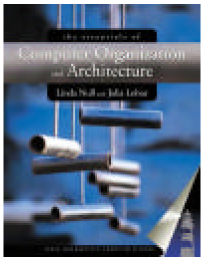

The Essentials of Computer Organization and Architecture

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Chapter 11 Instructor's Manual

Chapter Objectives

Chapter 11, Network Organization and Architecture, focuses on network organization and architecture, including network components and protocols. The OSI model and TCP/IP suite are introduced in the context of the Internet. This chapter is by no means intended to be comprehensive. The main objective is to put computer architecture in the correct context relative to network architecture. Lectures should focus on the following points:

- **History of networks.** A brief introduction into the roots and architecture of the Internet helps students to understand the current state of networking.
- **Network protocols**. Understanding network protocols is key in understanding how networks work. The deli parable is an excellent example to explain networking protocols in non-networking terms. Lecture can then focus on the OSI 7-layer model.
- **TCP/IP architecture.** IPv4 and IPv6 are the two versions of the Internet Protocol in use today. Understanding IP and TCP and their requirements is critical in understanding most Internet applications, which rely on these two protocols.
- **Network organization.** The physical network components (such as the transmission media, interface cards, repeaters, switches, hubs, bridges, gateways, and routers) are important as they must be chosen with consideration to the anticipated load and distance to be covered.
- **Digital links.** The physical components of a network must be connected, and there are many ways in which this can be achieved. In addition to FDM, PCM, TDM and PDH, ISDN and ATM are discussed. Although the public switched telephone network continues to be the Internet "on ramp" for most homes and small businesses, ISDn and DSL are offering some relief to overcrowded analog circuits. However, routing problems and congestion may require revamping the basic architecture that forms the foundation for the Internet.
- **A look at the Internet.** Traditional dialups and modems and DSL are the most common methods of ramping on to the Internet.

Required Lecture Time

The important concepts in Chapter 11 can typically be covered in 3 lecture hours (this provides only an exposure to the topics). However, if a teacher wants the students to have a mastery of all topics in Chapter 11, 11 lecture hours are more reasonable.

Lecture Tips

Much of the material in this chapter (as in Chapter 7) can be assigned for students to read (it is mostly intended as FYI material). However, discussing the historical development of the Internet is important, as understanding its roots is necessary for understanding why and how it currently functions. The section on routers and routing can be covered in a single lecture, as students often have problems understanding the routing protocols.

Answers to Exercises

1. In what is the traffic of an early business computer network different from that of an early scientific-academic network? Is there such a distinction between these two types of systems today?

Ans.

Business transactions tend to be frequent and short (few bytes are sent at once). Scientific data exchanges might be less frequent and longer, such as when data files, research papers and long e-mails are exchanged. With the advent of pervasive office automation, business exchanges now include word-processed documents, e-mails, spreadsheets, and small data collections, in addition to the frequent and short transactions that have always been characteristic of business data communication.

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2. Why is the ISO/OSI protocol stack called a reference model? Do you think this will always be the case?

Ans.

The ISO/OSI protocol stack is called a reference model because there are no commercial products that implement it to the letter of the specification. It was not originally intended to be a reference model, but a living communications specification. Should an implementation of the ISO/OSI model should ever become popular, then it would no longer be a reference model.

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3. How is a Network Layer protocol different from a Transport Layer protocol?

Ans.

The network layer establishes the route and ensures that the size is compatible with all equipment between the source and destination. The transport layer provides quality assurance. Therefore the protocol for the transport layer is much more sophisticated than a network layer protocol.

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4. Internet protocol standards are devised through the efforts of thousands of people all over the world—regardless of their having any particular background in data communications. On the other hand, proprietary protocols are created by a much smaller group of people, all of whom are directly or indirectly working for the same employer.
   a. What advantages and disadvantages and disadvantages do you think are offered by each approach? Which would produce a better product? Which would produce a product more quickly?
   b. Why do you think that the IETF approach has achieved ascendancy over the proprietary approach?

Ans.

a. Usually, smaller groups can get things done more quickly than larger groups. This is especially true if the group shares a common vision. Large groups are especially problematic when various committees must pass approval over the work.

b. With the IETF approach, it is possible that things could become deadlocked if the process degrades to an "us against them" factionalism. One reason that the IETF approach has worked so well is that the standards are free and open, and many of the people having a great deal of input are experts in their field. They care more about producing a good product than producing a profit for some corporate entity.

♦ 5. In our description of the Window field in the TCP header, we said:

Notice that if the receiver's application is running very slowly, say it's pulling data 1 or 2 bytes at a time from its buffer, the TCP process running at the receiver should wait until the application buffer is empty enough to justify sending another segment.

What is the "justification" for sending another segment?

Ans.

The payload (data) of the TCP segment should be made as large as possible so that the network overhead required to send the segment is minimized. If the payload consisted of only 1 or 2 bytes, the network overhead would be at least an order of magnitude greater than the data transmitted.

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6. The OSI protocol stack includes Session and Presentation layers in addition to its Application layer. TCP/IP applications, such as Telnet and FTP, have no such separate layers defined. Do you think that such a separation should be made? Give some advantages and disadvantages of incorporating the OSI approach into TCP/IP.

Ans.

The ISO/OSI protocol layers that this question speaks to are the Session, Presentation, and Application Layers. The general aim of layered protocols is to provide modularity and separation of duties so that one layer can be substituted for another should the need arise.

The most compelling argument in favor of providing separate functions at each of these layers takes into account the need for security. Having monolithic applications such as HTTP, FTP, and Telnet, requires these applications be rewritten if (more) secure communications are desired. In fact, we see this with HTTPs, sFTP, and SSL.

A counter argument is that these applications are not huge to begin with, so rewriting them to provide secure services is in fact easier and less complex than providing layered services.

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7. Why is the length of a TCP segment limited to 65,536 bytes?

Ans.

The answer to this question is revealed by looking at the definition of the Data Offset field of the TCP segment format. A TCP segment is the PDU of the Transmission Control Protocol. No particular segment can be larger than 65,536 bytes because the sequence number of each transmitted byte is stored in a 32-bit field in the TCP header.

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8. Why does the IETF use the word octet instead of byte? Do you think that this practice should continue?

Ans.

At the time that the Internet (DARPANet) specifications were being written, byte was considered to be a proprietary name used by the IBM corporation. (This may have intimated that IBM had influence in the protocols, or had sanctioned them.) Now that byte is part of everyone's lexicon, it may be time to end whatever confusion is caused by using the word octet. Your opinion may vary.

♦ 9. Into which class of networks do the following IP addresses fall?

♦ a. 180.265.14.3

♦ b. 218.193.149.222

♦ c. 92.146.292.7

Ans.

a. Class B   b. Class C   c. Class A

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10. Into which class of networks do the following IP addresses fall?

a. 223.52.176.62

b. 127.255.255.2

c. 191.57.229.163

Ans.

a. Class C   b. Class A   c. Class B

♦ 11. A station running TCP/IP needs to transfer a file to a host. The file contains 1024 bytes. How many bytes, including all of the TCP/IP overhead, would be sent assuming a payload size of 128 bytes and both systems are running IPv4. (Also assume that the three-way handshake and window size negotiation have been completed and that no errors occur during transmission.)

♦ a. What is the protocol overhead (stated as a percentage)?

♦ b. Perform the same calculation if both clients are using IPv6.

Ans.

a. We will assume that no options are set in the TCP or IP headers and ignore any session shutdown messages. There are 20 bytes in the TCP header and 20 bytes in the IP header. For a file size of 1024 bytes, using a payload of 128 bytes, 8 payloads will have to be sent. Thus, 8 TCP and IP headers. With 40 bytes overhead per transmission unit, we have 8×40 bytes of overhead added to the 1024 payload bytes, giving a total transmission of 1344 bytes. The overhead percentage is $320 \div 1344 \times 100\% = 23.8\%$.

b. The minimal IPv6 header is 40 bytes long. Added to the 20 bytes in the TCP header, each transmission contains 60 overhead bytes, for a total of 480 bytes to send the 8 payloads. The overhead percentage is: $480 \div 1504 \times 100\%$ or 31.9%.

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12. A station running TCP/IP needs to transfer a file to a host. The file contains 2048 bytes. How many bytes, including all of the TCP/IP overhead, would be sent assuming a payload size of 512 bytes and both systems are running IPv4? (Also assume that the three-way

handshake and window size negotiation have been completed and that no errors occur during transmission.)

a. What is the protocol overhead (stated as a percentage)?

b. Perform the same calculation if both clients are using IPv6.

Ans.

a. There are 20 bytes overhead in each of the TCP and IP headers for IPv6. (40 bytes total overhead). If we have a payload of 512 bytes, we need 2048/512 = 4 payloads. Thus, each IP datagram consists of 20 bytes IP overhead + 20 bytes TCP overhead + 512 bytes of payload = 552 bytes per datagram * 4 = 2208 bytes total. We have a total of 40*4 = 160 bytes of overhead. So the protocol overhead as a percentage is: 160/2208 * 100% = 7.25% (rounded to two places).

b. The IPv6 overhead is 40 bytes. Add this to the TCP overhead and we have 60 bytes per datagram. So the IPv6 protocol overhead as a percentage is: 240/2208 * 100% = 10.87% (rounded to two places).

♦13. Two stations running TCP/IP are engaged in transferring a file. This file is 100K long, the payload size is 100 bytes, and the negotiated window size is 300 bytes. The sender receives an ACK 1500 from the receiver.

♦a. Which bytes will be sent next?

♦b. What is the last byte number that can be sent without an ACK being sent by the receiver?

Ans.

a. Bytes 1500 through 1599 b. Byte 1799

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14. Two stations running TCP/IP are engaged in transferring a file. This file is 10K long, the payload size is 100 bytes, and the negotiated window size is 2000 bytes. The sender receives an ACK 900 from the receiver.

a. Which bytes will be sent next?

b. What is the last byte number that can be sent without an ACK being sent by the receiver?

Ans.

a. Bytes 900 through 999 b. Byte 2899

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15. What problems would present themselves if TCP did not allow senders and receivers to negotiate a timeout window?

Ans.

Slow connections may trigger a time out event in the sender because acknowledgements may not get back quickly enough. Fast connections may not time out quickly enough, wasting a whole lot of bandwidth while the sender is waiting to "be sure" of the timeout condition.

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16. IP is a connectionless protocol while TCP is connection-oriented. How can these two protocols coexist in the same protocol stack?

Ans.

TCP provides a means for setting up a reliable communications stream on top of the unreliable IP.

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17. Section 11.6.1 states that when using 4B/5B encoding, a signal-carrying capacity of 125MHz is required for a transmission medium to have a bit rate of 100Mbps.
   a. What signal-carrying capacity would be required if Manchester coding were used instead?
   b. What signal-carrying capacity would be required if modified frequency modulation (MFM) coding were used, assuming that the occurrence of a 0 or a 1 are equally-likely events?

   (Manchester and MFM coding are explained in Chapter 2.)

Ans.

Since Manchester requires two signal transitions for every bit, we would need 200MHz, while for MFM, we would need only 50 (as it has a transition only for each pair).

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18. a. The signal power for a particular class of network wiring is 8733.26 dB and the noise rating at that particular signal strength at 100MHz is 41.8 dB. Find the signal-to-noise ratio for this conductor.
   b. The signal-to-noise rating for the network wiring in Part a is 9.5 dB and the noise rating is 36.9 dB when a 200MHz signal is transmitted. What is the signal strength?

Ans.

a.

$$\text{Signal-to-Noise Ratio (dB)} = 10 \log_{10} \frac{8733.26 \text{ dB}}{41.8 \text{ dB}} = 23.2 \text{ dB}$$

b.

$$9.5 \text{ dB} = 10 \log_{10} \frac{\text{Signal dB}}{36.9 \text{ dB}} \Rightarrow \text{Signal dB} = 328.8\text{dB}$$

19.a. The signal power for a particular class of network wiring is 2898 dB and the noise rating at that particular signal strength at 100MHz is 40 dB. Find the signal-to-noise ratio for this conductor.

   b. The signal-to-noise rating for the network wiring in Part a is 0.32 dB and the noise rating is 35 dB when a 200MHz signal is transmitted. What is the signal strength?

Ans.

a.

$$\text{Signal-to-Noise Ratio (dB)} = 10 \log_{10} \frac{2898 \text{ dB}}{40 \text{ dB}} = 1.86 \text{ dB}$$

b.

$$0.32 \text{ dB} = 10 \log_{10} \frac{\text{Signal dB}}{35 \text{ dB}} \Rightarrow \text{Signal dB} = 37.68 \text{ dB}$$

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20. How big is a physical PDU? The answer to this question determines the number of simultaneous transmissions for many network architectures.

    If a signal propagates through copper wire at the rate of 2×108 meters per second, then on a carrier running at 10Mbps the length of each bit pulse is given by:

$$\frac{\text{Speed of propagation}}{\text{Speed of bus}} = \frac{2 \times 10^8 \text{ m/s}}{10 \times 10^6 \text{ b/s}} = 20 \text{ meters/bit.}$$

If a data frame is 512 bits long, then the entire frame occupies:

(Length of one bit) $\times$ (Frame size) = $20 \times 512 = 10,240$ meters.

a. How big is a 1024-bit packet if the network runs at 100Mbps?
b. How long is it if the network speed is increased to 155Mbps?
c. At 100Mbps, how much time elapses as one of these frames passes a particular point in the network?

Ans. (to a and c only)

$$\frac{\text{Speed of propagation}}{\text{Speed of bus}} = \frac{1.8 \times 10^8 \text{ m/s}}{100 \times 10^6 \text{ b/s}} = 1.8 \text{ meters/bit} \times 1024 \text{ bits} = 3643.2 \text{ meters.}$$

$$\frac{\text{Size of frame}}{\text{Speed of bus}} = \frac{1024 \text{ bits}}{1.8 \times 10^8 \text{ m/s}} = 568.8 \times 10^{-8} \text{ seconds or } 5.68\ \mu\text{sec.}$$

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21. It looks like the 4B/5B bit cells in Figure 11.14 are fairly small. How long, in reality, is such a bit cell on a 125MHz line? (Use the constants and formulas from the previous question.)

Ans.

Bit cell size = Speed of propagation * speed of one bit = 1.8×10^8 m/s * $(1/125) \times 10^{-6}$ s
= $1.8 * 0.008 * 10^2$ m = 1.44 meters.

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22. With reference to Figure 11.21, suppose Router 4 derives its routing table from the routing tables of Router 1 and Router 3. Complete the routing table for Router 4 using the same format as the routing table of the other three routers.

Ans.

Dest	Next Hop	Hop Count
A	R3	1
B	--	0
C	R3	2
D	--	0
L	R1	1
M	R1	1
N	R1	1
R	R3	2
T	R3	1
W	R3	1

---

23. Since trellis code and other phase-change modulation techniques have been used in increase the signal-carrying capacity of common telephone lines, couldn't we also do the same thing with digital lines?

Ans.

Digital lines, unlike analog lines, don't have as much "phase" to modulate. Their signals are a series of highs and lows with nothing in between (barring serious attenuation).

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24. In Section 11.8.1 we state, using Shannon’s law, that the maximum data rate for a standard analog telephone line is approximately 30,000 bps with a signal-to-noise ratio of 30dB. The ratio of signal to noise is 1000, because the signal-to-noise ratio is 10 $\log_{10}$ signal dB / noise dB, as explained in Section 11.6.1.

If a binary signal is sent over a 10KHz channel whose signal to noise ratio is 20dB, what is the maximum achievable data rate?

Ans.

$$20\text{ dB} = 10\log_{10}\left(\frac{S}{N}\right) \Rightarrow 2\text{ dB} = \log_{10}\left(\frac{S}{N}\right) \Rightarrow 10^2\text{ dB} = \left(\frac{S}{N}\right)$$

$$\begin{aligned}\text{DataRate}_{max} &= 10000\log_2(1 + 100)\\ &= 10000(6.6582) = 66.582\text{ Kbps}\end{aligned}$$

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25. Using the same idea as in the previous question, suppose a channel has an intended capacity of 10 Mbps with a bandwidth of 4MHz. What should is the minimum signal-to-noise ratio (in dB) that the channel must have in order for it to be possible to achieve this data rate?

Ans.

$$10\text{ Mbps} = 20\text{MHz}\log_2\left(1 + \frac{S}{N}\right)$$

$$2.5 = \log_2\left(1 + \frac{S}{N}\right)$$

$$2^{2.5} = \left(1 + \frac{S}{N}\right) = 5.656854249 \Rightarrow \frac{S}{N} = 4.656854249$$

$$\text{signal-to-noise ratio} = 10\log_{10}(4.656854249) = 6.6809$$

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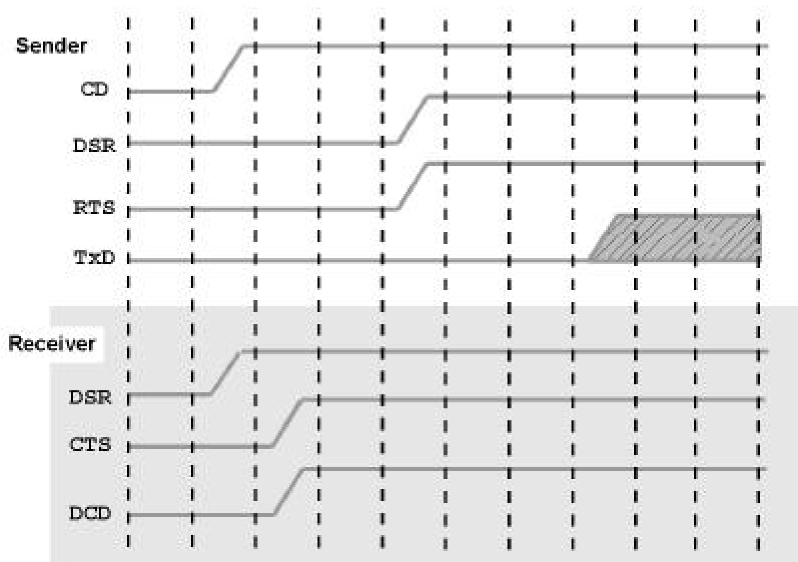
26. Construct a timing diagram to illustrate how two modems establish a connection using the RS-232 protocol described in the text.

Ans.

Owing to the nature of asynchronous communications, the time quanta shown in the diagram won't be equal, as with a bus timing diagram. The vertical divisions serve only to divide one state from the next.



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27. The North American TDM DS-0 frame takes 125µs to pass one point on the line. How many milliseconds does it take for the European equivalent to pass a point on the line?

Ans.

The frame is still 125µs. Transmission takes place at a higher frequency in the European system.

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28. Figure 11.33b shows a TCM constellation. Devise a code of bit strings that could be used for this constellation.
[Hint: The 32 signal points encode bit strings that are 4 bits long. The fifth bit is used for parity. Either even or odd parity may be used for this exercise.]

Ans.

A correct answer provides a unique representation for all of the digits from 0 through 15 (with appropriate parity). An good answer, supplies a rational pattern to the assignment. (The values could be assigned arbitrarily.) An excellent answer makes certain that adjacent points have at least two bits that are different, thus increasing their Hamming distance.

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29. Devise a trellis code modulation constellation and encoding scheme that will encode 4 bits per baud (including the parity bit).

Ans.

Refer to the previous answer.
