $0.1\mu\text{m}$  的频段中可以有 30THz。

8. It is desired to send a sequence of computer screen images over an optical fiber. The screen is 480 x 640 pixels, each pixel being 24 bits. There are 60 screen images per second. How much bandwidth is needed, and how many microns of wavelength are needed for this band at 1.30 microns? (M)

传输数据的速率为  $480 \times 640 \times 24 \times 60 \text{bps}$ ，即 442Mbps。

$$\Delta f = 4.42 \times 10^8$$

$$f = \frac{c}{\lambda} \quad \frac{df}{d\lambda} = -\frac{c}{\lambda^2}$$

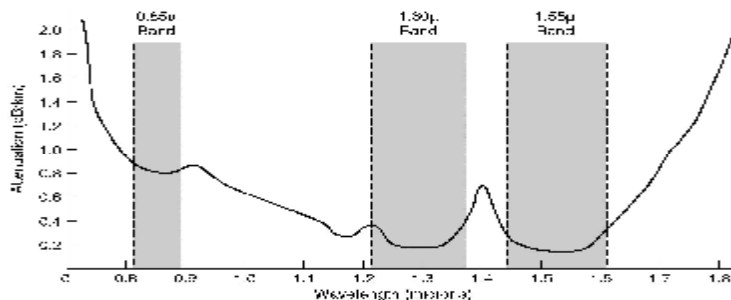
$$|\Delta \lambda| = \frac{\lambda^2 \Delta f}{c} = \frac{(1.3 \times 10^{-6})^2 \times 4.42 \times 10^8}{3 \times 10^8} = 2.5 \times 10^{-12} \text{ m} = 2.5 \times 10^{-6} \mu\text{m}$$

需要 442Mbps 的带宽，对应的波长范围是  $2.5 \times 10^{-6} \mu\text{m}$ 。

9. Is the Nyquist theorem true for optical fiber or only for copper wire? (D)

尼奎斯特定理是一个数学性质，不涉及技术处理。该定理说，如果你有一个函数，它的傅立叶频谱不包含高于  $f$  的正弦和余弦，那么以  $2f$  的频率采样该函数，那么你就可以获取该函数所包含的全部信息。因此尼奎斯特定理适用于所有介质。

10. In Fig. 2-6 the lefthand band is narrower than the others. Why? (E)



由于这 3 个波段的频率范围大体上相等，根据公式  $\Delta f = \frac{c}{\lambda^2} \Delta \lambda$ ， $\lambda$  小的波段  $\Delta \lambda$  也得小，才能保持  $\Delta f$  大约相等。

顺便指出，3 个带宽大致相同的事实是所使用的硅的种类的一个碰巧的特性反映。

11. Radio antennas often work best when the diameter of the antenna is equal to the wavelength of the radio wave. Reasonable antennas range from 1 cm to 5 meters in diameter. What frequency range does this cover? (E)

$$f = \frac{c}{\lambda}$$

当  $\lambda$  为 1cm 时， $f$  为 30GHz。  
当  $\lambda$  为 5m 时， $f$  为 60MHz。

12. Multipath fading is maximized when the two beams arrive 180 degrees out of phase. How much of a path difference is required to maximize the fading for a 50-km-long 1-GHz microwave link? (E)

由公式  $f = \frac{c}{\lambda}$ ，这里光速  $c = 300000 \text{ km/s}$ ，依题  $f = 1 \text{ GHz}$ ，所以微波的波长是 30cm。如果一个波比另一个波多行进 15cm，那么它们到达时将 180 异相。显然，答案与链路长度是 50km 的事实无关。

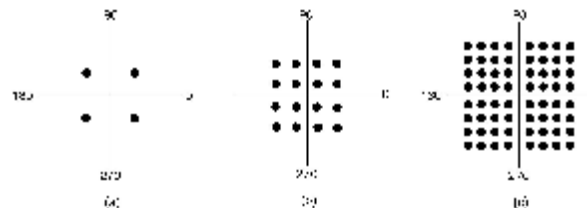
18. A simple telephone system consists of two end offices and a single toll office to which each end office is connected by a 1-MHz full-duplex trunk. The average telephone is used to make four calls per 8-hour workday. The mean call duration is 6 min. Ten percent of the calls are long-distance (i.e., pass through the toll office). What is the maximum number of telephones an end office can support? (Assume 4 kHz per circuit.) (E)

每部电话每小时做 0.5 次通话，每次通话 6 分钟。因此一部电话每小时占用一条电路 3 分钟， $60/3=20$ ，即 20 部电话可共享一条线路。由于只有 10% 的呼叫是长途，所以 200 部电话占用一条完全时间的长途线路。局间干线复用了  $1000000/4000=250$  条线路，每条线路支持 200 部电话，因此，一个端局可以支持的电话部数为  $200 \times 250 = 50000$ 。

21. The cost of a fast microprocessor has dropped to the point where it is now possible to put one in each modem. How does that affect the handling of telephone line errors? (E)

通常在物理层对于在线路上发送的比特不采取任何差错纠正措施。在每个调制解调器中都包括一个 CPU，使得有可能在第一层中包含错误纠正码，从而大大减少第二层所看到的错误率。由调制解调器做的错误处理可以对第二层完全透明。现在许多调制解调器都有内建的错误处理功能。

22. A modem constellation diagram similar to Fig. 2-25 has data points at the following coordinates: (1, 1), (1, -1), (-1, 1), and (-1, -1). How many bps can a modem with these parameters achieve at 1200 baud? (E)



每个波特有 4 个合法值，因此比特率是波特率的两倍。



对应于 1200 波特，数据速率是 2400bps。

23. A modem constellation diagram similar to Fig. 2-25 has data points at (0, 1) and (0, 2). Does the modem use phase modulation or amplitude modulation? (E)

相位总是 0，但使用两个振幅，因此这是直接的幅度调制。

24. In a constellation diagram, all the points lie on a circle centered on the origin. What kind of modulation is being used? (E)

如果所有的点都在同一圆周上，那么它们有着同样的幅度，所以没有使用幅度调制。在星座图中从来就不使用频率调制，所以，这里所采用的编码是纯相位调制。

25. How many frequencies does a full-duplex QAM-64 modem use? (E)

全双工的 QAM-64 使用了两个频率。一个给上行流，一个给下行流。调制机制本身只使用了相位和幅度调制，这里对频率不做调制。

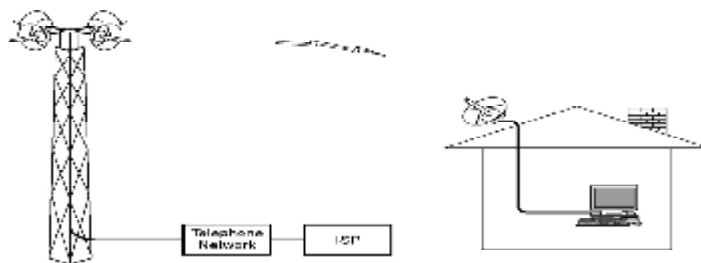
26. An ADSL system using DMT allocates 3/4 of the available data channels to the downstream link. It uses QAM-64 modulation on each channel. What is the capacity of the downstream link? (M)

DMT 指离散的多信道调制。这里总共有 256 条信道，减去 6 条给 POTS 以及再减少 2 条用于控制，余下的 248 条留给数据。依题其中的 3/4 即 186 条信道给下行流。

ADSL 是以 4000 baud/s 进行调制。所以对 QAM-64 (6 bits/baud) 可得每条信道的带宽为 24,000 bps

所以下行流总的带宽为  $24,000 \text{ bps} \times 186 = 4.464 \text{ Mbps}$

27. In the four-sector LMDS example of Fig. 2-30, each sector has its own 36-Mbps channel. According to queueing theory, if the channel is 50% loaded, the queueing time will be equal to the download time. Under these conditions, how long does it take to download a 5-KB Web page? How long does it take to download the page over a 1-Mbps ADSL line? Over a 56-kbps modem? (E)



LMDS 指本地多点分发服务。一个 5-KB 的网页数据量为 40,000 bits，对于 36 Mbps 的信道而言下载时间是 1.1msec，依题平均排队延迟时间也是 1.1msec，则总下载时间是 2.2msec，对 ADSL 而言并没有排队延迟时间，所以 1 Mbps 的下载时间是 40

msec，在 56 kbps 的条件下下载时间是 714 msec。

28. Ten signals, each requiring 4000 Hz, are multiplexed on to a single channel using FDM. How much minimum bandwidth is required for the multiplexed channel? Assume that the guard bands are 400 Hz wide. (E)

对于 10 个 4000 Hz 的信号，我们需要使用 9 个防护频段来避免可能的干扰。

所需要的最小带宽为  $4000 \times 10 + 400 \times 9 = 43,600 \text{ Hz}$

29. Why has the PCM sampling time been set at 125  $\mu\text{sec}$ ? (E)

125 $\mu\text{s}$  的采样时间对应于每秒 8000 次采样。一个典型的电话通道为 4kHz。根据尼奎斯特定理，为获取一个 4kHz 的通道中的全部信息需要每秒 8000 次的采样频率。

30. What is the percent overhead on a T1 carrier; that is, what percent of the 1.544 Mbps are not delivered to the end user? (M)

T1 线路的每一帧中，端点用户使用 193 位中的 168 (7\*24) 位，开销占 25 (=193-168) 位，因此开销比例等于 25/193=13%。

31. Compare the maximum data rate of a noiseless 4-kHz channel using

a. (a) Analog encoding (e.g., QPSK) with 2 bits per sample.

b. (b) The T1 PCM system. (E)

两种情况下均为 8000 次采样/秒。使用 2 进制编码，则对 a 每次采样中发送 2 位数据，对 T1 线路，每次采样发送 7 位数据。所以相对的最大数据传输率为：

(a) 每次采样 2 比特的模拟编码  $2H\log_2 V = 16 \text{ kbps}$

(b) T1 PCM 系统  $2H\log_2 V = 56 \text{ kbps}$

32. If a T1 carrier system slips and loses track of where it is, it tries to resynchronize using the 1st bit in each frame. How many frames will have to be inspected on average to resynchronize with a probability of 0.001 of being wrong? (M)

10 个帧。在数字通道上某些随机比特是 0101010101 模式的概率是 1/1024。察看 10 个帧，若每一帧中的第一位形成比特串 0101010101，则判断同步成功，而误判的概率为 1/1024，小于 0.001。

33. What is the difference, if any, between the demodulator part of a modem and the coder part of a codec? (After all, both convert analog signals to digital ones.) (M)

有区别。编码器接受任意的模拟信号，并从它产生数字信号。而解调器仅仅接受调制的正弦（或余弦）波，产生数字信号。

34. A signal is transmitted digitally over a 4-kHz noiseless channel with one sample every 125  $\mu$ sec. How many bits per second are actually sent for each of these encoding methods?(M)

c. (a) CCITT 2.048 Mbps standard.

d. (b) DPCM with a 4-bit relative signal value.

e. (c) Delta modulation.

a. CCITT 2.048Mbps 标准用 32 个 8 位数据样本组成一个 125 $\mu$ s 的基本帧, 30 个信道用于传信息, 2 个信道用于传控制信号。在每一个 4kHz 信道上发送的数据率就是  $8 \times 8000 = 64\text{kbps}$ 。

b. 差分脉码调制 (DPCM) 是一种压缩传输信息量的方法, 它发送的不是每一次抽样的二进制编码值, 而是两次抽样的差值的二进制编码。现在相对差值是 4 位, 所以对应每个 4kHz 信道实际发送的比特速率为  $4 \times 8000 = 32\text{kbps}$ 。

c. 增量调制的基本思想是: 当抽样时间间隔  $s_t$  很短时, 模拟数据在两次抽样之间的变化很小, 可以选择一个合适的量化值  $v$  作为阶距。把两次抽样的差别近似为不是增加一个  $v$  就是减少一个  $v$ 。这样只需用 1bit 二进制信息就可以表示一次抽样结果, 而不会引入很大误差。因此, 此时对应每个 4kHz 信道实际发送的数据速率为  $1 \times 8000 = 8\text{kHz}$ 。

35. A pure sine wave of amplitude A is encoded using delta modulation, with x samples/sec. An output of +1 corresponds to a signal change of  $+A/8$ , and an output signal of -1 corresponds to a signal change of  $-A/8$ . What is the highest frequency that can be tracked without cumulative error?(E)

在波的 1/4 周期内信号必须从 0 上升到 A。为了能够跟踪信号, 在  $T/4$  的时间内 (假定波的周期是 T) 必须采样 8 次, 即每一个全波采样 32 次, 采样的时间间隔是  $1/x$ , 因此波的全周期必须足够的长, 使得能包含 32 次采样, 即  $T > 32/x$ , 或  $f_{\max} = x/32$ 。

37. In Fig. 2-37, the user data rate for OC-3 is stated to be 148.608 Mbps. Show how this number can be derived from the SONET OC-3 parameters.(H)

SONET		SDH		Data rate (Mbps)	
Electrical	Optical	Optical	Gross	SPE	User
STS-1	OC-1		51.84	50.112	49.536
STS-3	OC-3	SIM-1	155.52	150.336	148.608
STS-9	OC-9	STM-3	466.56	451.008	445.824
STS-12	OC-12	STM-4	622.08	601.344	594.432
STS-18	OC-18	SIM-6	903.12	902.016	891.648
STS-24	OC-24	SIM-8	1244.16	1202.688	1188.864
STS-36	OC-36	STM-12	1866.24	1804.032	1783.296
STS-48	OC-48	STM-16	2488.32	2405.376	2377.728
STS-192	OC-192	STM-64	9953.28	9621.504	9510.912

基本的 SONET (同步光网络) 帧是每 125 $\mu$ s 产生 810 字节。由于 SONET 是同步的, 因此不论是否有实际要发送的数据, 帧都存在。每秒 8000 帧的速率正好符合所有数字电话系统中使用的 PCM 信道的采样率。对于 810 字节的 SONET 帧, 通常用 90 列乘以 9 行的矩形来描述, 每个单元对应一个字节。每秒传送 8000 次, 每次  $8 \times 810 = 6480$  位, 总数据传输率为 51.84Mbps。这就是基本的 SONET 信道, 它被称作同步传输信号 STS-1, 所有的 SONET 干线都是 STS-1 的倍数。每一帧的前 3 列被保留, 用于管理信息系统, 前 3 行包含段开销, 后 6 行包含线路开销。剩下的 87 列包含  $87 \times 9 \times 8 \times 8000 = 50.112\text{Mbps}$  的用户数据。用户数据 (称为同步载荷信封, 即 SPE) 可以从帧内的任一位置开始, 并不限于第 1 行, 第 4 列。线路开销的第一行包含指向 SPE 的第一字节的指针, SPE 的第一列是路径开销。

路径开销不是严格的 SONET 结构, 它在嵌入在载荷信封中。路径开销端到端的流过网络, 因此把它与端到端的运载用户信息的 SPE 相关联是有意义的。然而, 它从可提供给终端的用户数据中的 50.112Mbps 中又减去  $1 \times 9 \times 8 \times 8000 = 0.576\text{Mbps}$ , 使之变成 49.536Mbps。OC-3 相当于 3 个 OC-1 复用在一起, 因此其用户数据传输速率是  $49.536 \times 3 = 148.608\text{Mbps}$ 。

39. What is the essential difference between message switching and packet switching? (E)

报文交换中, 对于数据块的大小没有任何限制。

分组交换则对于数据块的大小有限制, 任何报文超出了这一限制都会被分割成小块的多组。

40. What is the available user bandwidth in an OC-12c connection? (H)

[[当一条线路 (例如 OC-3) 没有被复用, 而是仅传输来自一个源的数据, 则在线路名称后面加一个字母 c (表示 concatenation, 即串联)。因此, OC-3 表示了由 3 条独立的 OC-1 线路构成的一条 155.52Mbps 线路, 而 OC-3c 表示来自于单个源的 155.52Mbps 的数据流。OC-3c 流内的 3 个 OC-1 流被按列交替插入, 首先是流 1 的

第1列, 流2的第1列, 流3的第1列, 随后是流1的第2列, 流2的第2列, 以此类推, 最后形成270列宽9行高的帧。

OC-3c流中的用户实际数据传输速率比OC-3流的速率略高(149.760Mbps和148.608Mbps), 因为路径开销仅在SPE中出现一次, 而不是当使用3条单独OC-1流时出现的3次。换句话说, OC-3c中270列中的260列可用于用户数据, 而在OC-3中仅能使用258列。更高层次的串联帧(如OC-12c)也存在这样的情况。]]

OC-12c帧有 $12 \times 90 = 1080$ 列和9行。其中段开销和线路开销占 $12 \times 3 = 36$ 列, 这样同步载荷信封就有 $1080 - 36 = 1044$ 列。SPE中仅1列用于路径开销, 结果就是1043列用于用户数据。

由于每列9个字节, 因此一个OC-12c帧中用户数据比特数是 $8 \times 9 \times 1043 = 75096$ 。每秒8000帧, 得到用户数据速率 $75096 \times 8000 = 600768000\text{bps}$ , 即600.768Mbps。

所以, 在一条OC-12c连接中可提供的用户带宽是600.768Mbps。

41. Three packet-switching networks each contain  $n$  nodes. The first network has a star topology with a central switch, the second is a (bidirectional) ring, and the third is fully interconnected, with a wire from every node to every other node. What are the best-, average-, and-worst case transmission paths in hops? (E)

三种网络的特性如下:

星型: 最好为2, 最差为2, 平均为2;

环型: 最好为1, 最差为 $n/2$ , 平均为 $n/4$

如果考虑 $n$ 为奇偶数,

则 $n$ 为奇数时, 最坏为 $(n-1)/2$ , 平均为 $(n+1)/4$

$n$ 为偶数时, 最坏为 $n/2$ , 平均为 $n^2/4(n+1)$

全连接: 最好为1, 最差为1, 平均为1。

42. Compare the delay in sending an  $x$ -bit message over a  $k$ -hop path in a circuit-switched network and in a (lightly loaded) packet-switched network. The circuit setup time is  $s$  sec, the propagation delay is  $d$  sec per hop, the packet size is  $p$  bits, and the data rate is  $b$  bps. Under what conditions does the packet network have a lower delay?(M)

对于电路交换,  $t=s$ 时电路建立起来;  $t=s+x/d$ 时报文的最后一位发送完毕;  $t=s+x/d+kb$ 时报文到达目的地。而对于分组交换, 最后一位在 $t=x/b$ 时发送完毕。

为到达最终目的地, 最后一个分组必须被中间的路由器重发 $k-1$ 次, 每次重发花

时间 $p/b$ , 所以总的延迟为 $\frac{x}{b} + (k-1)\frac{p}{b} + kd$

为了使分组交换比电路交换快, 必须:  $\frac{x}{b} + (k-1)\frac{p}{b} + kd < s + \frac{x}{b} + kd$

所以:  $s > (k-1)\frac{p}{b}$

43. Suppose that  $x$  bits of user data are to be transmitted over a  $k$ -hop path in a packet-switched network as a series of packets, each containing  $p$  data bits and  $h$  header bits, with  $x \gg p + h$ . The bit rate of the lines is  $b$  bps and the propagation delay is negligible. What value of  $p$  minimizes the total delay? (M)

所需要的分组总数是 $x/p$ , 因此加上头信息的总数据量为 $(p+h)x/p$ 位。

源端发送这些位需要时间为 $(p+h)x/pb$ ,

中间的路由器重传最后一个分组所花的总时间为 $(k-1)(p+h)/b$ ,

因此我们得到的总的延迟为 $(p+h)\frac{x}{pb} + (p+h)(k-1)\frac{1}{b}$ ,

对该函数求 $p$ 的导数, 得到 $\frac{p-(p+h)x}{p^2} \cdot \frac{x}{b} + \frac{k-1}{b}$ , 令 $\frac{p-(p+h)x}{p^2} \cdot \frac{x}{b} + \frac{k-1}{b} = 0$ 得到

$$\frac{hx}{p^2} = k-1$$

因为 $p > 0$ , 所以 $p = \sqrt{\frac{hx}{k-1}}$ 故 $p = \sqrt{\frac{hx}{k-1}}$ 时能使总的延迟最小。

44. In a typical mobile phone system with hexagonal cells, it is forbidden to reuse a frequency band in an adjacent cell. If 840 frequencies are available, how many can be used in a given cell? (B)

每个蜂窝单元都有6个邻居。假定中间的单元使用频率A, 它的邻居则可以依次使用B, C, B, C, B, C, 也就是说, 只需用三个相互独立的单元就足够了。所以, 每个单元可以使用280种频率。

47. Sometimes when a mobile user crosses the boundary from one cell to another, the current call is abruptly terminated, even though all transmitters and receivers are functioning perfectly. Why? (E)

在邻近的蜂窝单元中频率不能复用。所以当一名用户从一个单元移动到另一个单元时, 必须给他分配一个新的频率。如果当用户移到一个新的单元, 但是当前的新单元中的所有频率都在使用中, 则用户的呼叫必须被终止。

**48. D-AMPS has appreciably worse speech quality than GSM. Is this due to the requirement that D-AMPS be backward compatible with AMPS, whereas GSM had no such constraint? If not, what is the cause? (M)**

It is not caused directly by the need for backward compatibility. The 30 kHz channel was indeed a requirement, but the designers of D-AMPS did not have to stuff three users into it. They could have put two users in each channel, increasing the payload before error correction from  $260 \times 50 = 13$  kbps to  $260 \times 75 = 19.5$  kbps. Thus, the quality loss was an intentional trade-off to put more users per cell and thus get away with bigger cells.

**49. Calculate the maximum number of users that D-AMPS can support simultaneously within a single cell. Do the same calculation for GSM. Explain the difference. (M)**

D-AMPS uses 832 channels (in each direction) with three users sharing a single channel. This allows D-AMPS to support up to 2496 users simultaneously per cell. GSM uses 124 channels with eight users sharing a single channel. This allows GSM to support up to 992 users simultaneously. Both systems use about the same amount of spectrum (25 MHz in each direction).

D-AMPS uses  $30 \text{ KHz} \times 832 = 24.96 \text{ MHz}$ . GSM uses  $200 \text{ KHz} \times 124 = 24.80 \text{ MHz}$ . The difference can be mainly attributed to the better speech quality provided by GSM (13 Kbps per user) over D-AMPS (8 Kbps per user).

**50. Suppose that A, B, and C are simultaneously transmitting 0 bits, using a CDMA system with the chip sequences of Fig. 2-45(b). What is the resulting chip sequence? (E)**

A: 0 0 1 1 0 1 1  
B: 0 0 1 0 1 1 0  
C: 0 1 1 1 0 0 0  
D: 0 1 0 0 0 1 0

(a)

A: (-1 -1 -1 +1 +1 -1 +1)  
B: (-1 -1 +1 -1 +1 -1 +1)  
C: (-1 +1 -1 +1 +1 -1 -1)  
D: (-1 +1 1 -1 1 -1 +1)

(b)

Six examples:

-1 -1	D	$S_1 = (-1 -1 -1 +1 +1 -1 -1)$
-1 -1	B + C	$S_2 = (-2 0 0 0 0 2 0 -2)$
1 0 -1	A + B	$S_3 = (0 0 -2 +2 0 -2 0 -2)$
1 0 1 -1	A + B + C	$S_4 = (-1 1 1 -3 1 1 -1 -1)$
1 1 1 1	A + B + C + D	$S_5 = (-4 0 -2 0 -2 0 -2 -2)$
1 1 0 1	A + B + C + D	$S_6 = (-2 2 0 2 0 2 4 0)$

(c)

$S_1 \cdot C = (1 +1 -1 +1 +1 -1 -1) \cdot 0 = 1$   
 $S_2 \cdot C = (2 +3 +0 +0 +2 +0 -2) \cdot 0 = 1$   
 $S_3 \cdot C = (0 +0 -2 +2 +0 -2 -0 -2) \cdot 0 = 0$   
 $S_4 \cdot C = (1 +1 +3 +3 +1 -1 -1 -1) \cdot 0 = 1$   
 $S_5 \cdot C = (1 -0 +2 +0 -2 +0 -2 +2) \cdot 0 = 1$   
 $S_6 \cdot C = (2 -2 +0 2 +0 2 4 +0) \cdot 0 = 1$

(d)

传输 0，则时间片序列取其补码，传输 1，则时间序列取其本身。

将 A, B, C 相加后，取补码得结果为: (+3 +1 +1 -1 -3 -1 -1 +1).

**51. In the discussion about orthogonality of CDMA chip sequences, it was stated that if  $S \cdot T = 0$  then  $S \cdot \tilde{T}$  is also 0. Prove this.(E)**

By definition 
$$S \cdot T = \frac{1}{m} \sum_{i=1}^m S_i T_i$$

If  $T$  sends a 0 bit instead of 1 bit, its chip sequence is negated, with the  $i$ -th element becoming  $\tilde{T}_i$ . Thus,

$$S \cdot T = \frac{1}{m} \sum_{i=1}^m S_i (-T_i) = -\frac{1}{m} \sum_{i=1}^m S_i T_i = 0$$

**52. Consider a different way of looking at the orthogonality property of CDMA chip sequences. Each bit in a pair of sequences can match or not match. Express the orthogonality property in terms of matches and mismatches.**

When two elements match, their product is +1. When they do not match, their product is -1. To make the sum 0, there must be as many matches as mismatches. Thus, two chip sequences are orthogonal if exactly half of the corresponding elements match and exactly half do not match.

**53. A CDMA receiver gets the following chips: (-1 +1 -3 +1 -1 -3 +1 +1). Assuming the chip sequences defined in Fig. 2-45(b), which stations transmitted, and which bits did each one send? (E)**

分别与 (-1 +1 -3 +1 -1 -3 +1 +1) 做内积:

A:  $(-1 +1 -3 +1 -1 -3 +1 +1) \cdot (-1 -1 -1 +1 +1 -1 +1) / 8 = 1$

B:  $(-1 +1 -3 +1 -1 -3 +1 +1) \cdot (-1 -1 +1 -1 +1 +1 -1) / 8 = -1$

C:  $(-1 +1 -3 +1 -1 -3 +1 +1) \cdot (-1 +1 -1 +1 +1 -1 -1) / 8 = 0$

D:  $(-1 +1 -3 +1 -1 -3 +1 +1) \cdot (-1 +1 -1 -1 -1 -1 +1) / 8 = 1$

所以 A, D 发送的都是 1 bits, B 发送的是 0 bit, C 没有发送

## Chapter 3 The Data Link Layer Problems

**1. An upper-layer packet is split into 10 frames, each of which has an 80 percent chance of arriving undamaged. If no error control is done by the data link protocol, how many times must the message be sent on average to get the entire thing through? (E)**

由于每一帧有 0.8 的概率正确到达，整个信息正确到达的概率为  $p = 0.8^{10} = 0.107$ 。

为使信息完整的到达接收方，发送一次成功的概率是  $p$ ，二次成功的概率是  $(1-p)p$ ，三次成功的概率为  $(1-p)^2 p$ ， $i$  次成功的概率为  $(1-p)^{i-1} p$ ，因此平均的发送次



数等于: 
$$E = \sum_{i=1}^{\infty} i p (1-p)^{i-1} = \frac{1}{p} = \frac{1}{0.107} \approx 9.3$$

2. The following character encoding is used in a data link protocol: A: 01000111; B: 11100011; FLAG: 01111110; ESC: 11100000 Show the bit sequence transmitted (in binary) for the four-character frame: A B ESC FLAG when each of the following framing methods are used:

- (a) Character count.
- (b) Flag bytes with byte stuffing.
- (c) Starting and ending flag bytes, with bit stuffing. (E)

结果为

(a)字符计数 00000101 01000111 11100011 11100000 01111110

(b)字节填充 01111110 01000111 11100011 11100000 11100000 11100000 01111110 01111110

(c)位填充 01111110 01000111 11010001 11100000 01111101 01111110

3. The following data fragment occurs in the middle of a data stream for which the byte-stuffing algorithm described in the text is used: A B ESC C ESC FLAG FLAG D. What is the output after stuffing? (E)

填充后结果为: A B ESC ESC C ESC ESC FLAG ESC FLAG D.

4. One of your classmates, Scrooge, has pointed out that it is wasteful to end each frame with a flag byte and then begin the next one with a second flag byte. One flag byte could do the job as well, and a byte saved is a byte earned. Do you agree? (E)

只用一个标记位无法知道一帧何时结束。如果帧流是无限量的,一个标记位或许是可以的。但是当一帧以标记位结束了之后,在一段时间内(比如15分钟)没有新的帧到达时,接收者如何来分辨出下一字节是真正的新帧的开始还是碰巧是线路上的噪声呢?这样的协议设计的过于简单了。

5. A bit string, 011110111110111110, needs to be transmitted at the data link layer. What is the string actually transmitted after bit stuffing? (E)

输出: 011110111110011111010.

6. When bit stuffing is used, is it possible for the loss, insertion, or modification of a single bit to cause an error not detected by the checksum? If not, why not? If so, how? Does the checksum length play a role here? (M)

可能。假定原来的正文包含位序列 01111110 作为数据。位填充之后,这个序列将变成 011111010。如果由于传输错误第二个0丢失了,收到的位串又变成

01111110,被接收方看成是帧尾。然后接收方在该串的前面寻找检验和,并对它进行验证。如果检验和是16位,那么被错误的看成是检验和的16位的内容碰巧经验证后仍然正确的概率是 $1/2^{16}$ 。如果这种概率的条件成立了,就会导致不正确的帧被接收。显然,检验和段越长,传输错误不被发现的概率会越低,但该概率永远不等于零。

16. Data link protocols almost always put the CRC in a trailer rather than in a header. Why? (E)

CRC是在发送期间进行计算的。一旦把最后一位数据送上外出线路,就立即把CRC编码附加在输出流的后面发出。如果把CRC放在帧的头部,那么就要在发送之前把整个帧先检查一遍来计算CRC。这样每个字节都要处理两遍,第一遍是为了计算检验码,第二遍是为了发送。把CRC放在尾部就可以把处理时间减半。

17. A channel has a bit rate of 4 kbps and a propagation delay of 20 msec. For what range of frame sizes does stop-and-wait give an efficiency of at least 50 percent? (E)

当发送一帧的时间等于信道的传播延迟的2倍时,信道的利用率为50%。或者说,当发送一帧的时间等于来回路程的传播延迟时,效率将是50%。而在帧长满足发送时间大于延迟的两倍时,效率将会高于50%。

现在发送速率为4Mb/s,发送一位需要 $0.25\mu s$ 。

$(20 \times 10^{-3} \times 2) \div (0.25 \times 10^{-6}) = 160000 \text{ bit}$

只有在帧长不小于160kb时,停等协议的效率才会至少达到50%。

18. A 3000-km-long T1 trunk is used to transmit 64-byte frames using protocol 5. If the propagation speed is 6  $\mu\text{sec/km}$ , how many bits should the sequence numbers be? (M)

为了有效运行,序列空间(实际上就是发送窗口大小)必须足够的大,以允许发送方在收到第一个确认应答之前可以不断发送。信号在线路上的传播时间为

$6 \times 3000 = 18000 \mu s$ ,即18ms。

在T1速率,发送64字节的数据帧需花的时间:  $64 \times 8 \div (1.536 \times 10^6) = 0.33 \mu s$ 。

所以,发送的第一帧从开始发送起,18.33ms后完全到达接收方。确认应答又花了很少的发送时间(忽略不计)和回程的18ms。这样,加在一起的时间是36.33ms。发送方应该有足够大的窗口,从而能够连续发送36.33ms。

$36.33 / 0.33 = 110$

也就是说,为充满线路管道,需要至少110帧,因此序列号为7位。

这一题有争议的地方是T1的速度,T1的数据速率根据它的结构应该是 $(193-1-24)/193 \times 1.544 = 1.344 \text{ mbps}$ ,但是答案中是 $(193-1)/193 \times 1.544 = 1.536 \text{ mbps}$ 。

**19. In protocol 3, is it possible that the sender starts the timer when it is already running? If so, how might this occur? If not, why is it impossible?(M)**

协议 3 中，发送者有可能启动一个已经运行着的计时器。假设发送者传输一帧并且迅速返回一个错误的确认。主循环将会执行一段时间并且将会发送一帧，然而这时计时器仍然是运行着的。

**20. Imagine a sliding window protocol using so many bits for sequence numbers that wraparound never occurs. What relations must hold among the four window edges and the window size, which is constant and the same for both the sender and the receiver.(M)**

使发送者的窗口为  $(S_l, S_u)$ ，接收者的窗口为  $(R_l, R_u)$ ，窗口大小为  $W$ ，则它们之间的关系应满足：

$$0 \leq S_u - S_l + 1 \leq W$$

$$R_u - R_l + 1 = W$$

$$S_l \leq R_l \leq S_u + 1$$

**21. If the procedure between in protocol 5 checked for the condition  $a \leq b \leq c$  instead of the condition  $a \leq b < c$ , would that have any effect on the protocol's correctness or efficiency? Explain your answer. (H)**

改变检查条件后，协议将变得不正确。

假定使用 3 位序列号，考虑下列协议运行过程：A 站刚发出 7 号帧；B 站接收到这个帧，并发出捎带应答 ack；A 站收到 ack，并发送 0~6 号帧。假定所有这些帧都在传输过程中丢失了；B 站超时，重发它的当前帧，此时捎带的确认号是 7；考察 A 站在  $r.rack = 7$  到达时的情况，关键变量是  $ack\_expected = 0$ ， $r.rack = 7$ ， $next\_frame\_to\_send = 7$ 。

修改后的检查条件将被置成“真”，不会报告已发现的丢失帧错误，而误认为丢失了的帧已被确认。另一方面，如果采用原先的检查条件，就能够报告丢失帧的错误。所以结论是：为保证协议的正确性，已接收的确认应答号应该小于下一个要发送的序列号。

**22. In protocol 6, when a data frame arrives, a check is made to see if the sequence number differs from the one expected and  $no\_nak$  is true. If both conditions hold, a NAK is sent. Otherwise, the auxiliary timer is started. Suppose that the else clause were omitted. Would this change affect the protocol's correctness?(M)**

可能导致死锁。

假定有一组帧正确到达，并被接收。然后，接收方会向前移动窗口。现在假定所

有的确认帧都丢失了，发送方最终会产生超时事件，并且再次发送第一帧，接收方将发送一个 NAK。然后 NONAK 被置成伪。假定 NAK 也丢失了。那么从这个时候开始，发送方会不断发送已经被接收方接受的帧。接收方只是忽略这些帧，但由于 NONAK 为伪，所以不会再发送 NAK，从而产生死锁。如果设置辅助计数器（实现“else”子句），超时后重发 NAK，终究会使双方重新获得同步。

**23. Suppose that the three-statement while loop near the end of protocol 6 were removed from the code. Would this affect the correctness of the protocol or just the performance? Explain your answer.(M)**

删除这一段程序会影响协议的正确性，导致死锁。因为这一段程序负责处理接收到的确认帧，没有这一段程序，发送方会一直保持超时条件，从而使得协议的运行不能向前进展。

**24. Suppose that the case for checksum errors were removed from the switch statement of protocol 6. How would this change affect the operation of the protocol? (M)**

这将会失去使用 NAKs 的目的，因此我们不得不回退到超时。尽管性能上有所下降，但是不影响协议的正确性。NAKs 并不是关键的。

**25. In protocol 6 the code for `frame_arrival` has a section used for NAKs. This section is invoked if the incoming frame is a NAK and another condition is met. Give a scenario where the presence of this other condition is essential. (M)**

这里要求  $r.rack + 1 < next\_frame\_to\_send$ 。考虑下列操作细节：

A 站发送 0 号帧给 B 站。B 站收到此帧，并发送 ACK 帧，但 ACK 丢失了。A 站发生超时，重发 0 号帧。但 B 站现在期待接收 1 号帧，因此发送 NAK，否定收到的 0 号帧。显然，现在 A 站最好不重发 0 号帧。由于条件  $r.rack + 1 < next\_frame\_to\_send$  不成立，所以用不着选择性重传 0 号帧，可以继续向前推进传送 1 号帧。这个例子就说明了这段程序中的另一个条件，即  $r.rack + 1 < next\_frame\_to\_send$  也是重要的。

**26. Imagine that you are writing the data link layer software for a line used to send data to you, but not from you. The other end uses HDLC, with a 3-bit sequence number and a window size of seven frames. You would like to buffer as many out-of-sequence frames as possible to enhance efficiency, but you are not allowed to modify the software on the sending side. Is it possible to have a receiver window greater than 1, and still guarantee that the protocol will never fail? If so, what is the largest window that can be safely used? (E)**

不可以。最大接收窗口的大小就是 1。现在假定该接收窗口值变为 2。开始时发送方发送 0 至 6 号帧，所有 7 个帧都被收到，并作了确认，但确认被丢失。现在接收方准备接收 7 号和 0 号帧，当重发的 0 号帧到达接收方时，它将会被缓存保留，

接收方确认 6 号帧。当 7 号帧到来的时候，接收方将把 7 号帧和缓存的 0 号帧传递给主机，导致协议错误。因此，能够安全使用的最大窗口值为 1。

**27. Consider the operation of protocol 6 over a 1-Mbps error-free line. The maximum frame size is 1000 bits. New packets are generated 1 second apart. The timeout interval is 10 msec. If the special acknowledgement timer were eliminated, unnecessary timeouts would occur. How many times would the average message be transmitted? (E)**

发送 1 位用时间  $1\mu s$ ，发送 1000bit 的最长帧花时间 1ms。由于超时间隔是 10ms，而 1s 才能产生一个新的数据帧，所以超时是不可避免的。假定 A 站向 B 站发送一个帧，正确到达接收方，但较长时间无反向交通。不久，A 站发生超时事件，导致重发已发过的一帧。

B 站发现收到的帧的序列号错误，因为该序列号小于所期待接收的序列号。因此 B 站将发送一个 NAK，该 NAK 会携带一个确认号，导致不再重发该帧。结果每个帧都被发送两次。

**28. In protocol 6,  $MAX\_SEQ = 2^n - 1$ . While this condition is obviously desirable to make efficient use of header bits, we have not demonstrated that it is essential. Does the protocol work correctly for  $MAX\_SEQ = 4$ , for example? (M)**

不能，协议的运行将会失败。当  $MaxSeq=4$ ，序列号的模数  $=4+1=5$ ，窗口大小将等于： $NrBufs \leq 5/2=2.5$ ，即得到， $NrBufs=2$ 。因此在该协议中，偶数序号使用缓冲区 1。这种映射意味着帧 4 和 0 将使用同一缓冲区。假定 0 至 3 号帧都正确收到了，并且都确认应答了，并且都确认应答了。如果随后的 4 号帧丢失，且下一个 0 号帧收到了，新的 0 号帧将被放到缓冲区 0 中，变量 `arrived[0]` 被置成“真”。这样，一个失序帧将被投递给主机。事实上，采用选择性重传的滑动窗口协议需要  $MaxSeq$  是奇数才能正确的工作。然而其他的滑动窗口协议的实现并不具有这一性质。

**29. Frames of 1000 bits are sent over a 1-Mbps channel using a geostationary satellite whose propagation time from the earth is 270 msec. Acknowledgements are always piggybacked onto data frames. The headers are very short. Three-bit sequence numbers are used. What is the maximum achievable channel utilization for**

d. (a) Stop-and-wait.

e. (b) Protocol 5.

f. (c) Protocol 6. (E)

对应三种协议的窗口大小值分别是 1、7 和 4。

使用卫星信道端到端的典型传输延迟是 270ms，以 1Mb/s 发送，1000bit 长的帧

的发送时间为 1ms。我们用  $t=0$  表示传输开始的时间，那么在  $t=1ms$  时，第一帧发送完毕； $t=271ms$  时，第一帧完全到达接收方； $t=272ms$ ，对第一帧的确认帧发送完毕； $t=542ms$ ，带有确认的帧完全到达发送方。因此一个发送周期为 542ms。如果在 542ms 内可以发送  $k$  个帧，由于每一个帧的发送时间为 1ms，则信道利用率为  $k/542$ ，因此：

(a)  $k=1$ ，最大信道利用率  $=1/542=0.18\%$

(b)  $k=7$ ，最大信道利用率  $=7/542=1.29\%$

(c)  $k=4$ ，最大信道利用率  $=4/542=0.74\%$

**30. Compute the fraction of the bandwidth that is wasted on overhead (headers and retransmissions) for protocol 6 on a heavily-loaded 50-kbps satellite channel with data frames consisting of 40 header and 3960 data bits. Assume that the signal propagation time from the earth to the satellite is 270 msec. ACK frames never occur. NAK frames are 40 bits. The error rate for data frames is 1 percent, and the error rate for NAK frames is negligible. The sequence numbers are 8 bits. (M)**

使用选择性重传滑动窗口协议，序列号长度是 8 位。窗口大小为 128。卫星信道端到端的传输延迟是 270ms。以 50kb/s 发送，4000bit (3960+40) 长的数据帧的发送时间是  $0.02*4000=80ms$ 。我们用  $t=0$  表示传输开始时间，那么， $t=80ms$ ，第一帧发送完毕；

$t=270+80=350ms$ ，第一帧完全到达接收方； $t=350+80=430ms$ ，对第一帧作捎带确认的反向数据帧可能发送完毕； $t=430+270=700ms$ ，带有确认的反向数据帧完全到达发送方。因此，周期为 700ms，发送 128 帧时间  $80*128=10240ms$ ，这意味着传输管道总是充满的。每个帧重传的概率为 0.01，对于 3960 个数据位，头开销为 40 位，平均重传的位数为  $4000*0.01=40$  位，传送 NAK 的平均位数为  $40*1/100=0.40$  位，所以每 3960 个数据位的总开销为 80.4 位。

因此，开销所占的带宽比例等于  $80.4/(3960+80.4)=1.99\%$ 。

**31. Consider an error-free 64-kbps satellite channel used to send 512-byte data frames in one direction, with very short acknowledgements coming back the other way. What is the maximum throughput for window sizes of 1, 7, 15, and 127? The earth-satellite propagation time is 270 msec. (M)**

使用卫星信道端到端的传输延迟为 270ms，以 64kb/s 发送，周期等于 604ms。发送一帧的时间为 64ms，我们需要  $604/64=9$  个帧才能保持通道不空。

对于窗口值 1，每 604ms 发送 4096 位，吞吐率为  $4096/0.604=6.8kb/s$ 。

对于窗口值 7，每 604ms 发送  $4096*7$  位，吞吐率为  $4096*7/0.604=47.5kb/s$ 。

对于窗口值超过 9 (包括 15、127)，吞吐率达到最大值，即 64kb/s。

32. A 100-km-long cable runs at the T1 data rate. The propagation speed in the cable is 2/3 the speed of light in vacuum. How many bits fit in the cable? (E)

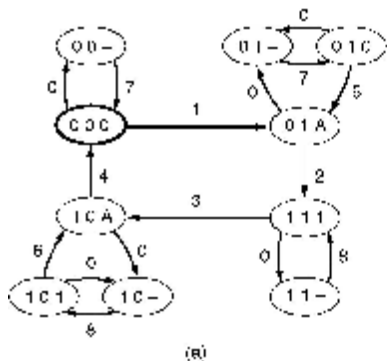
在该电缆中的传播速度是每秒钟 200 000km, 即每毫秒 200km, 因此 100km 的电缆将会在 0.5ms 内填满。T1 速率 125 $\mu$ s 传送一个 193 位的帧, 0.5ms 可以传送 4 个 T1 帧, 即 193\*4=772bit。

33. Suppose that we model protocol 4 using the finite state machine model. How many states exist for each machine? How many states exist for the communication channel? How many states exist for the complete system (two machines and the channel)? Ignore the checksum errors. (M)

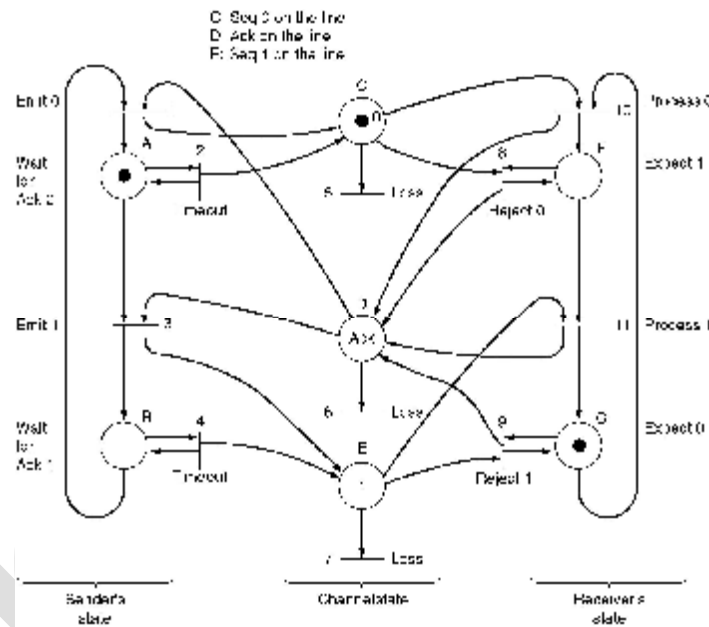
每台机器都有两个关键的变量 *next3frame3to3send* and *frame3expected*, 每个都可以取值 0 或 1。因此, 每台机器都有四种可能的状态。在信道上的一个消息包含了要被发送的帧的序列号和被确认的帧的序列号。因此, 存在四种类型的消息。信道可能在每个方向上有 0 或 1 条消息。所以, 信道上状态的数量是带着 0 条消息的 1 个, 带着 1 条消息的 8 个, 带着 2 条消息的 16 个 (每个方向上只有一条消息)。

总共有 1 + 8 + 16 = 25 种可能的信道状态。对完整的系统这隐含了 4\*4\*25 = 400 种可能的状态。

34. Give the firing sequence for the Petri net of Fig. 3-23 corresponding to the state sequence (000), (01A), (01—), (010), (01A) in Fig. 3-21. Explain in words what the sequence represents. (M)

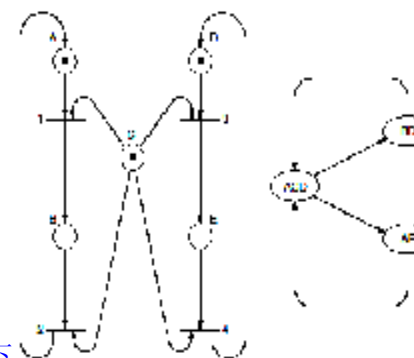


Transition	Who times?	Frame accepted	Frame emitted	To network layer
0	-	(frame lost)	-	-
1	S	0	A	Yes
2	S	A	-	-
3	D	1	A	Yes
4	S	A	0	-
5	D	0	A	No
6	S	1	A	No
7	S	(timeout)	0	-
8	S	(timeout)	-	-



激发序列是 10, 6, 2, 8。它通信去接收一个偶数, 确认丢失, 发送者超时, 并且重新由接收者生成确认。

35. Given the transition rules  $AC \rightarrow B$ ,  $B \rightarrow AC$ ,  $CD \rightarrow E$ , and  $E \rightarrow CD$ , draw the Petri net described. From the Petri net, draw the finite state graph reachable from the initial state ACD. What well-known concept do these transition rules model?(E)



Petri 网和状态图如下

系统模型是互斥的。B 和 E 是关键段它们不能同时被激活, 例如不允许状态 BE, 位置 C 代表一个符号它可以被 A 或 D 推出, 但是不能同时被 AD 推出。

本题是关于 Petri 网的, 根据题目的描述, 应该是 B 推出 C, E 推出 C, 所以答案的图箭头应该由 B、E 指向 C。



## Chapter 4 The Medium Access Control Sublayer Problems

1. For this problem, use a formula from this chapter, but first state the formula. Frames arrive randomly at a 100-Mbps channel for transmission. If the channel is busy when a frame arrives, it waits its turn in a queue. Frame length is exponentially distributed with a mean of 10,000 bits/frame. For each of the following frame arrival rates, give the delay experienced by the average frame, including both queueing time and transmission time. (E)

- a. (a) 90 frames/sec.
- b. (b) 900 frames/sec.
- c. (c) 9000 frames/sec.

排队理论延迟时间公式:  $T = 1/(\mu C - \lambda)$ , 这里信道容量  $C = 10^8$  and  $\mu = 10^{-4}$ , 所以  $T = 1/(10000 - \lambda)$  sec, 对上面的三种帧到达率  $\lambda$ , 有 (a) 0.1 msec, (b) 0.11 msec, (c) 1 msec. 对于 (c) 由  $\rho = \lambda/\mu C = 0.9$ , 10 倍延迟。

2. A group of N stations share a 56-kbps pure ALOHA channel. Each station outputs a 1000-bit frame on an average of once every 100 sec, even if the previous one has not yet been sent (e.g., the stations can buffer outgoing frames). What is the maximum value of N? (E)

对于纯的 ALOHA, 信道利用率为  $1/e^2 = 0.184$ , 可用的带宽是  $0.184 \times 56$  Kb/s =  $10.304$  Kb/s. 每个站需要的带宽为  $1000/100 = 10$  b/s. 而  $N = 10304/10 \approx 1030$  所以, 最多可以有 1030 个站, 即 N 的最大值为 1030。

3. Consider the delay of pure ALOHA versus slotted ALOHA at low load. Which one is less? Explain your answer. (E)

对于纯的 ALOHA, 发送可以立即开始。对于分隙的 ALOHA, 它必须等待下一个时隙。这样, 平均会引入半个时隙的延迟。因此, 纯 ALOHA 的延迟比较小。

4. Ten thousand airline reservation stations are competing for the use of a single slotted ALOHA channel. The average station makes 18 requests/hour. A slot is 125  $\mu$ sec. What is the approximate total channel load? (E)

每个终端每 200 ( $=3600/18$ ) 秒做一次请求, 总共有 10 000 个终端, 因此, 总的负载是 200 秒做 10000 次请求。平均每秒钟 50 次请求。每秒钟 8000 个时隙, 所以平均每个时隙的发送次数为  $50/8000 = 1/160$ 。

5. A large population of ALOHA users manages to generate 50 requests/sec, including both originals and retransmissions. Time is slotted in units of 40 msec.

- d. (a) What is the chance of success on the first attempt?

- e. (b) What is the probability of exactly k collisions and then a success?
- f. (c) What is the expected number of transmission attempts needed? (M)

(a) 在任一帧时间内生成 k 帧的概率服从泊松分布  $\Pr[k] = \frac{G^k e^{-G}}{k!}$

生成 0 帧的概率为  $e^{-G}$ ; 对于纯的 ALOHA, 发送一帧的冲突危险区为两个帧时, 在两帧内无其他帧发送的概率是  $e^{-G} \times e^{-G} = e^{-2G}$ ; 对于分隙的 ALOHA, 由于冲突危险区减少为原来的一半, 任一帧时内无其他帧发送的概率是  $e^{-G}$ 。

现在时隙长度为 40ms, 即每秒 25 个时隙, 产生 50 次请求, 所以每个时隙产生两个请求,  $G=2$ 。因此, 首次尝试的成功率是:  $e^{-2} = 1/e^2$

(b)  $(1 - e^{-G})^k e^{-G} = (1 - e^{-2})^k e^{-2} = 0.135 \times (1 - 0.135)^k = 0.135 \times 0.865^k$

(c) 尝试 k 次才能发送成功的概率 (即前 k-1 次冲突, 第 k 次才成功) 为:

$$p_k = e^{-G} (1 - e^{-G})^{k-1}$$

那么每帧传送次数的数学期望为  $E = \sum_{k=1}^{\infty} k p_k = \sum_{k=1}^{\infty} k e^{-G} (1 - e^{-G})^{k-1} = e^G = e^2 = 7.4$

6. Measurements of a slotted ALOHA channel with an infinite number of users show that 10 percent of the slots are idle.

- g. (a) What is the channel load, G?
- h. (b) What is the throughput?
- i. (c) Is the channel underloaded or overloaded?(E)

(a) 从泊松定律得到  $p_0 = e^{-G}$ , 因此  $G = -\ln p_0 = -\ln 0.1 = 2.3$

(b)  $S = G e^{-G}$ ,  $G = 2.3$ ,  $e^{-G} = 0.1$ ,  $S = 2.3 \times 0.1 = 0.23$

(c) 因为每当  $G > 1$  时, 信道总是过载的, 因此在这里信道是过载的。

7. In an infinite-population slotted ALOHA system, the mean number of slots a station waits between a collision and its retransmission is 4. Plot the delay versus throughput curve for this system. (M)

每帧传送次数的数学期望为:  $E = \sum_{k=1}^{\infty} k p_k = \sum_{k=1}^{\infty} k e^{-G} (1 - e^{-G})^{k-1} = e^G$

E 个事件为 E-1 个长度等于 4 个时隙的间隔时间所分隔。因此一个帧从第一次发送开始时间到最后一次尝试成功的发送开始时间之间的长度即延迟是  $4(e^G - 1)$ , 吞吐率  $S = G e^{-G}$ 。

对于每一个 G 值, 都可以计算出对应的延迟值  $D = 4(e^G - 1)$ , 以及吞吐率值  $S = G e^{-G}$ 。

按此方法即可画出时延对吞吐率的曲线。

**8. How long does a station,  $s$ , have to wait in the worst case before it can start transmitting its frame over a LAN that uses**

**j. (a) the basic bit-map protocol?**

**k. (b) Mok and Ward's protocol with permuting virtual station numbers? (M)**

(a) The worst case is: all stations want to send and  $s$  is the lowest numbered station. Wait time  $N$  bit contention period +  $(N-1) d$  bit for transmission of frames. The total is  $N+(N-1) d$  bit times. (b) The worst case is: all stations have frames to transmit and  $s$  has the lowest virtual station number.

Consequently,  $s$  will get its turn to transmit after the other  $N-1$  stations have transmitted one frame each, and  $N$  contention periods of size  $\log_2 N$  each.

Wait time is thus  $(N+1) \times d + N \times \log_2 N$  bits.

当竞争周期刚刚扫描过的时候， $N$  号站点正好由数据发送，所以他要等到数据 1 发送完（最差时前面  $N-1$  个站点都有数据要发，即  $(N-1) * d$ ），然后下一个扫描周期（ $N$  位），再等前面所有站点发送完数据；所以总的延迟为：

$$(N-1) * d + N + (N-1) * d = 2(N-1) * d + N ;$$

**9. A LAN uses Mok and Ward's version of binary countdown. At a certain instant, the ten stations have the virtual station numbers 8, 2, 4, 5, 1, 7, 3, 6, 9, and 0. The next three stations to send are 4, 3, and 9, in that order. What are the new virtual station numbers after all three have finished their transmissions? (E)**

当 4 站发送时，它的号码变为 0，而 0、1、2 和 3 号站的号码都增 1，10 个站点的虚站号变为 8, 3, 0, 5, 2, 7, 4, 6, 9, 1

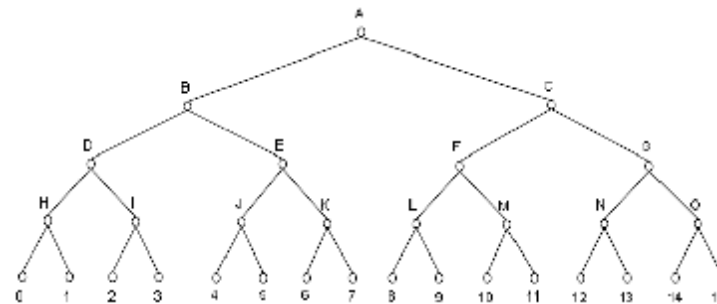
当 3 站发送时，它的号码变为 0，而 0、1 和 2 站的号码都增 1，10 个站点的虚站号变为：8, 0, 1, 5, 3, 7, 4, 6, 9, 2

最后，当 9 站发送时，它变成 0，所有其他站都增 1，结果是：9, 1, 2, 6, 4, 8, 5, 7, 0, 3。

**10. Sixteen stations, numbered 1 through 16, are contending for the use of a shared channel by using the adaptive tree walk protocol. If all the stations whose addresses are prime numbers suddenly become ready at once, how many bit slots are needed to resolve the contention? (E)**

在自适应树遍历协议中，可以把站点组织成二叉树（见图）的形式。在一次成功的传输之后，在第一个竞争时隙中，全部站都可以试图获得信道，如果仅其中之一需用信道，则发送冲突，则第二时隙内只有那些位于节点 B 以下的站（0 到 7）可以参加竞争。如其中之一获得信道，本帧后的时隙留给站点 C 以下的站；如果 B 点

下面有两个或更多的站希望发送，在第二时隙内会发生冲突，于是第三时隙内由 D 节点以下各站来竞争信道。



本题中，站 2、3、5、7、11 和 13 要发送，需要 11 个时隙，每个时隙内参加竞争的站的列表如下：

第一时隙：2、3、5、7、11、13

第二时隙：2、3、5、7

第三时隙：2、3

第四时隙：2

第五时隙：3

第六时隙：5、7

第七时隙：5

第八时隙：7

第九时隙：11、13

第十时隙：11

第十一时隙：13

**11. A collection of  $2^n$  stations uses the adaptive tree walk protocol to arbitrate access to a shared cable. At a certain instant, two of them become ready. What are the minimum, maximum, and mean number of slots to walk the tree if  $2^n \gg 1$ ? (H)**

$2^n$  个站点对应  $n+1$  级，其中 0 级有 1 个节点，1 级有 2 个节点， $n$  级有  $2^n$  个节点。在  $i$  级的每个节点下面所包括的站的个数等于总站数的  $1/2^i$ 。本题中所需的时隙数取决于为了到达准备好发送的两个站的共同先辈点必须往回走多少级。先计算这两个站具有共同的父节点的概率  $p_1$ 。在  $2^n$  个站中，要发送的两个站共享一个

指定的父节点的概率是  $\frac{1}{2^{n-1}} = 2^{-(n-1)}$

总共  $2^{n-1}$  个父节点，所以，
$$p_1 = \frac{1}{2^{n-1}(2^n - 1)} \cdot 2^{n-1} = \frac{1}{2^n - 1}$$

因为  $2^n \gg 1$ ，所以  $p \approx 2^{-n}$

在共享父节点条件下遍历树，从第二级开始每一级访问两个节点，这样遍历树所走过的节点总数  $n_1 = 1 + 2 + \dots + 2 + 2 = 1 + 2n$ ，接下来，我们考察两个发送站共享祖父节点的概率  $p_2$  和遍历树所走过的节点总数  $n_2$ 。此时在每个父节点下面仅可能有一个站发送。两个发送站共享一个指定的祖父节点的概率是  $1/C_2^{2^{n-1}}$ 。

共有  $2^{n-2}$  个祖父节点 
$$p_2 = \frac{2^{n-2}}{C_2^{2^{n-1}}} = \frac{1}{2^{n-1} - 1} \approx \frac{1}{2^{n-1}} = 2^{-n+1}$$

遍历树比  $1 + n$  减少两个节点，即  $n_2 = 1 + 2n - 2 = 2n - 1$

通过类似的分析和计算，可以得到，两个发送站共享曾祖父节点（属  $n-3$  级祖先节点）的概率是  $p_3 = 2^{-n+2}$

遍历树所经过的节点总数比  $n_2$  又少两个节点，

$$\begin{aligned} n_3 &= 2n - 1 - 2 = 2n - 3 \\ &\vdots \\ p_{i+1} &= 2^{-(n-i)} \\ n_{i+1} &= 2n + 1 - 2i \end{aligned}$$

因此，最坏的情形是  $2n+1$  个时隙（共享父节点），对应于  $i=0$ ；

最好的情形是 3 个时隙，对应于  $i=n-1$ （两个发送站分别位于左半树和右半树），

所以平均时隙数等于 
$$m = \sum_{i=0}^{n-1} 2^{-(n-i)} (2n + 1 - 2i)$$

该表达式可以简化为 
$$m = (1 - 2^{-n})(2n + 1) - 2^{-(n-1)} \sum_{i=0}^{n-1} i 2^i$$

最小为 3 个，最大为  $n+2$  个，给出的答案最大不对。

最大的情况为两个站点兄弟，共父母。

两个黄色的园代表要发数据的站点

所以在这种情况下的时槽为：数的路径长度加 2

即：  $n+2$

**12. The wireless LANs that we studied used protocols such as MACA instead of using CSMA/CD. Under what conditions, if any, would it be possible to use CSMA/CD instead?(E)**

如果所有站的发射有效范围都很大，以至于任一站都可以收到所有其他站发送的信号，那么任一站都可以与其他站以广播方式通信。在这样的条件下，CSMA/CD 可以工作的很好。

**13. What properties do the WDMA and GSM channel access protocols have in common? See Chap. 2 for GSM. (E)**

两种协议都使用 FDM 和 TDM 结合的方法，它们都可以提供专用的频道（波长），并且都划分时隙，实现 TDM。

**14. Six stations, A through F, communicate using the MACA protocol. Is it possible that two transmissions take place simultaneously? Explain your answer.(E)**

是的。想像一下它们都在一条直线上并且每个站都只能连到它最近的邻居，那么 A 可以发送给 B 同时 E 正发送给 F

**15. A seven-story office building has 15 adjacent offices per floor. Each office contains a wall socket for a terminal in the front wall, so the sockets form a rectangular grid in the vertical plane, with a separation of 4 m between sockets, both horizontally and vertically. Assuming that it is feasible to run a straight cable between any pair of sockets, horizontally, vertically, or diagonally, how many meters of cable are needed to connect all sockets using**

**l. (a) a star configuration with a single router in the middle?**

**m. (b) an 802.3 LAN? (E)**

(a) 从一到七层记数。在星形配置中，路由器在第四层中央。需要铜线的站个数 7

\*15 - 1 = 104 sites. 这些铜线的总长度 
$$4 \sum_{i=1}^7 \sum_{j=1}^{15} \sqrt{(i-4)^2 + (j-8)^2} = 1832 \text{ meters.}$$

(b) 对 802.3, 7 水平铜线每层需要 56 m 长，加上竖直方向的共 24 m，总共 416 m.

**16. What is the baud rate of the standard 10-Mbps Ethernet? (E)**

以太网使用曼彻斯特编码，这就意味着发送的每一位都有两个信号周期。标准以太网的数据率为 10Mb/s，因此波特率是数据率的两倍，即 20MBaud。

**17. Sketch the Manchester encoding for the bit stream: 0001110101.(E)**

信号是一个二值方波高 (H) 和低(L)，形式为 LHLHLHHLHLHLLHHLHHL.

**18. Sketch the differential Manchester encoding for the bit stream of the previous problem. Assume the line is initially in the low state.(E)**

形式为 HLHLHLLHLLHLHLHLLH.

**19. A 1-km-long, 10-Mbps CSMA/CD LAN (not 802.3) has a propagation speed of 200 m/μsec. Repeaters are not allowed in this system. Data frames are 256 bits long, including 32 bits of header, checksum, and other overhead. The first bit slot after a successful transmission is reserved for the receiver to capture the channel in order to send a 32-bit acknowledgement frame. What is the effective data rate, excluding overhead, assuming that there are no collisions?(M)**

依题知一公里的在铜缆中单程传播时间为  $1/200000=5\times 10^{-6}$  s=5 usec，往返传播时间为  $2t=10$  usec，一次完整的传输分为 6 步：

发送者侦听铜缆时间为 10usec，若线路可用

发送数据帧传输时间为  $256 \text{ bits} / 10\text{Mbps} = 25.6 \text{ usec}$

数据帧最后一位到达时的传播延迟时间为 5.0usec

接收者侦听铜缆时间为 10 usec，若线路可用

接收者发送确认帧用时 3.2 usec

确认帧最后一位到达时的传播延迟时间为 5.0 usec

总共 58.8sec，在这期间发送了 224 bits 的数据，所以数据传输率为 3.8 Mbps.

**20. Two CSMA/CD stations are each trying to transmit long (multiframe) files. After each frame is sent, they contend for the channel, using the binary exponential backoff algorithm. What is the probability that the contention ends on round  $k$ , and what is the mean number of rounds per contention period?(M)**

把获得通道的尝试从 1 开始编号。第  $i$  次尝试分布在  $2^{i-1}$  个时隙中。因此， $i$  次尝试碰撞的概率是  $2^{-(i-1)}$ ，开头  $k-1$  次尝试失败，紧接着第  $k$  次尝试成功的概率是：

$$p_k = (1 - 2^{-(k-1)})[2^{-0} \cdot 2^{-1} \cdot 2^{-2} \dots 2^{-(k-2)}] = (1 - 2^{-(k-2)})2^{-(k-1)(k-2)/2}$$

即：

$$P_k = (1 - 2^{-(k-1)}) \prod_{i=1}^{k-1} 2^{-(i-1)}$$

上式可简化为：

$$P_k = (1 - 2^{-(k-1)}) 2^{-(k-1)(k-2)/2}$$

所以每个竞争周期的平均竞争次数是  $\sum k p_k$

**21. Consider building a CSMA/CD network running at 1 Gbps over a 1-km cable with no repeaters. The signal speed in the cable is 200,000 km/sec. What is the minimum frame size?(E)**

对于 1km 电缆，单程传播时间为  $1/200000=5\times 10^{-6}$  s=5μs，往返传播时间为  $2t=10\mu s$ 。为了能够按照 CSMA/CD 工作，最小帧的发射时间不能小于  $10\mu s$ 。以 1Gb/s 速率工作， $10\mu s$  可以发送的比特数等于： $(10\times 10^{-6}) / (1\times 10^{-9}) = 10000$

因此，最小帧是 10000 bit = 1250 字节长。

**22. An IP packet to be transmitted by Ethernet is 60 bytes long, including all its headers. If LLC is not in use, is padding needed in the Ethernet frame, and if so, how many bytes?(E)**

最小的以太网帧是 64 bytes，包含了以太网地址帧头，类型/长度域，以及校验和。由于帧头域占用 18 bytes，并且分组是 60 bytes，总帧长是 78 bytes，这已经超过了 64-byte 的最小限制。因此，不必再填充了。

**23. Ethernet frames must be at least 64 bytes long to ensure that the transmitter is still going in the event of a collision at the far end of the cable. Fast Ethernet has the same 64-byte minimum frame size but can get the bits out ten times faster. How is it possible to maintain the same minimum frame size?(E)**

将快速以太网的电缆长度至为以太网的 1/10 即可。

**24. Some books quote the maximum size of an Ethernet frame as 1518 bytes instead of 1500 bytes. Are they wrong? Explain your answer. (E)**

以太网一帧中数据占用是 1500 bytes，但是把目的地地址，源地址，类型/长度域以及校验和域也算上，帧总长就为 1518 bytes

**25. The 1000Base-SX specification states that the clock shall run at 1250 MHz, even though gigabit Ethernet is only supposed to deliver 1 Gbps. Is this higher speed to provide for an extra margin of safety? If not, what is going on here?(E)**

传输数据用 10 位来表示 8 位的真实数据，编码的利用率是 80%，一秒钟可以传送 1250 mb 的数据，相当于  $125\times 10^6$  码字。每个码字代表的是 8 位有效数据，所以实际的数据传输率是 1000 mb/sec.

**26. How many frames per second can gigabit Ethernet handle? Think carefully and take into account all the relevant cases. Hint: the fact that it is gigabit Ethernet matters.(E)**

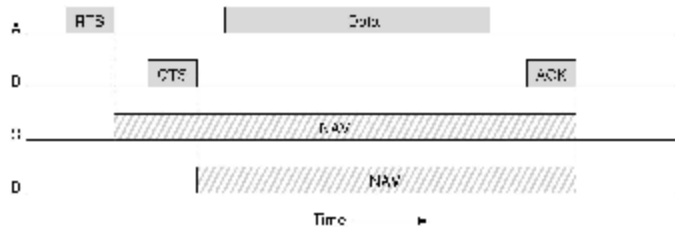
最小的以太网帧是 64bytes = 512 bits，所以依题 1 Gbps 的带宽可得  $1,953,125 = 2\times 10^6$  frames/sec，然而，这只是在充满最小的帧时是这样，如果没有充满帧，填充短帧至 4096 bits，这时每秒处理的帧的最大数量为 244,140 bytes，对于最大的帧长 12,144 bits，每秒处理的帧的最大数量为 82,345 frames/sec.

**27. Name two networks that allow frames to be packed back-to-back. Why is this feature worth having?(E)**



千兆以太网和 802.16 都有这个特性。这不仅对于提高带宽的使用效率很有用，同时对于帧长的要求限制不多。

28. In Fig. 4-27, four stations, A, B, C, and D, are shown. Which of the last two stations do you think is closest to A and why?(E)



站 C 最接近 A。因为 C 最先听到 A 发出的 RTS 并且通过插入一个 NAV 信号作为回应。D 对其没有回应，说明它不在 A 的频率范围内。

29. Suppose that an 11-Mbps 802.11b LAN is transmitting 64-byte frames back-to-back over a radio channel with a bit error rate of  $10^{-7}$ . How many frames per second will be damaged on average? (E)

一帧是 64bytes=512 bits, 位出错率为  $p=10^{-7}$ , 所有 512 位正确到达的概率为  $(1-p)^{512} = 0.9999488$ , 所以帧被破坏的概率约为  $5 \times 10^{-5}$ , 每秒钟发送的帧数为  $11 \times 10^6 / 512 = 21,484 \text{ frames/sec}$ , 将上两个数乘一下, 大约每秒钟有一帧被破坏。

30. An 802.16 network has a channel width of 20 MHz. How many bits/sec can be sent to a subscriber station? (E)

这取决于离子站有多远。如果子站就在附近, 那么使用 QAM-64 可得带宽 120 Mbps; 中等距离时, 使用 QAM-16 可得带宽 80 Mbps; 远程距离, QPSK 可得带宽 40 Mbps。

(原题给出的是 20mhz 的带宽, 要求的是数据率, 按照前面的 Nyquist 定理, 最大数据率应该是:  $2H \log N$ , 但是答案没有乘以 2。)

31. IEEE 802.16 supports four service classes. Which service class is the best choice for sending uncompressed video? (E)

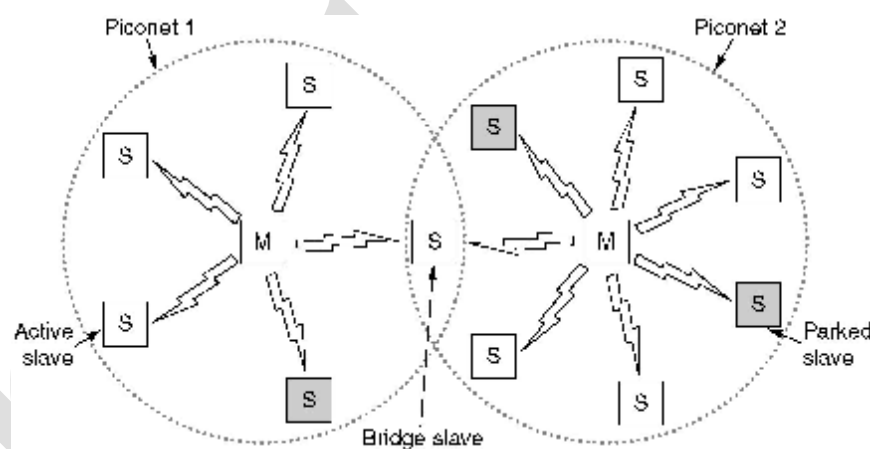
未压缩的视频有一个固定的位速率。每帧都有与前一帧相同的点数量, 因此, 可能要准确计算需要的带宽。最后, 最好选用固定位速率服务。

32. Give two reasons why networks might use an error-correcting code instead of error detection and retransmission. (M)

一个原因是实时服务质量的需要, 如果发现了一个错误, 并没有时间去做重传, 必须继续传下去, 这里可以使用转发时纠正错误。另一个原因在低质量的线路上(利用无线带宽), 错误率相当高最后所有的帧都必须重传, 并且重传时可能也会被破

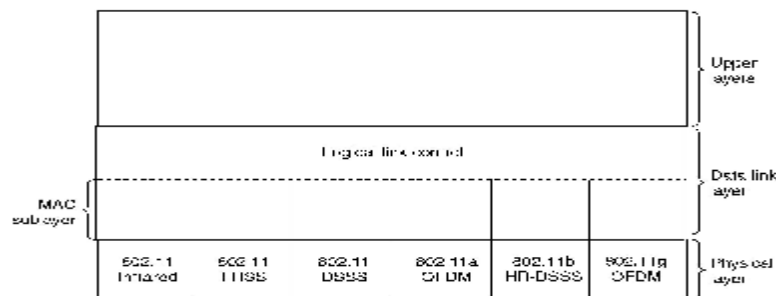
坏。为了避免这些, 转发时纠正错误码用于提高到达的帧的正确率的比例。

33. From Fig. 4-35, we see that a Bluetooth device can be in two piconets at the same time. Is there any reason why one device cannot be the master in both of them at the same time? (M)



一个设备不可能同时是两个微微网中的主节点。这样会引起两个问题: 首先, 头部只有三位地址可用, 每个微微网有 7 个从结点, 因此, 每个从结点没有独立的地址。其次, 起始帧的访问码源于主节点的标记符, 这是子节点区分消息来自于哪个微微网。如果两个重叠的微微网使用相同的访问码, 就会区分不出帧是属于哪个微微网的。实际上, 两个微微网会合并到一个大的微微网而不是相互独立的两个。

34. Figure 4-25 shows several physical layer protocols. Which of these is closest to the Bluetooth physical layer protocol? What is the biggest difference between the two? (E)



蓝牙同 802.11 一样使用的是 FHSS。最大的不同在于蓝牙每秒的跳数是 1600 hops, 远快于 802.11

**35. Bluetooth supports two types of links between a master and a slave. What are they and what is each one used for? (E)**

ACL 频道是异步的，随着不规则到达的帧产生数据。

SCO 频道是同步的，随着周期性到达的帧运行在一个稳定的速率下。

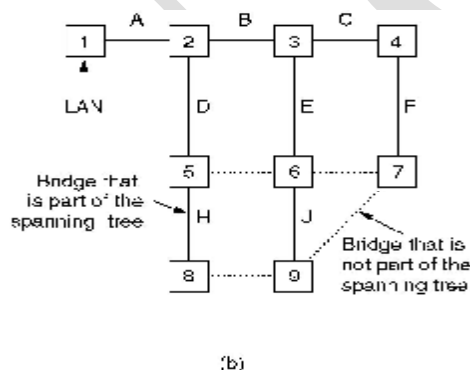
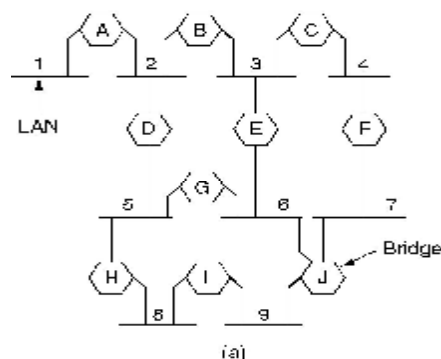
**36. Beacon frames in the frequency hopping spread spectrum variant of 802.11 contain the dwell time. Do you think the analogous beacon frames in Bluetooth also contain the dwell time? Discuss your answer. (M)**

不是的。在 802.11 中停延时间不是标准化的，所以它必须对到达的新站声明，在蓝牙里这需要 625usec。

没有必要对这去声明，所有的蓝牙设备中都有这样的硬件芯片，蓝牙被设计得便宜，并且固定跳率和停延时间的话处理芯片会更便宜。

**37. Consider the interconnected LANs shown in Fig. 4-44. Assume that hosts a and b are on LAN 1, c is on LAN 2, and d is on LAN 8. Initially, hash tables in all bridges are empty and the spanning tree shown in Fig 4-44(b) is used. Show how the hash tables of different bridges change after each of the following events happen in sequence, first (a) then (b) and so on. (M)**

- n. (a) a sends to d.
- o. (b) c sends to a.
- p. (c) d sends to c.
- q. (d) d moves to LAN 6.
- r. (e) d sends to a.



第一帧会被所有的网桥转发。这传完后，每个网桥都会通过它的散列表中的合适端口建一个到目的地 a 的表项。例如，D 当前的散列表为从 LAN 2 转发到目的地 a；第二帧将会被网桥 B, D, and A 收到。这些网桥会在它们的散列表中增加一个新的表

项，帧的目的地到 c；例如网桥 D 的散列表会又有一个从 LAN 2 转发到目的地 c 的表项；第三帧将会被网桥 H, D, A, 以及 B 收到，这些网桥将会在它们的散列表中增加一个表项，帧的目的地到 d；第四帧将会被网桥 E, C, B, D, 以及 A 收到，网桥 E 和 C 会在它们的散列表中增加一个表项，帧的目的地到 d，当网桥 D, B, 以及 A 更新它们的散列表项时，到达目的地 d

当 d 移到 LAN6 上去后，如果他发数据给 a 的话，路由 j，一定可以知道 d 在 LAN6 上的，答案中没有给出 j 的这一表项。

**38. One consequence of using a spanning tree to forward frames in an extended LAN is that some bridges may not participate at all in forwarding frames. Identify three such bridges in Fig. 4-44. Is there any reason for keeping these bridges, even though they are not used for forwarding? (E)**

网桥 G, I and J 不用于转发任何帧。主要原因在于有循环可以为一个扩展的 LAN 提高可靠性。如果当前生成树中的网桥有坏的，则动态生成树算法会重新配置一棵新的生成树，它可能会包含一个或多个不在原先的生成树中的网桥。

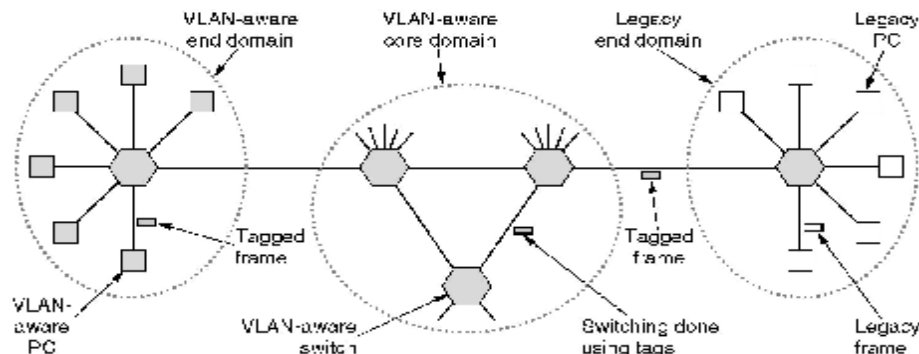
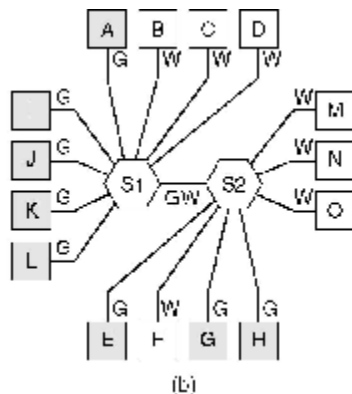
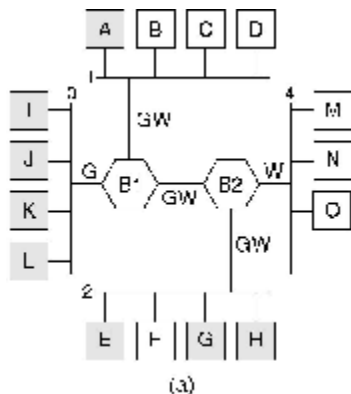
**39. Imagine that a switch has line cards for four input lines. It frequently happens that a frame arriving on one of the lines has to exit on another line on the same card. What choices is the switch designer faced with as a result of this situation? (E)**

最简单做选择就是不做特殊处理。每一个到来的帧被输出到底板并被发送到目标卡上。这时，卡内流量通过底板；另一种选择就是识别出这种情况并且对其专门处理，直接发送帧并且不通过底板。

**40. A switch designed for use with fast Ethernet has a backplane that can move 10 Gbps. How many frames/sec can it handle in the worst case? (E)**

最坏的情况就是无穷的 64-byte (512-bit) 帧流。如果底板能处理  $10^9$  bps，可处理的帧的数量为  $10^9 / 512 = 1,953,125$  frames/sec.

**41. Consider the network of Fig. 4-49(a). If machine J were to suddenly become white, would any changes be needed to the labeling? If so, what? (E)**



可以工作。进入核心域的帧都会是合法帧，因此帧会在第一个核心交换机里被加上标记，可以用 MAC 或 IP 地址。类似的，在出口处，交换机去掉这些标记后再输出帧。

端口 B1 到 LAN 3 需要被重新标记为 GW.

#### 42. Briefly describe the difference between store-and-forward and cut-through switches. (E)

一个存储转发交换机在它的表项里存储进来的每一帧，然后检查并且转发它；直通型交换机帧一进来就完全转发掉。只要目的地地址是可用的，转发就可以开始。

#### 43. Store-and-forward switches have an advantage over cut-through switches with respect to damaged frames. Explain what it is. (E)

存储转发型交换机存储整个帧然后转发它们。一个帧到来后，可以验证校验和，如果帧被破坏了，马上丢掉坏帧。用直通型交换机，坏帧不能被交换机丢掉，因为那时检测错误的同时帧就已经转掉了。

#### 44. To make VLANs work, configuration tables are needed in the switches and bridges. What if the VLANs of Fig. 4-49(a) use hubs rather than multidrop cables? Do the hubs need configuration tables, too? Why or why not? (E)

不需要，集线器只是将所有的输入线收集在一起，并没有进行配置。在集线器中不做路由，进入到集线器的每一帧分发到所有其它的线路上。

#### 45. In Fig. 4-50 the switch in the legacy end domain on the right is a VLAN-aware switch. Would it be possible to use a legacy switch there? If so, how would that work? If not, why not? (E)

### Chapter 5 The Network Layer Problems

#### 1. Give two example computer applications for which connection-oriented service is appropriate. Now give two examples for which connectionless service is best.(E)

文件传送、远程登录和视频点播需要面向连接的服务。另一方面，信用卡验证和其他的销售点终端、电子资金转移，以及许多形式的远程数据库访问生来具有无连接的性质，在一个方向上传送查询，在另一个方向上返回应答。

#### 2. Are there any circumstances when connection-oriented service will (or at least should) deliver packets out of order? Explain.(M)

有。中断信号不遵从顺序的投递，它会跳过在它前面的数据。例如是当一个终端用户键入退出（或 kill）键时，由退出信号产生的分组被立即发送，并且跳过了当前队列中等待程序处理的排在前面任何数据（即已经键入但没被程序读取的数据）。

#### 3. Datagram subnets route each packet as a separate unit, independent of all others. Virtual-circuit subnets do not have to do this, since each data packet follows a predetermined route. Does this observation mean that virtual-circuit subnets do not need the capability to route isolated packets from an arbitrary source to an arbitrary destination? Explain your answer. (E)

不对。为了使分组能从任意源到达任意目的地，连接建立时要选择路由，虚电路网络也需要这一能力。

#### 4. Give three examples of protocol parameters that might be negotiated when a connection is set up. (E)

在连接建立的时候可能要协商窗口的大小、最大分组尺寸和超时值。

5. 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, 15-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? (H)

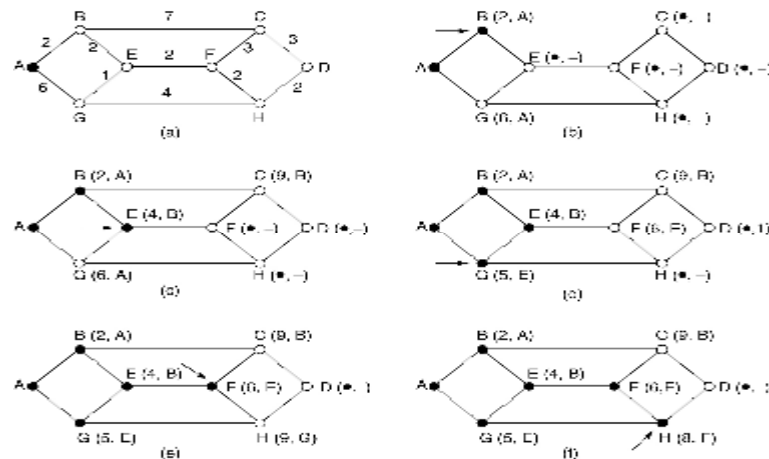
4 跳意味着引入了 5 个路由器。实现虚电路需要在 1000 秒内固定分配  $5 \times 8 = 40$  字节的存储器。实现数据报需要比实现虚电路多传送的头信息的容量等于  $(15 - 3) \times 4 \times 200 = 9600$  字节-跳段。

现在的问题就变成了 40000 字节-秒的存储器对比 9600 字节-跳段的电路容量的开销。如果存储器的使用期为两年，即  $3600 \times 8 \times 5 \times 52 \times 2 = 1.5 \times 10^7$  秒，一个字节-秒的代价为  $1 / (1.5 \times 10^7) = 6.7 \times 10^{-8}$  分，那么 40000 字节-秒的代价为 2.7 毫分。另一方面，1 个字节-跳段代价是  $10^{-6}$  分，9600 个字节-跳段的代价为  $10^{-6} \times 9600 = 9.6 \times 10^{-3}$  分，即 9.6 毫分，即在这 1000 秒内的时间内便宜大约 6.9 毫分。

6. Assuming that all routers and hosts are working properly and that all software in both is free of all errors, is there any chance, however small, that a packet will be delivered to the wrong destination? (E)

有可能。大的突发噪声可能破坏分组。使用  $k$  位的检验和，差错仍然有  $2^{-k}$  的概率被漏检。如果分组的目的地或虚电路号码被改变，分组将会被投递到错误的目的地，并可能被接收为正确的分组。换句话说，偶然的突发噪声可能把送往一个目的地的完全合法的分组改变成送往另一个目的地的也是完全合法的分组。

7. Consider the network of Fig. 5-7, but ignore the weights on the lines. Suppose that it uses flooding as the routing algorithm. If a packet sent by A to D has a maximum hop count of 3, list all the routes it will take. Also tell how many hops worth of bandwidth it consumes. (E)



所有的路由选择如下：ABCD, ABCF, ABEF, ABEG, AGHD, AGHF, AGEH, and AGEF, 所以总跳数为 24

8. Give a simple heuristic for finding two paths through a network from a given source to a given destination that can survive the loss of any communication line (assuming two such paths exist). The routers are considered reliable enough, so it is not necessary to worry about the possibility of router crashes. (E)

使用最短通路搜索算法选择一条路径，然后，删除刚找到的路径中的使用的所有的弧（对应各条链路）。接着，再运行一次最短通路搜索算法。这个第 2 条路径在第 1 条路径中有线路失效的情况下，可以作为替代路径启用；反之亦然。

9. Consider the subnet of Fig. 5-13(a). Distance vector routing is used, and the following vectors have just come in to router C: from B: (5, 0, 8, 12, 6, 2); from D: (16, 12, 6, 0, 9, 10); and from E: (7, 6, 3, 9, 0, 4). The measured delays to B, D, and E, are 6, 3, and 5, respectively. What is C's new routing table? Give both the outgoing line to use and the expected delay. (M)



A, B, C, D, E, F

通过 B 给出 (11, 6, 14, 18, 12, 8)



通过 D 给出 (19, 15, 9, 3, 12, 13)

通过 E 给出 (12, 11, 8, 14, 5, 9)

取到达每一目的地的最小值 (C 除外) 得到: (11, 6, 0, 3, 5, 8)

输出线路是: (B, B, -, D, E, B)

10. If delays are recorded as 8-bit numbers in a 50-router network, and delay vectors are exchanged twice a second, how much bandwidth per (full-duplex) line is chewed up by the distributed routing algorithm? Assume that each router has three lines to other routers. (E)

路由表的长度等于  $8 \times 50 = 400 \text{ bit}$ 。该表每秒钟在每条线路上发送 2 次, 因此  $400 \times 2 = 800 \text{ b/s}$ , 即在每条线路的每个方向上消耗的带宽都是 800 bps。

11. In Fig. 5-14 the Boolean OR of the two sets of ACF bits are 111 in every row. Is this just an accident here, or does it hold for all subnets under all circumstances? (M)

Source	Seq.	Age	Send flags			ACK flags			Data
			A	C	F	A	C	F	
A	21	60	0	1	1	1	0	0	
F	21	60	1	1	0	0	0	1	
E	21	59	0	1	0	1	0	1	
C	20	60	1	0	1	0	1	0	
D	21	59	1	0	0	0	1	1	

这个结论总是成立的。如果一个分组从某条线路上到达, 必须确认包的到达。如果线路上没有分组到达, 它就是在发送确认。情况 00 (没有分组到达并且不发送确认) 和 11 (到达和返回) 逻辑上错误, 因此不存在。

12. For hierarchical routing with 4800 routers, what region and cluster sizes should be chosen to minimize the size of the routing table for a three-layer hierarchy? A good starting place is the hypothesis that a solution with k clusters of k regions of k routers is close to optimal, which means that k is about the cube root of 4800 (around 16). Use trial and error to check out combinations where all three parameters are in the general vicinity of 16.

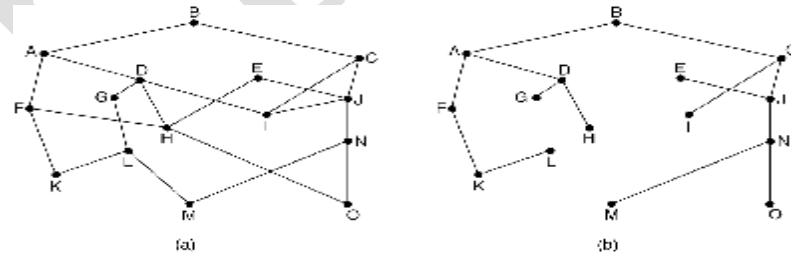
依题可选择 15 个群、16 个区, 每个区 20 个路由器时, 即使得  $4800 = 15 \times 16 \times 20$ , 这时每个路由器需要 20 个表项记录本地路由器, 15 个表项记录用于到同一群内其它区的路由, 14 个表项用于远程的群, 这时路由表尺寸最小为  $20 + 15 + 14$ 。

13. In the text it was stated that when a mobile host is not at home, packets sent to its home LAN are intercepted by its home agent on that LAN. For an IP network on an 802.3 LAN, how does the home agent accomplish this interception? (E)

Conceivably it might go into promiscuous mode, reading all frames dropped onto the LAN, but this is very inefficient. Instead, what is normally done is that the home agent tricks the router into thinking it is the mobile host by responding to ARP requests. When the router gets an IP packet destined for the mobile host, it broadcasts an ARP query asking for the 802.3 MAC-level address of the machine with that IP address. When the mobile host is not around, the home agent responds to the ARP, so the router associates the mobile user's IP address with the home agent's 802.3 MAC-level address.

14. Looking at the subnet of Fig. 5-6, how many packets are generated by a broadcast from B, using

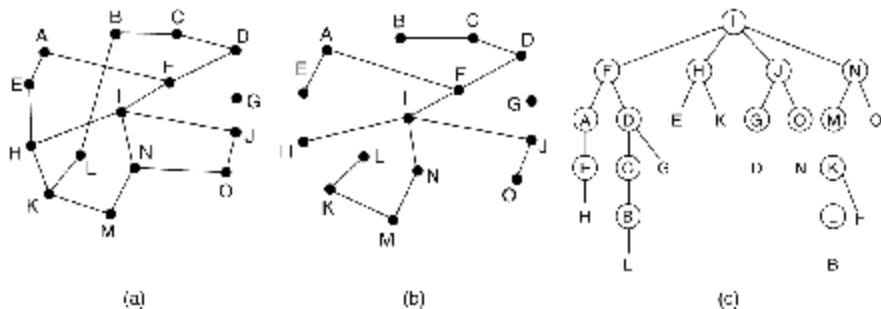
- (a) reverse path forwarding?
- (b) the sink tree? (M)



(1) 反向通路转发算法, 算法进行到 5 个跳段后结束, AC, FDIJ, KHG(D)(J)E(I)N, L(F)(E)(D)O(H)(J)M, (K)(G)(M)(H)(N)(L), 总共产生 28 个分组

(2) 使用汇集树算法, 需要 4 个跳段, AC, FDIJ, KGHEN, LMO, 总共产生 14 个分组。

15. Consider the network of Fig. 5-16(a). Imagine that one new line is added, between F and G, but the sink tree of Fig. 5-16(b) remains unchanged. What changes occur to Fig. 5-16(c)? (M)

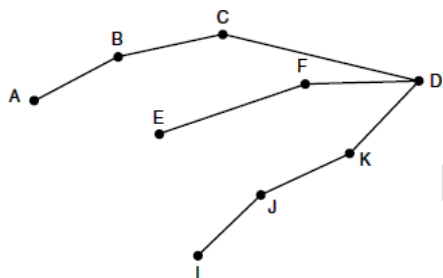


Node *F* currently has two descendants, *A* and *D*. It now acquires a third one, *G*, not circled because the packet that follows *IFG* is not on the sink tree. Node *G* acquires a second descendant, in addition to *D*, labeled *F*. This, too, is not circled as it does not come in on the sink tree.

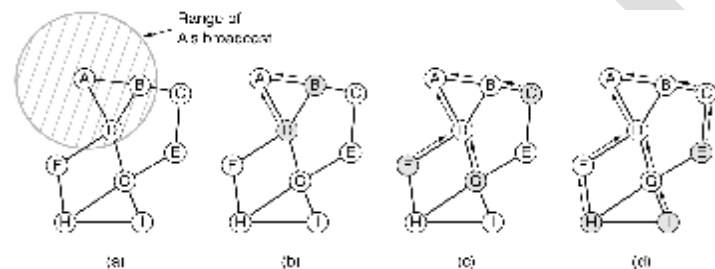
当前的 c 中的结点 F 有两个孩子 A 和 D，依题意不改变汇集树 b 的话，再为 F 加一个孩子 G，不会有环是因为 IFG 之后的分组不在汇集树 b 中。结点 G 下再加一个孩子 D，标记 F。同样，不会有环是因为在汇集树 b 中之后再没有分组进来。

**16. Compute a multicast spanning tree for router C in the following subnet for a group with members at routers A, B, C, D, E, F, I, and K. (E)**

多种生成树是可能的，例如其中一颗为：



**17. In Fig. 5-20, do nodes H or I ever broadcast on the lookup shown starting at A? (E)**



在 d 中，E，H，I 接收到了广播信息之后阴影节点是新的接收节点；箭头显示了可能的逆向路由路径。H 收到分组 A 后，它广播 A；然而，I 知道了如何到达 I，所以 I 不广播收到的分组。

**18. Suppose that node B in Fig. 5-20 has just rebooted and has no routing information in its tables. It suddenly needs a route to H. It sends out broadcasts with TTL set to 1, 2, 3, and so on. How many rounds does it take to find a route? (E)**

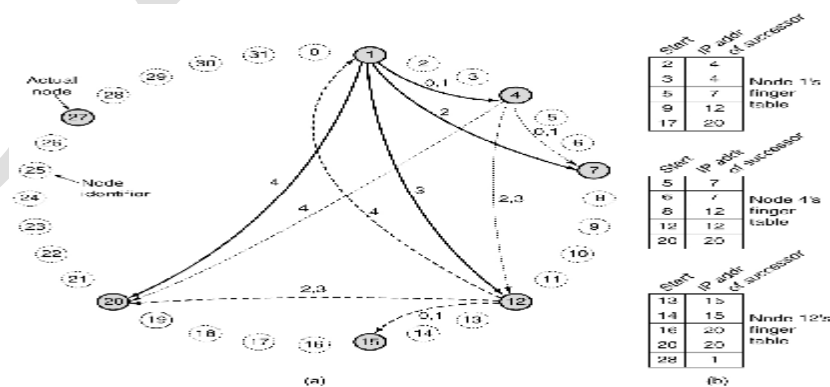
从结点 B 到 H 需要 3 跳，因此要花 3 圈来找到路由线路。

**19. In the simplest version of the Chord algorithm for peer-to-peer lookup, searches do not use the finger table. Instead, they are linear around the circle, in either direction. Can a node accurately predict which direction it should search? Discuss your answer. (E)**

可以大致估计，但不是很精确。假设有 1024 个结点标记，如果结点 300 正在查找结点 800，可能最好去顺时针方向查找，但是也许可能顺时针方向有 20 个真实的结点在结点 300 和结点 800 之间，而逆时针方向只有 16 个真实结点在它们之间。

散列函数 SHA-1 的目的在于生成一个非常流畅的分布使得结点密度在环上基本上是一样的。但是会总有统计学上的波动，因此直接向前的选择可能是错误的。

**20. Consider the Chord circle of Fig. 5-24. Suppose that node 10 suddenly goes on line. Does this affect node 1's finger table, and if so, how? (E)**



在表项 3 中的结点从 12 变 10。

**21. As a possible congestion control mechanism in a subnet using virtual circuits internally, a router could refrain from acknowledging a received packet until (1) it knows its last transmission along the virtual circuit was received successfully and (2) it has a free buffer. For simplicity, assume that the routers use a stop-and-wait protocol and that each virtual circuit has one buffer dedicated to it for each direction of traffic. If it takes *T* sec to transmit a packet (data or acknowledgement)**

and there are  $n$  routers on the path, what is the rate at which packets are delivered to the destination host? Assume that transmission errors are rare and that the host-router connection is infinitely fast. (M)

协议很不好。对时间以  $T$  秒为单位分时段。在时段 1 中, 源路由器发送第一个分组。在时段 2 的开始时第 2 个路由器收到了分组, 但是还没发送确认。在时段 3 的开始时第 3 个路由器收到了分组, 但也不发送确认。这样, 此后所有的路由器都不发送确认。仅当目的地主机从目的地路由器取得分组时才会发送第 1 个确认。现在确认开始往回传播。在源路由器可以发送第 2 个分组之前, 需要两次通过该子网, 所费时间为  $2(n-1)T$  秒/分组, 很显然, 这种协议的效率是很低的。

22. A datagram subnet allows routers to drop packets whenever they need to. The probability of a router discarding a packet is  $p$ . Consider the case of a source host connected to the source router, which is connected to the destination router, and then to the destination host. If either of the routers discards a packet, the source host eventually times out and tries again. If both host-router and router-router lines are counted as hops, what is the mean number of

c. (a) hops a packet makes per transmission?

d. (b) transmissions a packet makes?

e. (c) hops required per received packet?(M)

(1) 由源主机发送的每个分组可能行走 1 个跳段、2 个跳段或 3 个跳段。走 1 个跳段的概率为  $p$ , 走 2 个跳段的概率为  $(1-p)p$ , 走 3 个跳段的概率为  $(1-p)^2$ 。那么, 一个分组平均通路长度的期望值为:  $L=1*p+2*(1-p)p+3*(1-p)^2=p^2-3p+3$

即每次发送一个分组的平均跳段数是  $p^2-3p+3$ 。

(2) 一次发送成功(走完整个通路)的概率为  $(1-p)^2$ , 令  $a=(1-p)^2$ , 两次发射成功的概率等于  $(1-a)a$ , 三次发射成功的概率等于  $(1-a)^2a$ , ..., 因此一个分组平均

发送次数为:  $T = \sum_{i=1}^{\infty} i a (1-a)^{i-1} = \frac{1}{a} = \frac{1}{(1-p)^2}$ , 即一个分组平均要发送  $1/(1-p)^2$  次。

(3) 每一个接收到的分组行走的平均跳段数等于:  $H=L*T=(p^2-3p+3)/(1-p)^2$

23. Describe two major differences between the warning bit method and the RED method. (E)

首先, 警告位方法通过设置一个特殊的位来显示地发送一个拥塞标记给源。而 RED 方法是通过简单地丢掉源分组中的一个来隐式标记。

其次, 警告位方法只有在没有缓冲区空间时才丢掉一个分组, 而 RED 方法是在所有的缓冲区空间被消耗完才丢弃分组。

24. Give an argument why the leaky bucket algorithm should allow just one packet per tick, independent of how large the packet is. (M)

通常计算机能够以很高的速率产生数据, 网络也可以用同样的速率运行。然而, 路由器却只能在短时间内以同样高的速率处理数据。对于排在队列中的一个分组, 不管它有多大, 路由器必须做大约相同分量的工作。显然, 处理 10 个 100 字节长的分组所作的工作比处理 1 个 1000 字节长的分组要做的工作多得多。

25. The byte-counting variant of the leaky bucket algorithm is used in a particular system. The rule is that one 1024-byte packet, or two 512-byte packets, etc., may be sent on each tick. Give a serious restriction of this system that was not mentioned in the text. (E)

不可以发送任何大于 1024 字节的分组。

26. An ATM network uses a token bucket scheme for traffic shaping. A new token is put into the bucket every 5  $\mu$ sec. Each token is good for one cell, which contains 48 bytes of data. What is the maximum sustainable data rate?(E)

每 5 $\mu$ s 产生一个令牌, 1 秒中可以发送 200,000 个信元。每个信元含有 48 个数据字节, 即  $8 \times 48 = 384$  bit。最大的可持续的净数据速率为  $384 \times 2 \times 10^5 = 76.8$  Mb/s

27. A computer on a 6-Mbps network is regulated by a token bucket. The token bucket is filled at a rate of 1 Mbps. It is initially filled to capacity with 8 megabits. How long can the computer transmit at the full 6 Mbps?(E)

由公式  $S=C/(M-P)$ , 这里的  $S$  表示以秒计量的突发时间长度,  $M$  表示以每秒字节计量的最大输出速率,  $C$  表示以字节计的桶的容量,  $P$  表示以每秒字节计量的令牌到达速率。则:  $S=((8*10^6)/8)/((6*10^6)/8 - (1*10^6)/8) = 1.6$  s

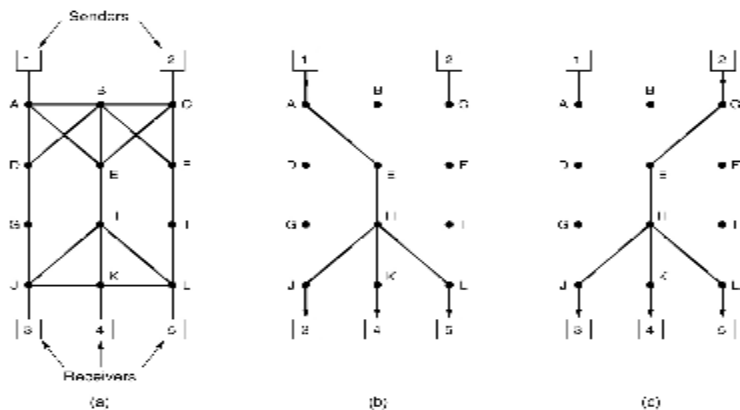
因此, 计算机可以用完全速率 6Mb/s 发送 1.6 s 的时间。

28. Imagine a flow specification that has a maximum packet size of 1000 bytes, a token bucket rate of 10 million bytes/sec, a token bucket size of 1 million bytes, and a maximum transmission rate of 50 million bytes/sec. How long can a burst at maximum speed last? (E)

令最大突发时间长度为  $\Delta t$  秒。在极端情况下, 漏桶在突发期间的开始是充满的 (1MB), 这期间数据流入桶内  $10\Delta t$  MB, 流出包含  $50\Delta t$  MB, 由等式  $1+10\Delta t=50\Delta t$ , 得到  $\Delta t=1/40$ s, 即 25ms。因此, 以最大速率突发传送可维持 25ms 的时间。

29. The network of Fig. 5-37 uses RSVP with multicast trees for hosts 1 and 2 as shown. Suppose that host 3 requests a channel of bandwidth 2 MB/sec for a flow from host 1 and another channel of bandwidth 1 MB/sec for a flow from host 2. At the same time, host 4 requests a channel of bandwidth 2 MB/sec for a flow from host

1 and host 5 requests a channel of bandwidth 1 MB/sec for a flow from host 2. How much total bandwidth will be reserved for these requests at routers A, B, C, E, H, J, K, and L?(E)



带宽流量如下:  $A \rightarrow 2$ ,  $B \rightarrow 0$ ,  $C \rightarrow 1$ ,  $E \rightarrow 3$ ,  $H \rightarrow 3$ ,  $J \rightarrow 3$ ,  $K \rightarrow 2$  和  $L \rightarrow 1$

30. The CPU in a router can process 2 million packets/sec. The load offered to it is 1.5 million packets/sec. If a route from source to destination contains 10 routers, how much time is spent being queued and serviced by the CPUs?(E)

依题知  $\mu = 2$  million,  $\lambda = 1.5$  million, 可知  $\rho = \lambda/\mu = 0.75$ , 从排队理论可知, 每个分组经历的延迟是空系统中的四倍。空系统中的时延是 500 nsec, 这里为  $2\mu\text{sec}$ 。经过 10 个路由器, 排队总时间为  $20\mu\text{sec}$ 。

31. Consider the user of differentiated services with expedited forwarding. Is there a guarantee that expedited packets experience a shorter delay than regular packets? Why or why not? (E)

无法保证, 如果快速的分组太多, 它们分配的带宽可能会比常规的分组的性能更差。

32. Is fragmentation needed in concatenated virtual-circuit internets or only in datagram systems? (E)

都需要分割功能。即使是在一个串接的虚电路网络中, 沿通路的某些网络可能接受 1024 字节分组, 而另一些网络可能仅接受 48 字节分组, 分割功能仍然是需要的。

33. Tunneling through a concatenated virtual-circuit subnet is straightforward: the multiprotocol router at one end just sets up a virtual circuit to the other end and passes packets through it. Can tunneling also be used in datagram subnets? If so, how? (E)

可以。只需把分组封装在属于所经过的子网的数据报的载荷段中, 并进行发送。

34. Suppose that host A is connected to a router R 1, R 1 is connected to another router, R 2, and R 2 is connected to host B. Suppose that a TCP message that contains 900 bytes of data and 20 bytes of TCP header is passed to the IP code at host A for delivery to B. Show the Total length, Identification, DF, MF, and Fragment offset fields of the IP header in each packet transmitted over the three links. Assume that link A-R1 can support a maximum frame size of 1024 bytes including a 14-byte frame header, link R1-R2 can support a maximum frame size of 512 bytes, including an 8-byte frame header, and link R2-B can support a maximum frame size of 512 bytes including a 12-byte frame header. (M)

在 R1 最初的 IP 数据报会被分割成两个 IP 数据报, 以后不会再分割了。

链路 A-R1: Length = 940; ID = x; DF = 0; MF = 0; Offset = 0

链路 R1-R2:

(1) Length = 500; ID = x; DF = 0; MF = 1; Offset = 0

(2) Length = 460; ID = x; DF = 0; MF = 0; Offset = 60

链路 R2-B:

(1) Length = 500; ID = x; DF = 0; MF = 1; Offset = 0

(2) Length = 460; ID = x; DF = 0; MF = 0; Offset = 60

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

如果线路的比特率为 b, 则每秒钟分组数量为  $b/8192$ , 那么发送一个分组所需的时间为  $8192/b$ ; 输出 65,536 个分组要花费  $2^{29}/b \text{ sec}$ , 依题分组的生存期为 10s, 将  $2^{29}/b = 10$ , 可得 b 为 53,687,091 bps

40. A large number of consecutive IP address are available starting at 198.16.0.0. Suppose that four organizations, A, B, C, and D, request 4000, 2000, 4000, and 8000 addresses, respectively, and in that order. For each of these, give the first IP address assigned, the last IP address assigned, and the mask in the w.x.y.z/s notation. (M)

A:  $4000 \rightarrow 2^{12}$ ; B:  $2000 \rightarrow 2^{11}$ ; C:  $4000 \rightarrow 2^{12}$ ; D:  $8000 \rightarrow 2^{13}$ ;

始地址, 尾地址, 和子网掩码如下:

A: 198.16.0.0 – 198.16.15.255 子网写作 198.16.0.0/20

B: 198.16.16.0 – 198.16.23.255 子网写作 198.16.16.0/21

C: 198.16.32.0 – 198.16.47.255 子网写作 198.16.32.0/20



D: 198.16.64.0 – 198.16.95.255 子网写作 198.16.64.0/19

46. ARP and RARP both map addresses from one space to another. In this respect, they are similar. However, their implementations are fundamentally different. In what major way do they differ? (E)

在 RARP 的实现中有一个 RARP 服务器 负责回答查询请求。

在 ARP 的实现中没有这样的服务器，主机自己回答 ARP 查询。

52. The Protocol field used in the IPv4 header is not present in the fixed IPv6 header. Why not? (E)

设置协议段的目的是要告诉目的地主机把 IP 分组交给那一个协议处理程序。中途的路由器并不需要这一信息，因此不必把它放在主头中。实际上，这个信息存在于头中，但被伪装了。最后一个（扩展）头的下一个字段就用于这一目的。

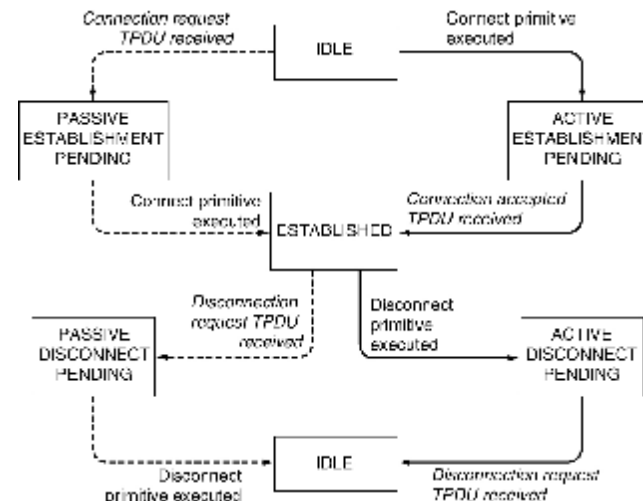
## Chapter 6 The Transport Layer Problems

1. In our example transport primitives of Fig. 6-2, LISTEN is a blocking call. Is this strictly necessary? If not, explain how a nonblocking primitive could be used. What advantage would this have over the scheme described in the text?(E)

Primitive	Packet sent	Meaning
LISTEN	(none)	Block until some process tries to connect
CONNECT	CONNECTION REQ.	Actively attempt to establish a connection
SEND	DATA	Send information
RECEIVE	(none)	Block until a DATA packet arrives
DISCONNECT	DISCONNECTION REQ.	This side wants to release the connection

不是。事实上，LISTEN 调用可以表明建立新连接的意愿，但不封锁。当有了建立连接的尝试时，调用程序可以被提供一个信号。然后，它执行，比如说，OK 或 REJECT 来接受或拒绝连接。然而，在原先的封锁性方案中，就缺乏这种灵活性。

2. In the model underlying Fig. 6-4, it is assumed that packets may be lost by the network layer and thus must be individually acknowledged. Suppose that the network layer is 100 percent reliable and never loses packets. What changes, if any, are needed to Fig. 6-4?(E)



从“被动连接建立在进行中”到“已建立”的虚线不再依确认的传输情况而定。该变迁可立即发生。实质上，“被动连接建立在进行中”状态已经消失，因为它们什么时候都不可见。

3. In both parts of Fig. 6-6, there is a comment that the value of SERVER\_PORT must be the same in both client and server. Why is this so important? (E)

如果客户机发送一个分组给 SERVER\_PORT 并且服务器当时并没有侦听这一端口，那么这个分组将不会被投递给服务器。

4. Suppose that the clock-driven scheme for generating initial sequence numbers is used with a 15-bit wide clock counter. The clock ticks once every 100 msec, and the maximum packet lifetime is 60 sec. How often need resynchronization take place

a. (a) in the worst case?

b. (b) when the data consumes 240 sequence numbers/min? (M)

时钟驱动方案的基本思想是同一时间不会有两个活动的 TPDUs 使用相同的序列号。序列号空间应该足够大，使得当编号循环一周时，具有相同号码的旧的 TPDUs 已经不复存在。

(a) 时钟大循环周期是  $2^{15}$ ，即 32768 滴答，每滴答 100ms，即 0.1 秒，所以大循环周期是 3276.8s。假定数据产生速率非常低（接近零），那么发送方在

3276.8-60=3271.8 秒时进入禁止区，需要进行一次重新同步。

(b) 每分钟使用 240 个序列号，即每秒使用 4 个号码，如果时间以  $t$  表示（以秒为单位），那么实际使用中的序列号是  $4t$  个。当接近大循环的末尾时以及在下一大循环的开始阶段， $4t$  有一定的大小，位于禁止区的上方，现在由于每秒钟 10 个滴答，禁止区的左边是  $10(t-3216.8)$ 。令  $4t=10(t-3216.8)$ ，得  $t=5316.3$  秒。即当  $t=5316.3$  时，开始进入禁止区，需要进行一次重新同步。

**5. Why does the maximum packet lifetime,  $T$ , have to be large enough to ensure that not only the packet but also its acknowledgements have vanished? (M)**

首先看三次握手过程是如何解决延迟的重复到达的分组所引起的问题的。

正常情况下，当主机 1 发出连接请求时，主机 1 选择一个序号  $x$ ，并向主机 2 发送一个包含该序号的请求 TPDU；接着，主机 2 回应一个接受连接的 TPDU，确认  $x$ ，并声明自己所选用的初始序列号  $y$ ；最后，主机 1 在其发送的第一个数据 TPDU 中确认主机 2 所选择的初始序列号。

当出现延迟的重复的控制 TPDU 时，一个 TPDU 是来自于一个已经释放的连接延迟重复的连接请求（CONNECTION REQUEST），该 TPDU 在主机 1 毫不知情的情况下到达主机 2。

主机 2 通过向主机 1 发送一个接受连接的 TPDU（CONNECTION ACCEPTED）来响应该 TPDU，而该接受连接的 TPDU 的真正目的是证实主机 1 确实试图建立一个新的连接。在这一点上，关键在于主机 2 建议使用  $y$  作为从主机 2 到主机 1 交通的初始序列号，从而说明已经不存在包含序列号为  $y$  的 TPDU，也不存在对  $y$  的应答分组。当第二个延迟的 TPDU 到达主机 2 时， $z$  被确认而不是  $y$  被确认的事实告诉主机 2 这是一个旧的重复的 TPDU，因此废止该连接过程。在这里，三次握手协议是成功的。

最坏的情况是延迟的“连接请求”和对“连接被接收”的确认应答都在网络上存活。可以设想，当第 2 个重复分组到达时，如果在网上还存在一个老的对序列号为  $y$  的分组的确认应答，显然会破坏三次握手协议的正常工作，故障性的产生一条没有人真正需要的连接，从而导致灾难性的后果。

**6. Imagine that a two-way handshake rather than a three-way handshake were used to set up connections. In other words, the third message was not required. Are deadlocks now possible? Give an example or show that none exist. (M)**

我们知道，3 次握手完成两个重要功能，既要双方做好发送数据的准备工作（双方都知道彼此已准备好），也要允许双方就初始序列号进行协商，这个序列号在握手过程中被发送与确认。

现在把三次握手改成仅需要两次握手，死锁是可能发生的。例如，考虑计算机 A 和 B 之间的通信。假定 B 给 A 发送一个连接请求分组，A 收到了这个分组，并

发送了确认应答分组。按照两次握手的协定，A 认为连接已经成功的建立了，可以开始发送数据分组。

可是，B 在 A 的应答分组在传输中被丢失的情况下，将不知道 A 是否已经准备好，不知道 A 建议什么样的序列号用于 A 到 B 的交通，也不知道 A 是否同意 A 所建议的用于 B 到 A 交通的初始序列号，B 甚至怀疑 A 是否收到自己的连接请求分组。在这种情况下，B 认为连接还未建立成功，将忽略 A 发来的任何数据分组，只等待接收连接确认应答分组。而 A 在发出的分组超时后，重复发送同样的分组。这样就形成了死锁。

**8. Consider the problem of recovering from host crashes (i.e., Fig. 6-18). If the interval between writing and sending an acknowledgement, or vice versa, can be made relatively small, what are the two best sender-receiver strategies for minimizing the chance of a protocol failure? (E)**

Strategy used by sending host	Strategy used by receiving host					
	First ACK, then write			First write, then ACK		
	AC(W)	AWC	C(AW)	C(WA)	W AC	WC(A)
Always retransmit	OK	DUP	OK	OK	DUP	DUP
Never retransmit	LOST	OK	LOST	LOST	OK	OK
Retransmit in S0	OK	DUP	LOST	LOST	DUP	OK
Retransmit in S1	LOST	OK	OK	OK	OK	DUP

OK = Protocol functions correctly  
DUP = Protocol generates a duplicate message  
LOST = Protocol loses a message

AW（先确认后写）WA（先写后确认）C（崩溃）

如果 AW 或 WA 间隔时间很短，事件 AC（W）和 W（CA）就不太可能发生。此时的最好发送方策略是，如果崩溃恢复时处于状态 S1，应该重传最后一个 TPDU，接收方采用顺序 AW 或 WA 则无关紧要。

**9. Are deadlocks possible with the transport entity described in the text (Fig. 6-20)? (E)**

该传输实体有可能死锁。当双方同时执行 RECEIVE 时就会进入死锁状态。

**10. Out of curiosity, the implementer of the transport entity of Fig. 6-20 has decided to put counters inside the sleep procedure to collect statistics about the conn array. Among these are the number of connections in each of the seven possible states,  $n_i$  ( $i = 1, \dots, 7$ ). After writing a massive FORTRAN program to analyze the data, our implementer discovers that the relation  $n_i = \text{MAX\_CONN}$  appears to**

always be true. Are there any other invariants involving only these seven variables? (E)

有,  $n_2+n_3+n_6+n_7=1$  .

因为状态 listening ( $n_2$ )、waiting ( $n_3$ )、sending ( $n_6$ ) 和 receiving ( $n_7$ ) 都意味着用户被封锁, 因此当处在其中的一个状态时, 就不可能是在另一个状态。

答案中出现了一个 listen 状态, 但在实际的状态转换图中没有该状态, 这里对应的是调用 listen 函数并 sleep() 的情况。

11. What happens when the user of the transport entity given in Fig. 6-20 sends a zero-length message? Discuss the significance of your answer. (E)

长度为零的报文被另一边接收。这种报文的发送可以被用来表示文件结束的信号。

12. For each event that can potentially occur in the transport entity of Fig. 6-20, tell whether it is legal when the user is sleeping in sending state. (M)

因为文件处于封锁状态, 所有的传输层原语都不可能执行。因此, 仅分组到达事件是可能的, 而且还不是所有的到达事件。事实上, 仅仅跟呼叫请求、清除请求、数据分组和信用量分组这几个分组到达有关的事件是合法的。

原来的答案一共有四种合法的状态, 但实际上 CallReq 和 DataPkt 是不合法的, 应该只有 ClearReq 和 Credit 两种可能的情况。

13. Discuss the advantages and disadvantages of credits versus sliding window protocols. (M)

滑动窗口协议比较简单, 仅需要管理窗口边缘一组参数, 而且, 对于到达顺序有错的 TPDU 不会引起窗口增加和减少方面的问题。而信用量方案比较灵活, 允许独立于确认, 动态的管理缓冲区。

14. Why does UDP exist? Would it not have been enough to just let user processes send raw IP packets? (E)

仅仅使用 IP 分组还不够。IP 分组包含 IP 地址, 该地址指定一个目的地机器。一旦这样的分组到达了目的地机器, 网络控制程序如何知道该把它交给哪个进程呢? UDP 分组包含一个目的地端口, 这一信息是必须的, 因为有了它, 分组才能够被投递给正确的进程。

15. Consider a simple application-level protocol built on top of UDP that allows a client to retrieve a file from a remote server residing at a well-known address. The client first sends a request with file name, and the server responds with a sequence of data packets containing different parts of the requested file. To ensure reliability and sequenced delivery, client and server use a stop-and-wait protocol. Ignoring the

obvious performance issue, do you see a problem with this protocol? Think carefully about the possibility of processes crashing. (E)

有可能客户得到的是错误的文件。假设客户 A 发送一个请求  $f1$  并且这时当掉了。只有一个客户 B 使用着同样的协议去请求另一个文件  $f2$ , 假设客户 B 与 A 运行在同一台机器上 (有着相同的 IP 地址), 向 A 之前一正在用的同一端口绑定它的 UDP 端口号。另外, 假设 B 的请求丢失了。当前面的 A 的回复到达时, B 却收到了它并把它当成是自己请求的回答。

16. A client sends a 128-byte request to a server located 100 km away over a 1-gigabit optical fiber. What is the efficiency of the line during the remote procedure call? (E)

128 字节等于 1024 位, 在 1Gb/s 的线路上发送 1000 位需要  $1\mu s$  的时间。光在光导纤维中的传播速度是 200km/ms, 请求到达服务器需要传输 0.5ms 的时间, 应答返回又需要 0.5ms 的传输时间。总的看来, 1000 位在 1ms 的时间内传输完成。这等效于占用带宽 1Mb/s, 即线路效率是 0.1%。

17. Consider the situation of the previous problem again. Compute the minimum possible response time both for the given 1-Gbps line and for a 1-Mbps line. What conclusion can you draw? (E)

在 1Gb/s, 响应时间由光的速度决定。可以取得的最好情况是 1ms。在 1Mb/s, 发射 1024 位需要大约 1ms 的时间, 再经过 0.5ms 最后一位到达服务器, 还需要另外 0.5ms 应答才能返回, 这是最好的情况。因此, 最好的 RPC 时间是 2ms。结论是, 线路速度改善到 1000 倍, 性能仅改善到 2 倍。对于这种应用, 除非千兆位线路特别便宜, 否则是不值得拥有的。

18. Both UDP and TCP use port numbers to identify the destination entity when delivering a message. Give two reasons for why these protocols invented a new abstract ID (port numbers), instead of using process IDs, which already existed when these protocols were designed. (E)

主要原因有三点: 首次, 进程 ID 是与操作系统层的。使用进程 ID 的话将会使协议依赖于操作系统。其次, 一个进程可能建立多个通信信道。如果一个进程 ID 作为目的地标识符将不能够区分出使用哪条通信信道。最后, 让进程侦听已知的端口容易, 但是侦听已知的进程 ID 是不可能的。

19. What is the total size of the minimum TCP MTU, including TCP and IP overhead but not including data link layer overhead? (E)

默认段长是 536 bytes, TCP 和 IP 头部各 20 bytes, 所以总共 576 bytes。

20. Datagram fragmentation and reassembly are handled by IP and are invisible to TCP. Does this mean that TCP does not have to worry about data arriving in the wrong order? (E)

不能这样认为。尽管到达的每个数据报都是完整的，但可能到达的数据报的顺序是错误的，因此，TCP 必须准备适当的重组报文的各个部分。

21. RTP is used to transmit CD-quality audio, which makes a pair of 16-bit samples 44,100 times/sec, one sample for each of the stereo channels. How many packets per second must RTP transmit? (M)

每份采样占  $16 \times 2 \text{ bits} = 4 \text{ bytes}$ ，默认 RTP 报文为 1024 字节，刚每个分组有 256 份采样。 $(44,100 \text{ samples/s}) / (256 \text{ samples/packet}) = 172$  个分组。

默认 RTP 报文为 1024 字节，但 RFC 中并没有此限制，具体解释有待商议。

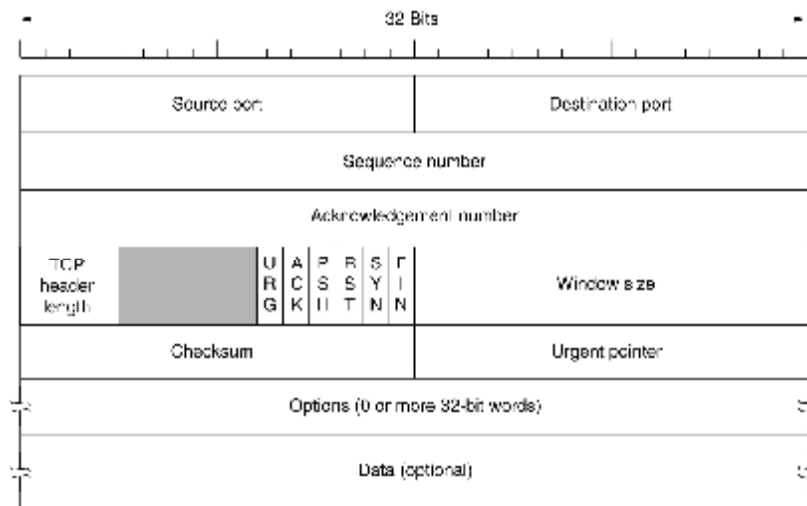
22. Would it be possible to place the RTP code in the operating system kernel, along with the UDP code? Explain your answer. (E)

当然可以。只要调用程序能够提供所需要的信息就可以，但是这么做并没有什么好的理由。

23. A process on host 1 has been assigned port p, and a process on host 2 has been assigned port q. Is it possible for there to be two or more TCP connections between these two ports at the same time?

不可以。一条连接仅仅用它的套接口标识。因此， $(1, p) - (2, q)$  是在这两个端口之间唯一可能的连接。

24. In Fig. 6-29 we saw that in addition to the 32-bit Acknowledgement field, there is an ACK bit in the fourth word. Does this really add anything? Why or why not? (E)

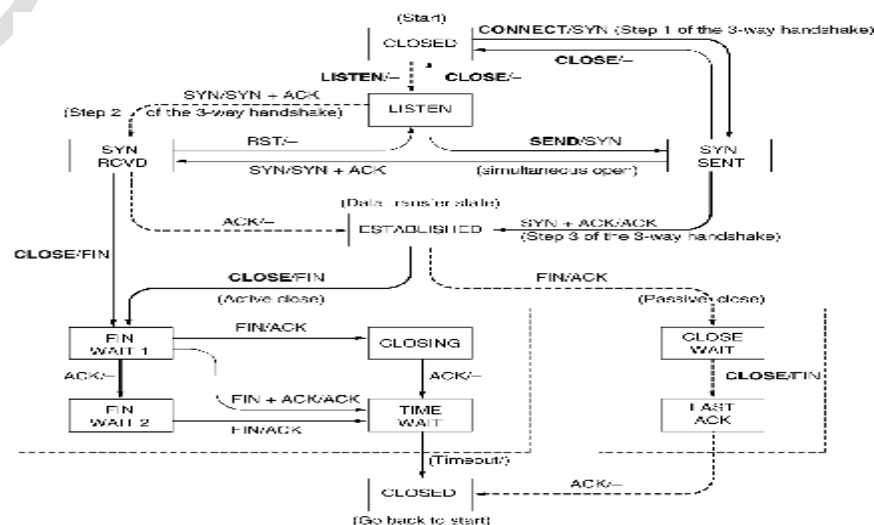


ACK 用于说明 32 位确认域是否被使用。如果没有 ACK，32 位确认域将会一直被使用，如果有必要确认一个已经确认的字符。对于通常的数据传输并不是绝对必要的，但是它用在三次握手协议的第 2 和第 3 段报文中，这里它在建立连接时起着关键作用。

25. The maximum payload of a TCP segment is 65,495 bytes. Why was such a strange number chosen? (E)

整个 TCP 报文段必须适配 IP 分组 65,515 字节的载荷段。因为 TCP 头最少 20 个字节，所以只剩下 65,495 字节用于 TCP 数据。

26. Describe two ways to get into the SYN RCVD state of Fig. 6-33. (E)





一条途径是从 LISTEN 开始。如果收到一个 SYN，那么协议进入 SYN RECD 状态。

另一条途径是一个进程试图做一个主动打开操作，并发送一个 SYN。如果另一方也做打开操作，并收到一个 SYN，那么也将进入 SYN RECD 状态。

**27. Give a potential disadvantage when Nagle's algorithm is used on a badly-congested network. (E)**

Nagle 算法建议，当数据一次一个字节的来到发送方时，只发送第一个字节，并且缓冲所有其他内容，直到所发出的字节被确认为止。然后在一个 TCP 报文段中发送所有缓冲的字符。接着又开始缓冲，直到前一个报文段中的所有字节又被确认。这样，如果用户键入的速度足够快，而网络比较慢的话，那么在每个报文段中都可以有相当数量的字符。该算法还允许输入足够的数据以填满半个窗口或一个最大报文段的情况下发送一个新的分组。在这种运行方式下，尽管用户是以均匀的速度键入，而字符却是以突发的方式回印。用户可能敲击了好几个键，而屏面上什么都没有显示，然后突然的在屏幕上显示出所有已键入的字符。人们可能对此感到恼火。

**28. Consider the effect of using slow start on a line with a 10-msec round-trip time and no congestion. The receive window is 24 KB and the maximum segment size is 2 KB. How long does it take before the first full window can be sent? (E)**

按照慢启动算法，经过 10、20、30、40ms 后拥塞窗口大小分别为 4、8、16、32，所以在 40ms 后将按照  $\min\{24, 32\}=24\text{KB}$  发送数据。

**29. Suppose that the TCP congestion window is set to 18 KB and a timeout occurs. How big will the window be if the next four transmission bursts are all successful? Assume that the maximum segment size is 1 KB. (E)**

由于发生了超时，下一次传输将是 1 个最大报文段，然后是 2 个、4 个、8 个最大报文段，所以在 4 次突发量传输后，拥塞窗口将是 8K 字节

最终窗口大小应该为 9KB。当第一次发送成功后，1KB->2KB，第二次成功后，2KB->4KB，第三次成功后，4KB->8KB，第四次成功后，窗口大小 8KB->9KB

**30. If the TCP round-trip time, RTT, is currently 30 msec and the following acknowledgements come in after 26, 32, and 24 msec, respectively, what is the new RTT estimate using the Jacobson algorithm? Use  $a = 0.9$ . (E)**

RTT：是当前到达目的地的最佳估计值；M：应答花了多长时间。

更新 RTT 值公式： $RTT = a RTT + (1 - a) M$

现在， $a=0.9$ ， $RTT=30\text{ms}$ ， $M_1=26$ ， $M_2=32$ ， $M_3=24$ ，

因此，新的 RTT 估算值分别是 29.6ms、29.84ms、29.256ms。

**31. A TCP machine is sending full windows of 65,535 bytes over a 1-Gbps channel that has a 10-msec one-way delay. What is the maximum throughput achievable? What is the line efficiency? (E)**

每  $10\text{ms} \times 2 = 20\text{ms}$  可以发送一个窗口大小的交通量，因此每秒 50 个窗口。

$65536 \times 8 \times 50 = 26.2\text{Mb/s}$ ； $26.2/1000 = 2.6\%$

所以，最大的数据吞吐率为 26.2Mb/s，线路效率为 2.6%。

**32. What is the fastest line speed at which a host can blast out 1500-byte TCP payloads with a 120-sec maximum packet lifetime without having the sequence numbers wrap around? Take TCP, IP, and Ethernet overhead into consideration. Assume that Ethernet frames may be sent continuously. (H)**

目标是在 120sec 内发送  $2^{32}$  字节的数据，即 35,791,394 bytes/sec

由每帧发送 1500-byte 的数据，总有  $35,791,394 / 1500 = 23,860$  frames/sec

TCP 头部 20 字节，IP 头部 20 字节，以太头部 26 字节。即发送 1500-byte 的数据，每帧需发送 1566-byte 的信息。所以发送所需的总的带宽为  $1500\text{byte/frame} \times 23,860 \text{ frames/sec} \times 8 = 299 \text{ Mbps}$ 。如果发送速率再比这快，那么我们将会有冒着在两个不同的 TCP 段中有着相同的序列号的风险。

**33. In a network that has a maximum TPDU size of 128 bytes, a maximum TPDU lifetime of 30 sec, and an 8-bit sequence number, what is the maximum data rate per connection? (M)**

具有相同编号的 TPDU 不应该同时在网络中传输，因此必须保证当序列号循环回来重复使用的时候，具有相同序列号的 TPDU 已经从网络中消失。现在存活时间是 30 秒，那么在 30 秒的时间内发送方发送的 TPDU 的数目不能多于 255 个。

$255 \times 128 \times 8 / 30 = 8738\text{b/s}$

所以，每条连接的最大数据速率是 8738b/s。

**34. Suppose that you are measuring the time to receive a TPDU. When an interrupt occurs, you read out the system clock in milliseconds. When the TPDU is fully processed, you read out the clock again. You measure 0 msec 270,000 times and 1 msec 730,000 times. How long does it take to receive a TPDU?(E)**

计算平均值： $(270000 \times 0 + 730000 \times 1) / (270000 + 730000) = 0.73 \text{ ms}$

因此，接收一个 TPDU 花 730 微秒的时间。

**35. A CPU executes instructions at the rate of 1000 MIPS. Data can be copied 64 bits at a time, with each word copied costing 10 instructions. If an coming packet has to be copied four times, can this system handle a 1-Gbps line? For simplicity,**

assume that all instructions, even those instructions that read or write memory, run at the full 1000-MIPS rate. (M)

拷贝 64bits = 8 bytes = 4 word 要用  $4 \times 10 = 40$  条指令, 40 条指令花 40ns,

因此每个字节需要  $40/8 \times 4 = 20$ ns 的 CPU 时间来复制处理。系统的处理能力  $1000/20 = 50$ MB/s = 400Mbps, 这远远小于 1Gb/s 的处理需求, 所以如果没有其它的限, 这个系统能够在 1Gb/s 的线路上处理。

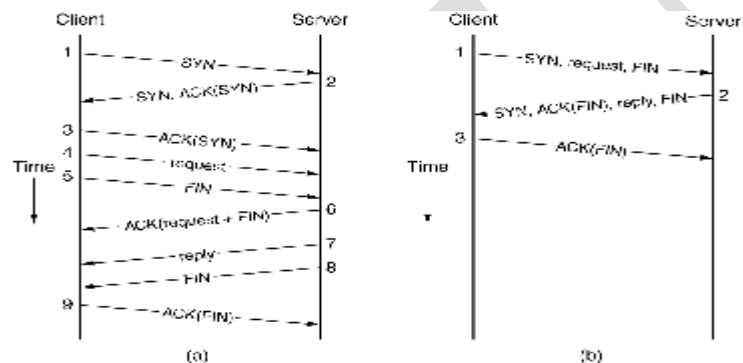
36. To get around the problem of sequence numbers wrapping around while old packets still exist, one could use 64-bit sequence numbers. However, theoretically, an optical fiber can run at 75 Tbps. What maximum packet lifetime is required to make sure that future 75 Tbps networks do not have wraparound problems even with 64-bit sequence numbers? Assume that each byte has its own sequence number, as TCP does. (E)

顺序号的总个数是  $2^{64}$  个字节, 约为  $2 \times 10^{19}$  字节; 发送器每秒钟消耗顺序号个数为:  $75 / 8 = 9.375 \times 10^{12}$  bytes/sec;  $(2 \times 10^{19}) / (9.375 \times 10^{12}) \approx 2 \times 10^6$ , 所以顺序号循环一周所花的时间为  $2 \times 10^6$  s, 约为 23 天。因此, 最长的分组生命周期小于 3 个星期可以避免顺序号循环重复的问题。

37. Give one advantage of RPC on UDP over transactional TCP. Give one advantage of T/TCP over RPC. (M)

在 UDP 上的 RPC (远程过程调用) 只使用了两个分组而事务型 TCP 使用了三个分组。然而如果应答在一个分组中不适用那么 RPC 会有问题存在。

38. In Fig. 6-40(a), we see that it takes 9 packets to complete the RPC. Are there any circumstances in which it takes exactly 10 packets? (E)



有。分组 6 中包含了对 request 的 FIN 的确认。如果每一个被单独确认的话, 那么我们的分组序列就会有 10 个。或者, 分组 9 中确认了 reply 和 FIN, 对它们每一个也可以单独去确认。所以这里有 9 个分组只是碰巧而已。

39. In Sec. 6.6.5, we calculated that a gigabit line dumps 80,000 packets/sec on the host, giving it only 6250 instructions to process it and leaving half the CPU time for applications. This calculation assumed a 1500-byte packet. Redo the calculation for an ARPANET-sized packet (128 bytes). In both cases, assume that the packet sizes given include all overhead. (E)

$$1 / (10^9 / (128 \times 8)) = 10^{-6} \text{ s}$$

1μs 可以处理完一个分组。考虑一半的 CPU 时间, 要求 0.5μs 处理一个分组。在 0.5μs 内 100MIPS 的计算机可以执行 50 条指令。

所以对于每个 ARPANET 分组, 计算机可以执行  $50 \times (1500/128) = 585$  条指令; 每秒可以得到的指令数为  $6250 / (1500/128) = 533$  条

40. For a 1-Gbps network operating over 4000 km, the delay is the limiting factor, not the bandwidth. Consider a MAN with the average source and destination 20 km apart. At what data rate does the round-trip delay due to the speed of light equal the transmission delay for a 1-KB packet? (E)

光在光纤和铜导线中的速度大约为每毫秒 200km。对于一条 20km 的线路, 单向延迟是 100μs, 往返延迟是 200μs。1K 字节就是 8192 位。如果发送 8192 位的时间为 200μs, 那么发送延迟就等于传播延迟。

所以, 数据传输速率为  $8192 / (2 \times 10^{-4}) = 40$  Mb/s。

41. Calculate the bandwidth-delay product for the following networks: (1) T1 (1.5 Mbps), (2) Ethernet (10 Mbps), (3) T3 (45 Mbps), and (4) STS-3 (155 Mbps).

Assume an RTT of 100 msec. Recall that a TCP header has 16 bits reserved for Window Size. What are its implications in light of your calculations? (M)

由题易算出延迟时间为 12.5 ms

带宽延迟的乘积计算结果为: (1) 18.75 KB, (2) 125 KB, (3) 562.5 KB, (4) 1.937 MB;

一个 16-bit 的窗口尺寸意味着一个发送者至多能够发送  $2^{16} = 64$ KB, 之后就等待确认。这意味着如果网络技术使用的是以太网, T3 或 STS-3, 则一个发送者不能使用 TCP 持续传输并且保持管道是充满的状态。

42. What is the bandwidth-delay product for a 50-Mbps channel on a geostationary satellite? If the packets are all 1500 bytes (including overhead), how big should the window be in packets?

同步卫星的往返延迟大约 540 msec, 所以一个 50 Mbps 带宽的带宽延迟乘积是  $27 \text{ Mb} = 3.375 \text{ Mbytes}$ 。若每个分组 1500 bytes, 则  $3.375 \text{ Mbytes} / 1500 = 2250$  个分组就可以充满管道。所以窗口应该至少容纳 2250 个分组。

## Chapter 7 The Application Layer Problems

**5. DNS uses UDP instead of TCP. If a DNS packet is lost, there is no automatic recovery. Does this cause a problem, and if so, how is it solved? (E)**

DNS is idempotent. Operations can be repeated without harm. 请求 DNS 时会同时角发计时器, 超时后会重新发出请求, 这样处理不会有危险。

**6. In addition to being subject to loss, UDP packets have a maximum length, potentially as low as 576 bytes. What happens when a DNS name to be looked up exceeds this length? Can it be sent in two packets? (E)**

这种情况不会出现。DNS 名必须短于 256 bytes, 标准是这样。因此, 所有的 DNS 名都可以放在一个最小长度的分组中。

**7. Can a machine with a single DNS name have multiple IP addresses? How could this occur? (E)**

可能。IP 地址由网络号和主机号组成。如果一台机器有有两块以太网卡, 那么它可以在两个相互独立的网络中, 这时它就需要两个 IP 地址了。

**8. Can a computer have two DNS names that fall in different top-level domains? If so, give a plausible example. If not, explain why not. (E)**

可能。比如 [www.large-bank.com](http://www.large-bank.com) 与 [www.large-bank.ny.us](http://www.large-bank.ny.us) 两个网址就可能有相同的 IP 地址。这样通过 *com* 和国家域都可以访问。

**9. The number of companies with a Web site has grown explosively in recent years. As a result, thousands of companies are registered in the com domain, causing a heavy load on the top-level server for this domain. Suggest a way to alleviate this problem without changing the naming scheme (i.e., without introducing new top-level domain names). It is permitted that your solution requires changes to the client code. (M)**

可以有几种途径来对应这一问题。一种方案就是把 top-level 服务器转换成服务器群。另一种主方案就是使用 26 个独立的服务器, 依次以 a, b, ... 来命名; 每过一段时间后 (比如 3 年) 引入新的服务器, 旧的服务器继续工作以给人们提供机会来适应他们的软件。

**10. Some e-mail systems support a header field Content Return:. It specifies whether the body of a message is to be returned in the event of nondelivery. Does this field belong to the envelope or to the header? (E)**

属于信封。因为发送系统需要知道它的值来处理那些还没被发送的邮件。

**12. A person's e-mail address is his or her login name @ the name of a DNS domain with an MX record. Login names can be first names, last names, initials,**

**and all kinds of other names. Suppose that a large company decided too much e-mail was getting lost because people did not know the login name of the recipient. Is there a way for them to fix this problem without changing DNS? If so, give a proposal and explain how it works. If not, explain why it is impossible. (E)**

可以做到, 并且相对比较容易。当有邮件到达时, SMTP 守护进程接受邮件并在 *RCPT TO* 中查找登陆名。这些名字位于某个文件或数据库中。在这个文件里可以通过别名 (如 Johnson) 的方式追加指向某人的邮箱, 这样邮件就能够使用人们的真实名字进行发送了。

**13. A binary file is 3072 bytes long. How long will it be if encoded using base64 encoding, with a CR+LF pair inserted after every 80 bytes sent and at the end? (E)**

base 64 编码先将消息分解成每 3 个字节一组的 1024 个单元。再将每组编码为 4 个字节, 这样总共有 4096 个字节。把这些字节以 80 个字节每组分行, 可以得到 52 行, 再加上 52 个 CR 和 52 个 LF (每个 CR/CF 都占一个字节), 这样得到的总长度为  $4096+52+52=4200$  字节。

**17. Suppose that someone sets up a vacation daemon and then sends a message just before logging out. Unfortunately, the recipient has been on vacation for a week and also has a vacation daemon in place. What happens next? Will canned replies go back and forth until somebody returns? (E)**

双方都会向对方发送一条封装回复消息, 然后条自机器都会记录对方的邮件地址, 如果发现已经对其进行了回复, 则新的封装回复消息不会被再发送。

**18. In any standard, such as RFC 822, a precise grammar of what is allowed is needed so that different implementations can interwork. Even simple items have to be defined carefully. The SMTP headers allow white space between the tokens. Give two plausible alternative definitions of white space between tokens. (M)**

任意序列拥有一个或多个空白或标记符;

任意序列拥有一个或多个空白或标记符或回退符。

**19. Is the vacation daemon part of the user agent or the message transfer agent? Of course, it is set up using the user agent, but does the user agent actually send the replies? Explain your answer. (E)**

真正的回应是由消息传输代理做出的。当一个 SMTP 连接到来时, 消息传输代理必须检查守护进程是否启动来对到来的邮件进行反应; 如果启动了, 发送一个应达。用户传输代理不能这样处理, 因为它只有在用户从假期回来时才被调用。

**20. POP3 allows users to fetch and download e-mail from a remote mailbox. Does this mean that the internal format of mailboxes has to be standardized so any POP3**



program on the client side can read the mailbox on any mail server? Discuss your answer. (E)

不必标准化。POP3 程序并不是真正的与远程邮箱接触。它发送命令给邮件服务器上的 POP3 守护进程。只要那个守护进程理解邮件格式，POP3 程序就能工作。因此，一个邮件服务器可以从一种格式改变到另一种格式而不用告诉给客户，只要它同时在它的 POP3 守护进程做相应改变使其能理解新格式即可。

**21. From an ISP's point of view, POP3 and IMAP differ in an important way. POP3 users generally empty their mailboxes every day. IMAP users keep their mail on the server indefinitely. Imagine that you were called in to advise an ISP on which protocol it should support. What considerations would you bring up? (E)**

存储用户邮件要占用磁盘空间，这会产生费用。正因为如此，所以产生了使用 POP3 的争论。另外，ISP 可以支付少量的磁盘存储费用，因此邮件转变为收费的。最后对 IMAP 的争论在于其鼓励用户把邮件保存在服务器上并且为使用的磁盘存储支付费用。

**22. Does Webmail use POP3, IMAP, or neither? If one of these, why was that one chosen? If neither, which one is it closer to in spirit? (E)**

这两种方案都不使用。但是它所采用的办法与 IMAP 很类似，因为它们两个都允许一个远程客户去管理一个远程邮箱。相反，POP3 只是把邮箱发送给客户并且在邮箱那边做处理。

**23. When Web pages are sent out, they are prefixed by MIME headers. Why? (E)**

浏览器必须要知道页面是否为 text, audio, video 或其它类型。MIME headers 提供了这些信息。

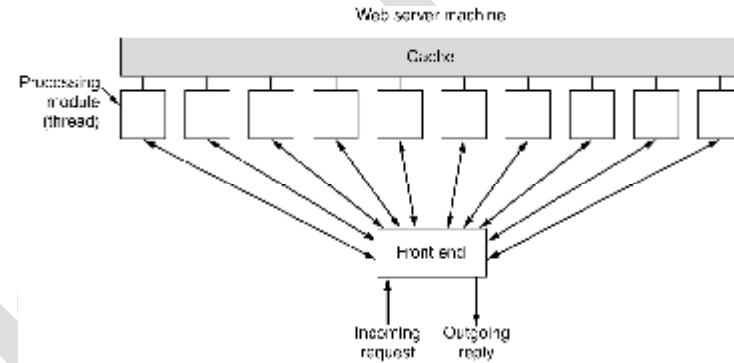
**24. When are external viewers needed? How does a browser know which one to use? (E)**

如果浏览器收到一个带有 MIME 类型的页面并且它不能够处理，这时它会调用一个外部查看器来显示页面。它会在一张配置表中或是从用户处查找查看器的名字。

**25. Is it possible that when a user clicks on a link with Netscape, a particular helper is started, but clicking on the same link in Internet Explorer causes a completely different helper to be started, even though the MIME type returned in both cases is identical? Explain your answer. (M)**

是有这样的可能。启动哪种帮助文件取决于内部浏览器的配置表。Netscape 和 IE 可能在配置上有所不同。进一步来讲，IE 采用的文件扩展类型比 MIME 更少，并且文件扩展类型可能会引入不同于 MIME 的帮助文件。

**26. A multithreaded Web server is organized as shown in Fig. 7-21. It takes 500  $\mu$ sec to accept a request and check the cache. Half the time the file is found in the cache and returned immediately. The other half of the time the module has to block for 9 msec while its disk request is queued and processed. How many modules should the server have to keep the CPU busy all the time (assuming the disk is not a bottleneck)? (E)**



如果一个模块收到了两个请求，一个接收到，一个被错过。

CPU 处理时间  $500\mu s * 2 = 1\text{ ms}$ ； CPU 等待时间为 9ms；

所以 CPU 利用率为 10%，需要 10 个模块。

**28. Although it was not mentioned in the text, an alternative form for a URL is to use the IP address instead of its DNS name. An example of using an IP address is <http://192.31.231.66/index.html>. How does the browser know whether the name following the scheme is a DNS name or an IP address? (E)**

DNS 名不能以数字结尾，因此这里没有二义性存在。

**29. Imagine that someone in the CS Department at Stanford has just written a new program that he wants to distribute by FTP. He puts the program in the FTP directory ftp/pub/freebies/newprog.c. What is the URL for this program likely to be? (E)**

The URL 可能是 <ftp://www.cs.stanford.edu/ftp/pub/freebies/newprog.c>

**30. In Fig. 7-25, [www.aportal.com](http://www.aportal.com) keeps track of user preferences in a cookie. A disadvantage of this scheme is that cookies are limited to 4 KB, so if the preferences are extensive, for example, many stocks, sports teams, types of news stories, weather for multiple cities, specials in numerous product categories, and more, the 4-KB limit may be reached. Design an alternative way to keep track of preferences that does not have this problem. (E)**



Domain	Path	Content	Expires	Secure
loms-casino.com	/	CustomerID=497793621	15-10-02 17:00	Yes
joes-store.com	/	Carl=1-00501;1-07031;2-13721	11-10-02 14:22	No
uportal.com	/	Pruls=Stk:SUNW+ORCL:Spt:Jets	31-12-10 23:59	No
sneaky.com	/	UserID=3627239101	31-12-12 23:59	No

可以这样来处理：只是将 customer ID 放入 cookie 中并且在服务器上的数据库中存储由该 customer ID 索引的用户喜好。通过这种方式喜好记录的大小就没有限制了。

**31. Sloth Bank wants to make on-line banking easy for its lazy customers, so after a customer signs up and is authenticated by a password, the bank returns a cookie containing a customer ID number. In this way, the customer does not have to identify himself or type a password on future visits to the on-line bank. What do you think of this idea? Will it work? Is it a good idea? (E)**

从技术上看，这种方案可以工作，但是这并不是一种好的主意。所有的用户必须做的就是修改 cookie 来取得其它用户的银行帐号。有这样的 cookie 能保证用户身份是安全的，但是用户应该被要求输入一个密码来证明他的身份。

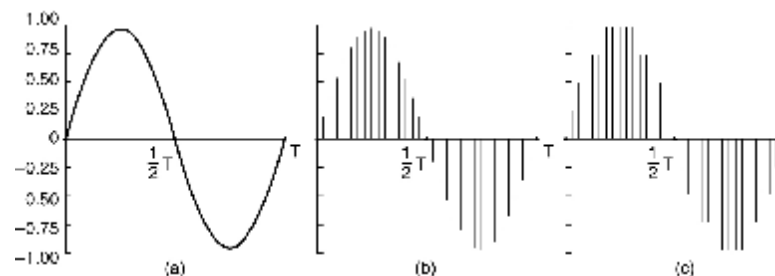
**42. Does it make sense for a single ISP to function as a CDN? If so, how would that work? If not, what is wrong with the idea?(E)**

当然可以。ISP 去到许多内容提供者那里并且取得他们的允许复制的内容到 ISP 的网站上。内容提供者可能要支付一定的费用。这种方案的缺点在于对 ISP 而言，这是一项很大的任务去联系众多的内容提供者。相对而言由 CDN(内容分发网络)来做这些更容易一些。

**45. A compact disc holds 650 MB of data. Is compression used for audio CDs? Explain your reasoning. (E)**

Audio needs 1.4 Mbps, which is 175 KB/sec. On a 650-MB device, there is room for 3714 sec of audio, which is just over an hour. CDs are never more than an hour long, so there is no need for compression and it is not used.

**46. In Fig. 7-57(c) quantization noise occurs due to the use of 4-bit samples to represent nine signal values. The first sample, at 0, is exact, but the next few are not. What is the percent error for the samples at  $1/32$ ,  $2/32$ , and  $3/32$  of the period? (E)**



The true values are  $\sin(2\pi i/32)$  for  $i$  from 1 to 3. Numerically, these sines are 0.195, 0.383, and 0.556. They are represented as 0.250, 0.500, and 0.500, respectively. Thus, the percent errors are 28, 31, and 10 percent, respectively.

**47. Could a psychoacoustic model be used to reduce the bandwidth needed for Internet telephony? If so, what conditions, if any, would have to be met to make it work? If not, why not? (E)**

理论上是可以使用的，而网络电话是实时的。对音乐而言，没有意义去花 5 分钟时间去编码 3 分钟的音乐。对于实时速度，这不起作用。心理声学压缩可以用于电话，但仅仅在于如果具有这样的芯片并且它能够使用压缩处理延时在 1 msec 左右。

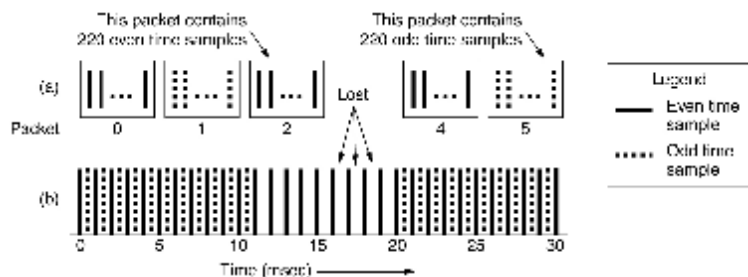
**48. An audio streaming server has a one-way distance of 50 msec with a media player. It outputs at 1 Mbps. If the media player has a 1-MB buffer, what can you say about the position of the low-water mark and the high-water mark? (M)**

缓冲区内定义一个低水印标记和一个高水印标记：

当服务器停止发送数据以后，缓冲区将开始变得空起来。当它到达低水印标记时，媒体播放器告诉服务器继续开始发送数据。设置低水印标记的位置时必须确保缓冲区不会出现数据供应不足的情况。从服务器端取得一个暂停命令要 50 msec，这段时间内将会有  $1\text{Mbps} \times 50\text{msec} = 50000\text{bits} = 6250\text{bytes}$  的数据到来，所以低水印标记要置为 6250 bytes 以上，不妨置为 50000 bytes；

服务器往外发送数据，直至缓冲区被填至高水印标记，然后媒体播放器告诉服务器暂停发送。缓冲区的高水印标记与尾部之间的距离必须大于网络的带宽-延迟之乘积： $1\text{Mbps} \times 50\text{msec} = 50000\text{bits} = 6250\text{bytes}$ ，不妨置为 500,000 bytes；

**49. The interleaving algorithm of Fig. 7-60 has the advantage of being able to survive an occasional lost packet without introducing a gap in the playback. However, when used for Internet telephony, it also has a small disadvantage. What is it?(E)**



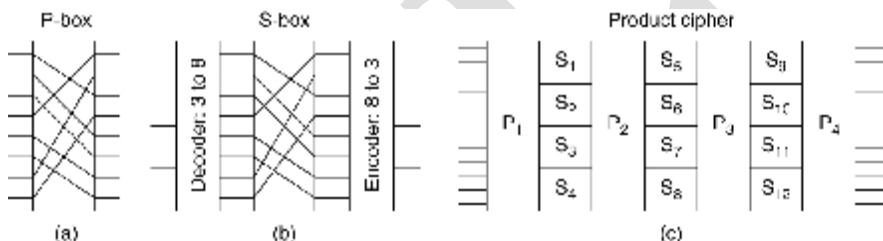
这会引入额外的延迟。在直接转发的机制中，在 5 msec 后，第一个分组可以被发送。而在这种机制下，系统必须等待 10 msec，之后它才能发送前 5 msec 的采样。

## Chapter 8 Network Security Problems

6. A fundamental cryptographic principle states that all messages must have redundancy. But we also know that redundancy helps an intruder tell if a guessed key is correct. Consider two forms of redundancy. First, the initial  $n$  bits of the plaintext contain a known pattern. Second, the final  $n$  bits of the message contain a hash over the message. From a security point of view, are these two equivalent? Discuss your answer. (E)

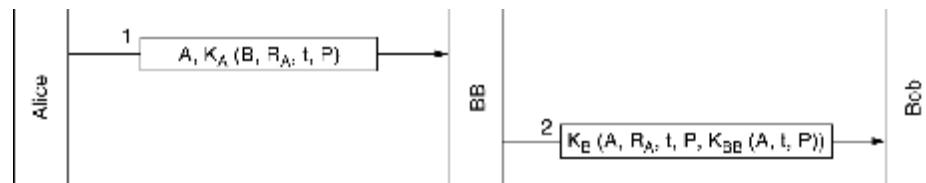
如果入侵者有足够大的计算能力，那这两种方式是一样的效果。但是如果不是这样的情况，那么第二种方法好一些。它会迫使入侵者做计算来尝试密钥的正确性，如果这种计算开销很大的话，会减慢入侵的速度。

7. In Fig. 8-6, the P-boxes and S-boxes alternate. Although this arrangement is esthetically pleasing, is it any more secure than first having all the P-boxes and then all the S-boxes? (E)



是的，更安全些。一系列连续的 P 盒可以被一个 P 盒来代替。S 盒也类似。

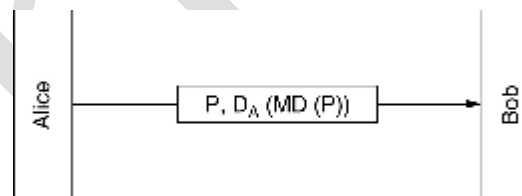
17. The signature protocol of Fig. 8-18 has the following weakness. If Bob crashes, he may lose the contents of his RAM. What problems does this cause and what can he do to prevent them? (E)



来自最后一条报文的  $R_{AS}$  可能仍在 RAM 中。如果它丢失了，那么 Trudy 就能够尝试着重放最近的报文给 Bob，并希望他看不出这是份复本。

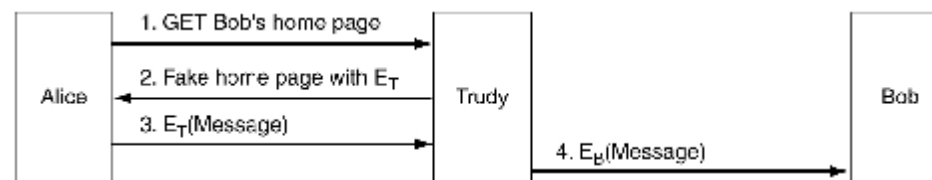
一种解决方式就是对 Bob 而言把每条进来的报文的  $R_A$  写到磁盘上后再开始工作。这样的话，重放就不会起作用。但是，现在有一个危险就是如果一个请求被写在磁盘的同时机器当掉了，那么这次请求将会永远得不到执行。

18. In Fig. 8-20, we see how Alice can send Bob a signed message. If Trudy replaces  $P$ , Bob can detect it. But what happens if Trudy replaces both  $P$  and the signature? (E)



如果 Trudy 把  $P$  和签名都替换了，当 Bob 用 Alice 的公钥去解密签名，他得到的不是明文  $P$  的正确的报文摘要。Trudy 可以放入一条错误的报文并且计算它的报文摘要，但是她不能用 Alice 的私钥对报文进行签名。

22. Consider the failed attempt of Alice to get Bob's public key in Fig. 8-23. Suppose that Bob and Alice already share a secret key, but Alice still wants Bob's public key. Is there now a way to get it securely? If so, how? (E)



可以的。Alice 用共享的密钥加密一个一次性的密钥并且发送给 Bob，Bob 送回一个用共享密钥加密的报文，报文中包含那个一次性的密钥以及他自己的一次性密钥，还有公钥。Trudy 不能伪造这条报文，并且要是她发送了随机的报文过去，当解密时它并没有包含 Alice 的一次性密钥。为了完成协议，Alice 会用 Bob 的公钥加密 Bob 的一次性密钥并发送回给 Bob。

**24. Suppose that a system uses PKI based on a tree-structured hierarchy of CAs. Alice wants to communicate with Bob, and receives a certificate from Bob signed by a CA X after establishing a communication channel with Bob. Suppose Alice has never heard of X. What steps does Alice take to verify that she is talking to Bob? (M)**

First Alice establishes a communication channel with X and asks X for a certificate to verify his public key. Suppose X provides a certificate signed by another CA Y. If Alice does not know Y, she repeats the above step with Y.

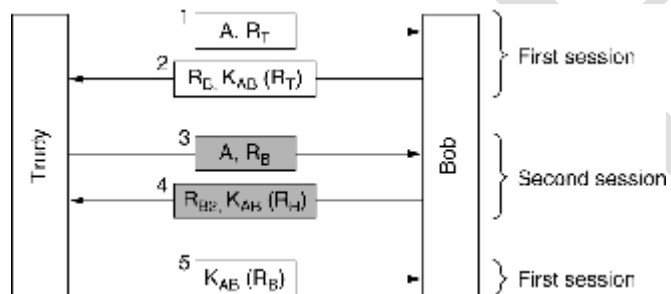
Alice continues to do this, until she receives a certificate verifying the public key of a CA Z signed by A and Alice knows A's public key. Note that this may continue until a root is reached, that is, A is the root. After this Alice verifies the public keys in reverse order starting from the certificate that Z provided. In each step during verification, she also checks the CRL to make sure that the certificate provided have not been revoked. Finally, after verifying Bob's public key, Alice ensures that she is indeed talking to Bob using the same method as in the previous problem.

**25. Can IPsec using AH be used in transport mode if one of the machines is behind a NAT box? Explain your answer. (E)**

No. AH in transport mode includes the IP header in the checksum. The NAT box changes the source address, ruining the checksum. All packets will be perceived as having errors.

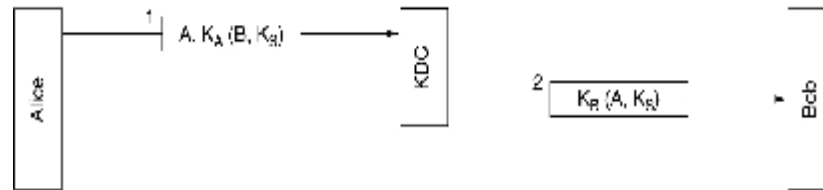
**26. Give one advantage of HMACs over using RSA to sign SHA-1 hashes. (E)**  
HMACs 的计算速度要快得多。

**30. Change one message in protocol of Fig. 8-34 in a minor way to make it resistant to the reflection attack. Explain why your change works. (E)**



在消息 2 中，把  $R_B$  放在加密的报文里而不是放在外面。通过这种方式，Trudy 就不能够发现  $R_B$  了并且反射攻击也不起作用。

**33. In the protocol of Fig. 8-39, why is A sent in plaintext along with the encrypted session key? (E)**



KDC（密钥分发中心）需要某种方式来区分是谁发出的报文，因此应用了一个 KDC 和 A 共享的密钥来加密会话密钥  $K_S$

**34. In the protocol of Fig. 8-39, we pointed out that starting each plaintext message with 32 zero bits is a security risk. Suppose that each message begins with a per-user random number, effectively a second secret key known only to its user and the KDC. Does this eliminate the known plaintext attack? Why? (M)**

不会。所有的 Trudy 不得不做的是采集来自或发送到同一用户的两条报文。她能够试关用同一个密钥去解密这两条。如果它们的随机数域是相同的，OK，她得到了正确的密钥。所有的所做的机制只是增加了她的工作负担而已。

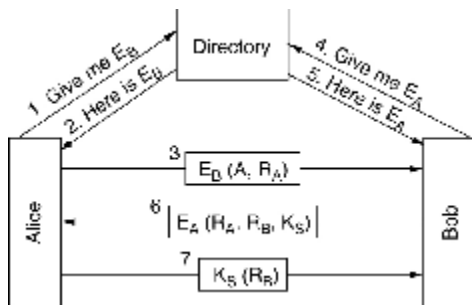
**35. In the Needham-Schroeder protocol, Alice generates two challenges,  $R_A$  and  $R_{A2}$ . This seems like overkill. Would one not have done the job? (E)**

两个随机数被用于不同的目的。 $R_A$  用在使 Alice 相信她正在与 KDC 进行通话。 $R_{A2}$  用于使 Alice 相信她正在与 Bob 进行通话。两者都是必需的。

**36. Suppose an organization uses Kerberos for authentication. In terms of security and service availability, what is the effect if AS or TGS goes down?(M)**

If AS goes down, new legitimate users will not be able to authenticate themselves, that is, get a TGS ticket. So, they will not be able to access any servers in the organization. Users that already have a TGS ticket (obtained from AS before it went down) can continue to access the servers until their TGS ticket lifetime expires. If TGS goes down, only those users that already have a server ticket (obtained from TGS before it went down) for a server S will be able to access S until their server ticket lifetime expires. In both cases, no security violation will occur.

**37. In the public-key authentication protocol of Fig. 8-43, in message 7,  $R_B$  is encrypted with  $K_S$ . Is this encryption necessary, or would it have been adequate to send it back in plaintext? Explain your answer. (M)**



发送回加密的  $R_B$  并不是本质上的。Trudy 没有办法知道它，并且它也不会被再使用了，所以它并不算是秘密。另一方面，这么做允许 allows a tryout of  $K_S$  to make doubly sure that it is all right before sending data. Also, why give Trudy free information about Bob's random number generator? In general, the less sent in plaintext, the better, and since the cost is so low here, Alice might as well encrypt  $R_B$ .