Chap 4, THE MEDIUM ACCESS CONTROL SUBLAYER

2. A group of N stations share a 56-kbps pure ALOHA channel. Each station outputs a 1000-bit frame on average 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?

答：pure ALOHA的效率约为1/(2e)=0.184，故可用带宽为 0.184*56 kbps = 10.3 kbps，每个站每秒发送10个bits，故为10bps，因此最多可容纳10300/10 = 1030个站。

4. 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.
(a) What is the chance of success on the first attempt?
(b) What is the probability of exactly k collisions and then a success?
(c) What is the expected number of transmission attempts needed?

答：每个时槽为40msec推知每秒有25个时槽，每秒有50个发送请求，每个时槽内的帧请求期望值为G=2。

(a)，一个“帧时”内生成k 帧的概率服从泊松分布：$\Pr[k] = \frac{G^k e^{-G}}{k!}$

对于分槽Aloha，首次发送时别人不发送的概率是

Pr[0] = e^(-2) = 0.135.

(b)，由于Pr[0]=e^(-2)，所以有冲突的概率是1-e^(-2)，故刚好发生k次冲突然后一次成功的概率是

(1-e^(-2))^k * e^(-2) = 0.865^k * 0.135.

(c)，设(b)情况的概率为p(k+1)，则每帧所需传送次数k的期望值为

$$\mathrm{E}(k) = \sum_{k=1}^{\infty} kp(k) == \sum_{k=1}^{\infty} k(1-e^{-2})^{k-1}e^{-2} = e^2 = 7.4$$

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

(a) What is the channel load, G?

(b) What is the throughput?

(c) Is the channel underloaded or overloaded?

答：见课堂练习

ALOHA Exercise

Example:

- Measurement of a slotted ALOHA channel with an infinite number of users show that 20 percent of slots are idle .

What is the channel load, G?

What is the throughput?

Is the channel underloaded or overloaded?

a) **Poisson distribution:** $P_r[k] = G^k e^{-G}/k!$

$P_0=e^{-G}$, so $G= -\ln P_0 = -\ln 0.2 = 1.6$

b) $S = G*e^{-G} = G*0.2 = 1.6*0.2 = 0.32$

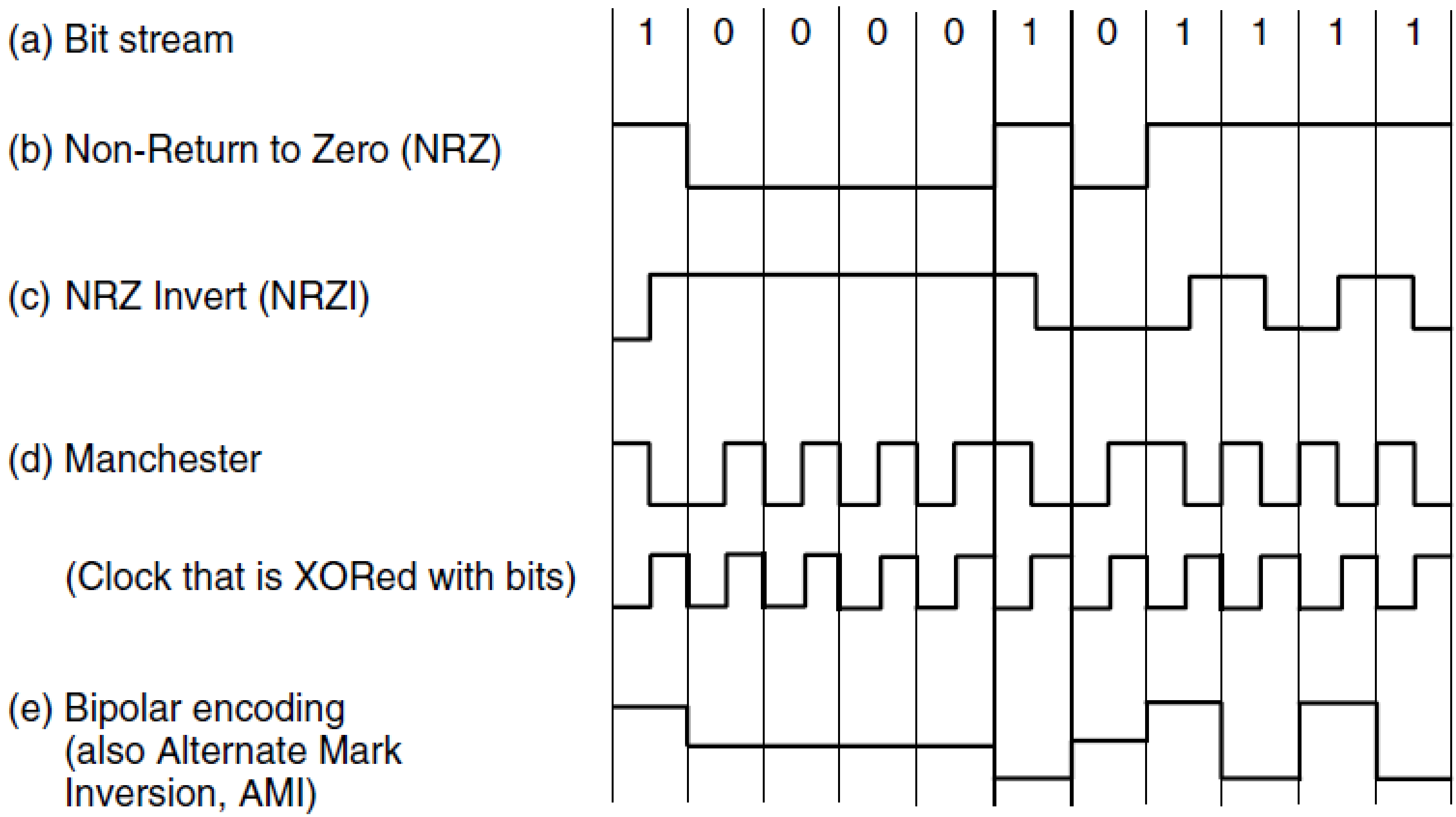
c) G = 1.6 >1, so the channel is overloaded.

13. What is the baud rate of classic 10-Mbps Ethernet?

答：传统10兆以太网使用曼彻斯特编码，曼彻斯特编码意味着每发送一个bit需要两个信号周期，故波特率为20Mbaud.

14. Sketch the Manchester encoding on a classic Ethernet for the bit stream 0001110101.

答：曼彻斯特编码使用一个高电平H连一个低电平L表示1，一个L连一个H表示0，故答案为LHLHLHHLHLHLLHHLLHHL。



15. 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?

答：1000米长的线往返需要10μsec。一次完整的单帧传输包含以下环节：

1、获取网线需要10 μsec，因为第一个bit往返至少10 μsec。
2、发送一个帧需要256/10=25.6 μsec。
3、单程传输时间5 μsec，最后一个bit在发送完后5 μsec后到。
4、接收方获取网线也要10 μsec。
5、发送确认帧32bits需要3.2 μsec.
6、确认帧全部到达发送方再需要单程传输时间5 μsec。

以上合计为58.8 μsec，期间发送一个帧内包含224个数据位，故而有效的数据传输率是224/58.8 = 3.8 Mbps。

18. 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?

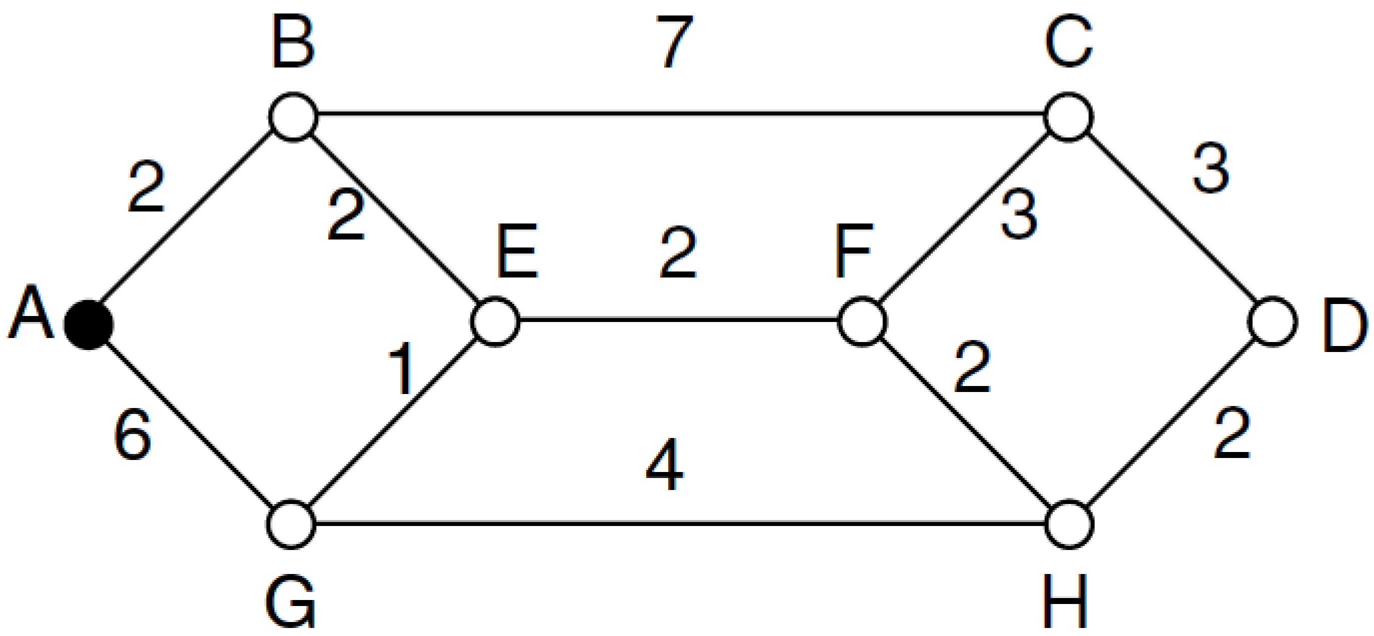
答：将快速以太网里头的最大线段延迟限制指定为普通以太网的十分之一即可。

19. Some books quote the maximum size of an Ethernet frame as 1522 bytes instead of 1500 bytes. Are they wrong? Explain your answer.

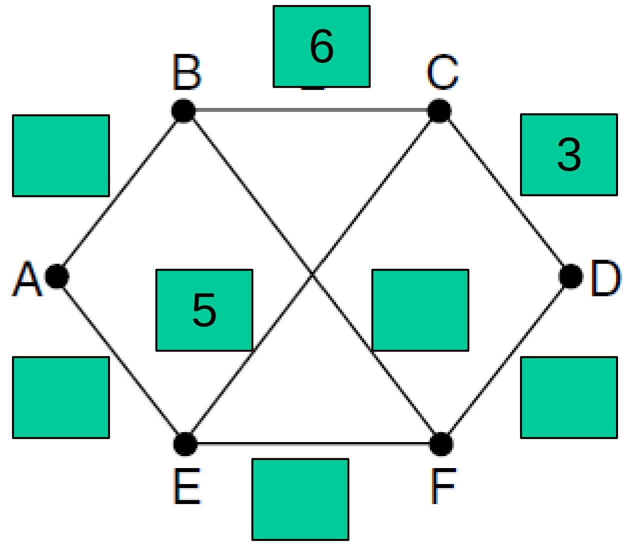
答：没有错。书上1500bytes主要是有效载荷，如果算上发送方和接收方地址，类型，校验码信息之类的，总计为1518bytes，如果再加4字节的VLAN信息，就是1522bytes。

Chap 5

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.



6. Consider the network of Fig. 5-12(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 cost of the links from *C* 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 cost.



Going via *B* gives (11, 6, 14, 18, 12, 8).
Going via *D* gives (19, 15, 9, 3, 12, 13).
Going via *E* gives (12, 11, 8, 14, 5, 9).
Taking the minimum (11, 6, 0, 3, 5, 8).
The outgoing lines are (*B*, *B*, –, *D*, *E*, *B*).

26. Suppose that instead of using 16 bits for the network part of a class B address originally, 20 bits had been used. How many class B networks would there have been?

With a 2-bit prefix, there would have been 18 bits left over for the network.
Consequently, the number of networks would have been 2^18 or 262,144.
However, all 0s and all 1s are special, so only 262,142 are available.

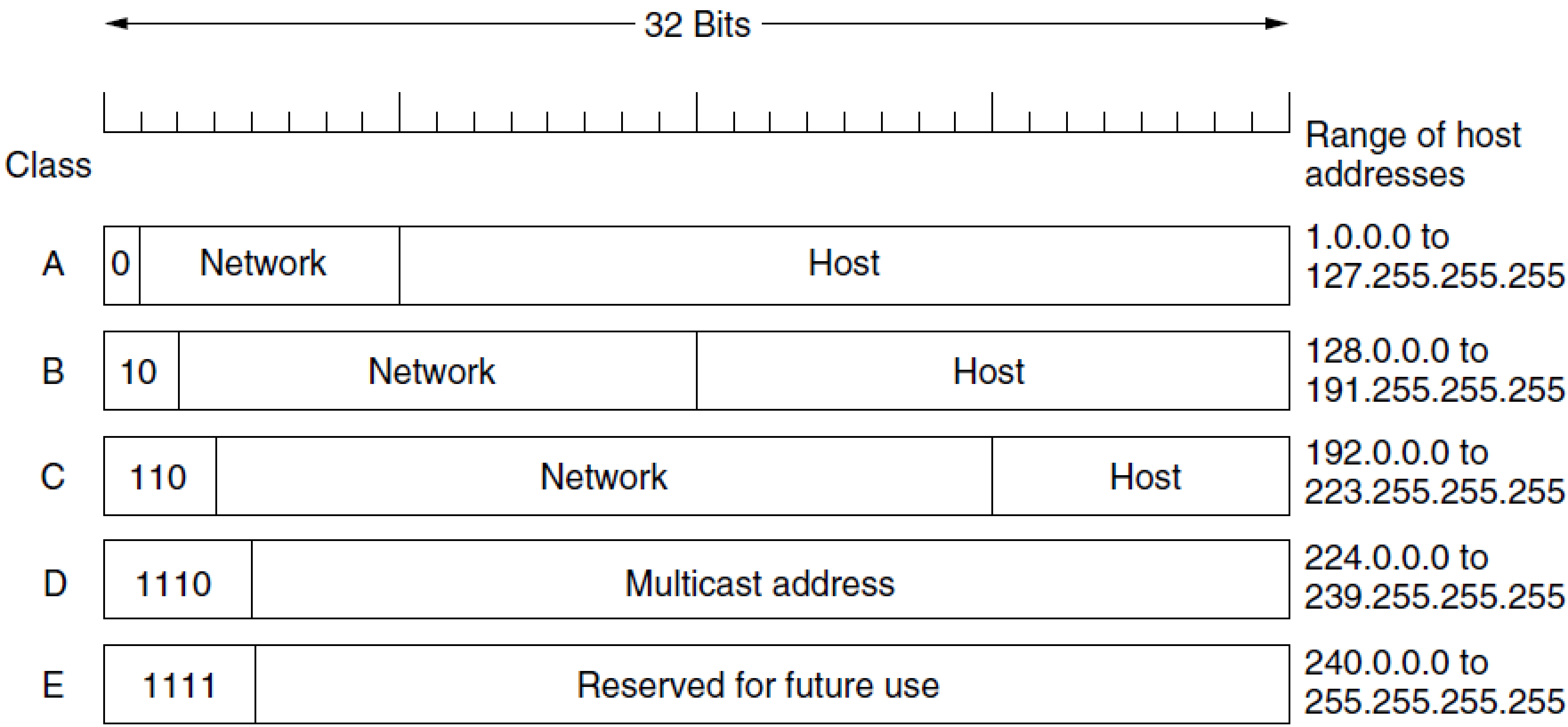


Figure 5-53. IP address formats.

30. A large number of consecutive IP addresses 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.

To start with, all the requests are rounded up to a power of two. The starting address, ending address, and mask are as follows:
A: 198.16.0.0 – 198.16.15.255 written as 198.16.0.0/20
B: 198.16.16.0 – 198.16.23.255 written as 198.16.16.0/21
C: 198.16.32.0 – 198.16.47.255 written as 198.16.32.0/20
D: 198.16.64.0 – 198.16.95.255 written as 198.16.64.0/19

33. A router has the following (CIDR) entries in its routing table:

Address/mask	**Next hop**
135.46.56.0/22	Interface 0
135.46.60.0/22	Interface 1
192.53.40.0/23	Router 1
default	Router 2

For each of the following IP addresses, what does the router do if a packet with that address arrives?

(a) 135.46.63.10
(b) 135.46.57.14
(c) 135.46.52.2
(d) 192.53.40.7
(e) 192.53.56.7

The packets are routed as follows:
(a) Interface 1
(b) Interface 0
(c) Router 2
(d) Router 1
(e) Router 2

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

Connection-oriented service (TCP):
File transfer (FTP), remote login (Telnet), SMTP, HTTP and video on demand.

Connectionless service (UDP):

Credit card verification and other point-of-sale terminals, electronic funds transfer, and many forms of remote database access are inherently connectionless, with a query going one way and the reply coming back the other way.

3. Give three examples of protocol parameters that might be negotiated when a connection is set up.

window size
maximum packet size
data rate
timer values

Chap 6

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?

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.	Request a release of the connection

Figure 6-2. The primitives for a simple transport service.

The LISTEN call could indicate a willingness to establish new connections but not block. When an attempt to connect was made, the caller could be given a signal. It would then execute, say, OK or REJECT to accept or reject the connection.
In our original scheme, this flexibility is lacking.

3. In the underlying model of 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?

The dashed line from *PASSIVE ESTABLISHMENT PENDING* to *ESTABLISHED* is no longer contingent on an acknowledgement arriving. The transition can happen immediately. In essence, the *PASSIVE ESTABLISHMENT PENDING* state disappears, since it is never visible at any level.

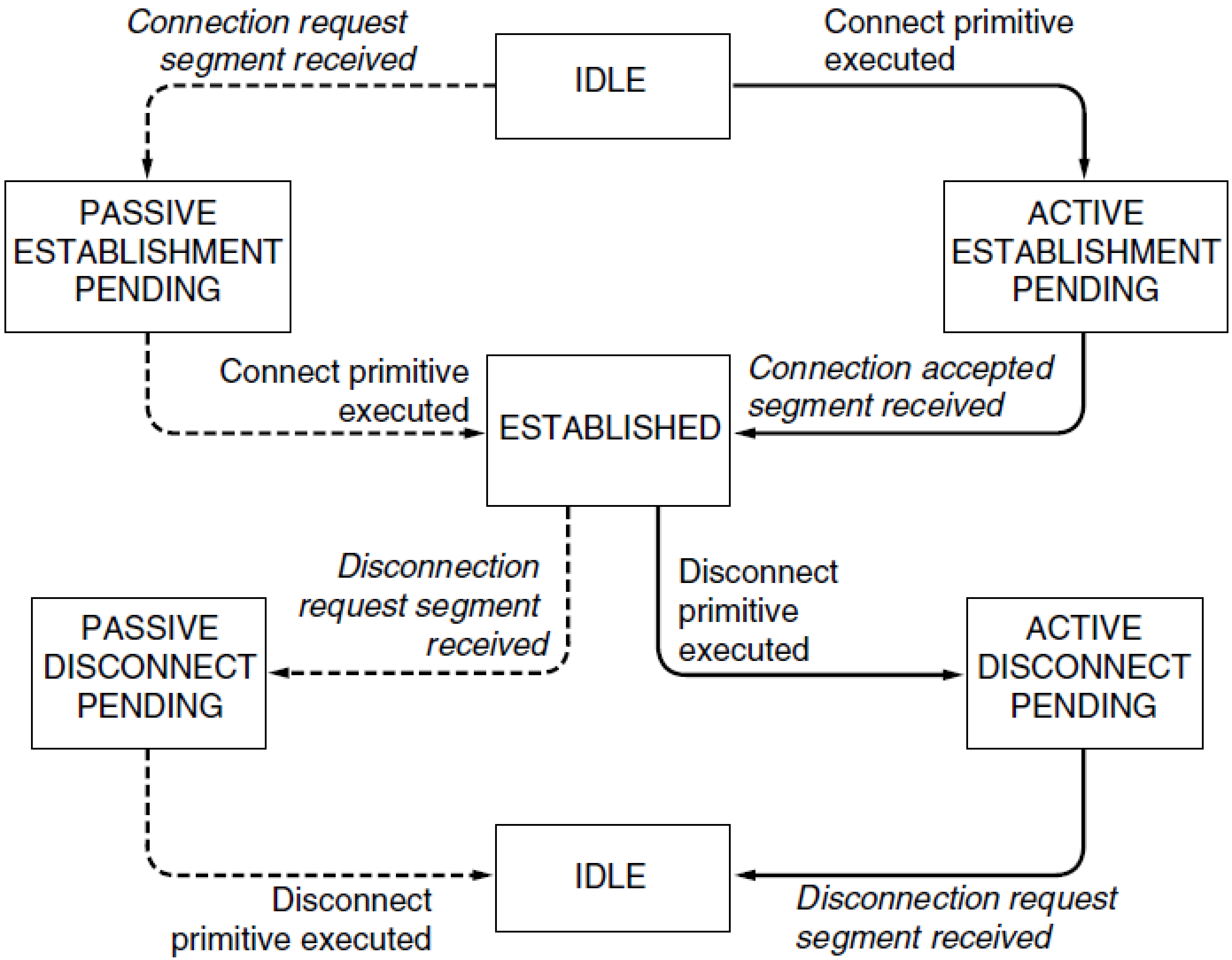


Figure 6-4. A state diagram for a simple connection management scheme. Transitions labeled in italics are caused by packet arrivals. The solid lines show the client's state sequence. The dashed lines show the server's state sequence.

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

The sliding window is simpler, having only one set of parameters (the window edges) to manage. Furthermore, the problem of a window being increased and then decreased, with the segments arriving in the wrong order, does not occur.
However, the credit scheme is more flexible, allowing a dynamic management of the buffering, separate from the acknowledgements.

36. In a network whose max segment is 128 bytes, max segment lifetime is 30 sec, and has 8-bit sequence numbers, what is the maximum data rate per connection?

A sender may not send more than 255 segments in 30 sec, so the data rate is no more than 255 X 128 X 8 /30 bits/sec = 8.704 kbps.

39. 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.

The size of the sequence space is 2^64 bytes, which is about 2 X 10^19 bytes. A 75-Tbps transmitter uses up sequence space at a rate of 9. 375 X 10^12 sequence numbers per second. It takes 2 million seconds to wrap around. Since there are 86,400 seconds in a day, it will take over 3 weeks to wrap around, even at 75 Tbps. A maximum packet lifetime of less than 3 weeks will prevent the problem. In short, going to 64 bits is likely to work for quite a while.

1. 在 OSI 参考模型中，下列各层中不属于通信子网的是(　　)

(A) 物理层　(B) 数据链路层　(C) 网络层　(D) 会话层

2. 当前因特网所采用的协议族是（　　）

(A) TCP/IP　(B) NCP　(C) UNIX　(D) ACM

3. OSI 参考模型中，加密和解密是（　　）层的功能。

(A) 传输　(B) 会话　(C) 表示　(D) 应用

4. IEEE 802.3 标准为第一代 10Mbps 以太网使用的访问方法是（　　）CSMA/CD。

(A) 1-persistent　(B) p-persistent　(C) non-persistent　(D) 以上均不是

5. PCM 是（　　）转换的一个实例。

(A) digital-to-digital　(B) digital-to-analog　(C) analog-to-analog　(D) analog-to-digital

DACAD

1. 循环冗余校验 CRC 中的生成式包含（　　　）因子时，可检测出所有的奇数位错误。

(A) x　　(B) x+1　　(C) 1　　(D)以上均不是

2. 本地电话网络采用（　　　）交换技术。

(A) 电路　　(B) 分组　　(C) 报文　　(D)以上均不是

3. 采用位填充法进行成帧，成帧标识为 01111110 。如果需要传送的比特串为 011111011111110，则经位填充后，此比特串变为 (　　　)(不包括起始和结束标志)

(A) 0111110011111011110　　(B) 011111001111110110

(C) 011111010111011110　　(D) 011111011111111110

4. 若数据链路的发送窗口尺寸 WT=15，在发送 7 号帧、并接到 5 号帧的确认帧后，发送方还可连续发送（　　　）。

(A) 4 帧　　(B) 5 帧　　(C)10 帧　　(D)13 帧

5. N 个站共享一个 200 kbps 的纯 ALOHA 信道。每个站平均每 10 秒输出一个 10000 位长的帧（即使前面的帧还没有被发送出去），N 最大可以为（　　）。

(A) 16　　(B) 36　　(C)64　　(D)128

BAADB

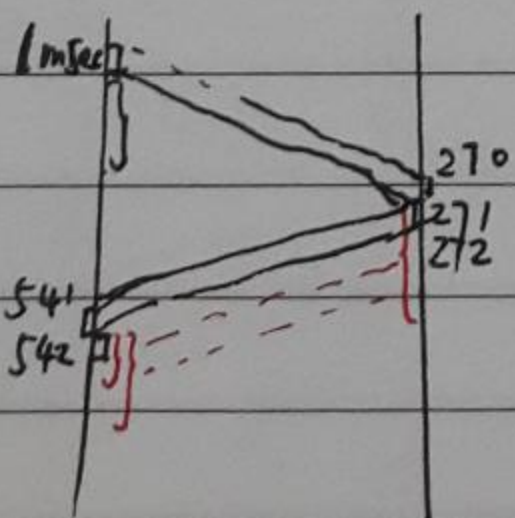
1. ___（1）___定理定义了无噪声信道理论上的最大数据传输速率，___（2）___定理定义了加性白噪声信道理论上的最大数据传输速率。
2. 一个 64-QAM 信号的波特率是 2000，其比特率是___（3）___。
3. 在回退 N 帧协议中，如果用 5 个 bit 序号对数据帧进行编号，发送窗口大小的最大值是___（4）___，接收窗口大小的最大值是___（5）___。
4. 若码字包含 m 个信息位和 r 个校验位，为了纠正单比特错误，m 与 r 应满足的关系是___（6）___。常见的纠正单比特错误的纠错码是___（7）___。
5. IEEE 802.11 标准采用___（8）___协议进行多路访问控制。
6. 就成帧方法而言，PPP 面向___（9）___填充，HDLC 面向___（10）___填充。

（1）	Nyquist/奈奎斯特；	（6）	$(m + r + 1) <= 2^r$；
（2）	Shannon/香农	（7）	海明码
（3）	12000bps	（8）	CSMA/CA
（4）	31	（9）	字节
（5）	1	（10）	比特

32. 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

(a) Stop-and-wait?
(b) Protocol 5?
(c) Protocol 6?

1 Mbps ⇒ 1帧1000bits 发送耗时1msec



故一帧返往周期542msec

最大可达利用率在无差错情况时发生

① 停等协议，542ms里发1帧，利用率1/542.

② 协议5发送窗大小为7 ⇒ 7/542.

③ 协议6发送窗大小为4 ⇒ 4/542.

UDP的校验和计算机制，什么类型的错误可以/无法检测?

TCP/UDP的区别，与IP的关系。

ADSL即非对称数字用户线路，全称Asymmetric Digital Subscriber Line，是一种数据传输方式，其上行和下行带宽不对称，因此称为非对称数字用户线路。

海明距离：两个码字的对应比特取值不同的比特数称为这两个码字的海明距离。

NAT、DHCP、ARP的概念与机制。

CSMA/CA：带有冲突避免机制的载波侦听多路访问协议

VLAN即虚拟局域网（Virtual Local Area Network），是一组逻辑上的设备和用户，这些设备和用户并不受物理位置的限制，可以根据功能、部门及应用等因素将它们组织起来，相互之间的通信就好像它们在同一个网段中一样，由此得名虚拟局域网。

数据报和虚电路的适用场合，根据具体情况分析采用那种方式？以及一些相关机制如服务质量、拥塞控制等。

5.1.5 Comparison of Virtual-Circuit and Datagram Subnets

Issue	Datagram subnet	Virtual-circuit subnet
Circuit setup	Not needed	Required
Addressing	Each packet contains the full source and destination address	Each packet contains a short VC number
State information	Routers do not hold state information about connections	Each VC requires router table space per connection
Routing	Each packet is routed independently	Route chosen when VC is set up; all packets follow it
Effect of router failures	None, except for packets lost during the crash	All VCs that passed through the failed router are terminated
Quality of service	Difficult	Easy if enough resources can be allocated in advance for each VC
Congestion control	Difficult	Easy if enough resources can be allocated in advance for each VC

1. “三次握手”建立连接

答：答：在建立连接过程中，使用“三次握手”的方式为通信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，即通信双方进行相互之间的通信参数（例如计数器初始值）的设置和交换（两次握手），第三次握手的意义在于表明连接建立成功，通讯正式开始，而第三次握手往往伴随正式数据发送而行，被称为“捎带应答”。

1. ARQ

答：自动重传请求（Automatic Repeat Request），通过接收方请求发送方重传出错的数据报文来恢复出错的报文，是通信中用于处理信道所带来差错的方法之一，有时也被称为后向纠错。

2. 一个系统的协议结构有n 层。应用程序产生M 字节长的报文。网络软件在每层都加上h 字节长的协议头。那么，网络带宽中有多大比率用于协议头信息的传输？

答：总共有n 层，每层加h 字节，在每个报文上附加的头字节的总数等于hn，因此头消耗的有关空间所占的网络带宽的比率为hn / (M+hn)。

3.一个以太网的MAC子层自上层收到1510字节数据，这些数据可以被封装在一个以太网帧中吗？如果不可以，需要发送几个以太网帧？每个帧中的数据字段大小分别是多少？

答：以太网帧数据字段最大长度为1500字节，因此1510字节的上层数据需要被封装在两个帧中，其中第一个帧的数据字段为1500字节，第二个帧的数据字段为46个字节（10个有效上层数据+36个填充字节）。

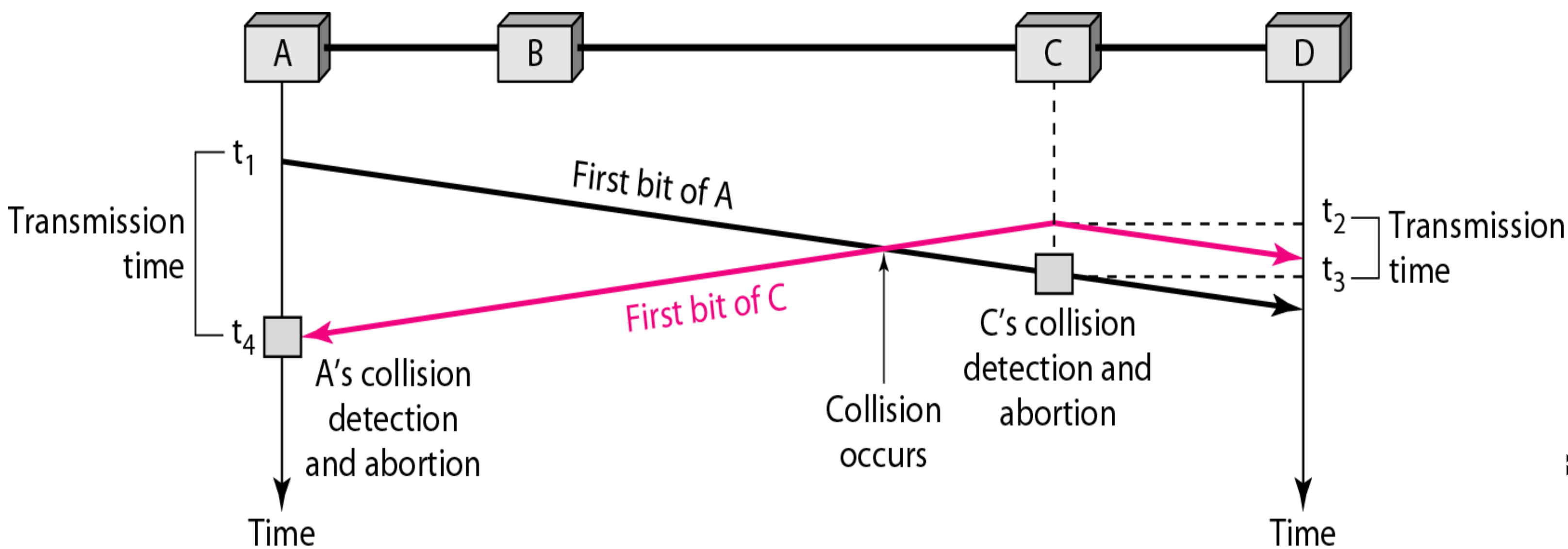
1. （10分）在下图中，数据传输速率为10Mbps，站点A与C相距2000m，传播速率为2×10^8 m/s。站点A在t1 = 0 时刻发送一个长帧，站点C在t2 = 3 μs 时刻发送一个长帧。假设帧足够长，能保证两个站点均能监测到冲突的发生。试问：

1） 站点C听到冲突的时刻t3；

2） 站点A听到冲突的时刻t4；

3） 站点A在监测到冲突前已经发送的比特数；

4） 站点C在监测到冲突前已经发送的比特数。



答:

$t1 = 0$, $t2 = 3\mu s$

1. **（2分）$t3 - t1 = 2000\ m / 2 \times 10^8\ m/s = 10\mu s$,**
   $t3 = 10\mu s + t1 = 10\mu s$;
2. **（2分）$t4 - t2 = 2000\ m / 2 \times 10^8\ m/s = 10\mu s$,**
   $t4 = 10\mu s + t2 = 13\mu s$;
3. **（3分）站点A发送数据的时间为$t4 - t1 = 13 - 0 = 13\mu s$, 10 Mbps * 13μs = 130 bits;**
4. **（3分）站点C发送数据的时间为$t3 - t2 = 10 - 3 = 7\mu s$, 10 Mbps * 7μs = 70 bits.**