Chapter 1 Assignments

Review 1: What advantages does TDM have over FDM in a circuit-switched network? Please compare different multiplexing technologies used in Mobile phone system: TDM, FDM, CDM, and explain how 4G uses TDM or FDM.

Solutions:

TDM and FDM are two widely used multiple access techniques. FDM is older technique often used for analog signals whereas TDM is relatively a newer technique used for digital signals. TDM advantage over FDM is that it offers bandwidth saving with ATDM (allocate time slots on demand dynamically) and there is low interference between the signals that are being multiplexed.

4G uses TDM and FDM for multiplexing and duplexing. For TD-LTE, the receive channel and the transmit channel take turns (i.e. divide the time between them) on the same frequency band. The time divisions are asymmetric, meaning that more time-slots are allocated to data going from the tower to the phone than from the phone to the tower. Whereas for FD-LTE, two frequencies are allocated, one for the transmit channel and the other for the receive channel. Orthogonal frequency division multiple access (OFDMA) is the multiple access method used in the LTE downlink. The LTE uplink is based on the single-carrier frequency division multiple access (SD-FDMA) mode.

Problem6: In this problem we consider sending real-time voice from Host A to Host B over a packet-switched network (VoIP). Host A converts analog voice to a digital 64 kbps bit stream on the fl y. Host A then groups the bits into 56-byte packets. There is one link between Host A and B; its transmission rate is 2 Mbps and its propagation delay is 10 msec. As soon as Host A gathers a packet, it sends it to Host B. As soon as Host B receives an entire packet, it converts the packet's twobits to an analog signal. How much time elapses from the time a bit is created (from the original analog signal at Host A) until the bit is decoded (as part of the analog signal at Host B)?

Solutions:

Consider the first bit in a packet. Before this bit can be transmitted, all of the bits in the packet must be generated. This requires

$$\frac{56 \cdot 8}{64 \times 10^3} \text{sec} = 7\text{msec.}$$

The time required to transmit the packet is

$$\frac{56 \cdot 8}{2 \times 10^6} \text{sec} = 224 \mu \,\text{sec.}$$

Propagation delay = 10 msec.
A similar analysis shows that decoding we need 7msec.

So the delay until decoding is

$$7\text{msec} + 224\mu\,\text{sec} + 10\text{msec} + 7\text{msec} = 24.224\text{msec}$$

Problem 7: Suppose users share a 3 Mbps link. Also suppose each user requires 150 kbps when transmitting, but each user transmits only 10 percent of the time. (See the discussion of statistical multiplexing in Section 1.3.)

a. When circuit switching is used, how many users can be supported?

b. For the remainder of this problem, suppose packet switching is used. Find the probability that a given user is transmitting.

c. Suppose there are 120 users. Find the probability that at any given time, exactly n users are transmitting simultaneously. *(Hint:* Use the binomial distribution.)

d. Find the probability that there are 21 or more users transmitting simultaneously.

Solutions:

a) 20 users can be supported.

b) $p = 0.1$.

c) $\binom{120}{n} p^n (1-p)^{120-n}$.

d) $1 - \sum_{n=0}^{20} \binom{120}{n} p^n (1-p)^{120-n}$ = 1-0.992=0.008

Problem 24: Suppose two hosts, A and **B,** are separated by 20,000 kilometers and are connected by a direct link of R = 2 Mbps. Suppose the propagation speed over the link is 2.5×10^8 meters/sec.

a. Calculate the bandwidth-delay product, $R \times d_{prop}$

b. Consider sending a file of 800,000 bits from Host A to Host B. Suppose the file is sent continuously as one large message. What is the maximum number of bits that will be in the link at any given time?

c. Provide an interpretation of the bandwidth-delay product.

d. What is the width (in meters) of a bit in the link? Is it longer than a football field?

e. Derive a general expression for the width of a bit in terms of the propagation speed s, the transmission rate R, and the length of the link m.

Solutions:

a) 160,000 bits

b) 160,000 bits

c) The bandwidth-delay product of a link is the maximum number of bits that can be in the link.
d) the width of a bit = length of link / bandwidth-delay product, so 1 bit is 125 meters long, which is longer than a football field
e) *s/R*

Problem 30: In modern packet-switched networks, the source host segments long, application-layer messages (for example, an image or a music file) into packets and sends the packets into the network. The receiver then reassembles the packets back into the original message. We refer to this process as *message segmentation.* Figure 1.28 illustrates the end-to-end of a message with and without message segmentation. Consider a message that is 8×10^6 bits long that is to be sent from source to destination in Figure 1.28. Suppose each link in the figure is 2 Mbps. Ignore propagation, queuing, and processing delays.

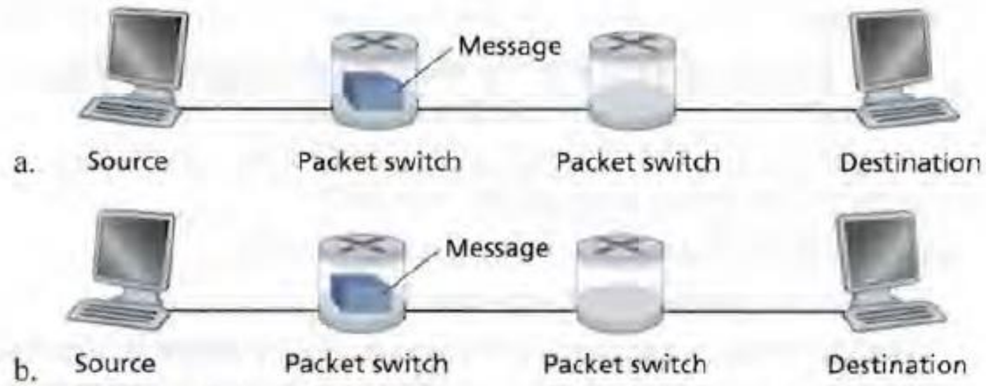

Figure 1.28 • End-to-end message transport: (a) without message segmentation; (b) with message segmentation.

a. Consider sending the message from source to destination *without* message segmentation. How long does it take to move the message from the source host to the first packet switch? Keeping in mind that each switch uses store-and-forward packet switching, what is the total time to move the message from source host to destination host?
b. Now suppose that the message is segmented into 4,000 packets, with each packet being 2,000 bits long. How long does it take to move the first packet from source host to the first switch? When the first packet is being sent from the first switch to the second switch, the second packet is being sent from the source host to the first switch. At what time will the second packet be fully received at the first switch?
c. How long does it take to move the file from source host to destination host when message segmentation is used? Compare this result with your answer in part (a) and comment.
d. Discuss the drawbacks of message segmentation.

Solutions:

a) Time to send message from source host to first packet switch =

$\frac{8\times10^6}{2\times10^6}\sec = 4\sec$. With store-and-forward switching, the total time to move message from source host to destination host = $4\sec\times 3\,hops = 12\sec$

b) Time to send 1st packet from source host to first packet switch = .

$\frac{2\times10^3}{2\times10^6}\sec = 1\,m\sec$. Time at which 2nd packet is received at the first switch = time at which 1st packet is received at the second switch =

$2\times 1\,m\sec = 2\,m\sec$

c) Time at which 1st packet is received at the destination host =

$1\,m\sec\times 3\,hops = 3\,m\sec$. After this, every 1msec one packet will be received; thus time at which last (4000th) packet is received =

$3\,m\sec + 3999*1m\sec = 4.002\sec$. It can be seen that delay in using message segmentation is significantly less (almost 1/3rd).

d) Drawbacks:
- Packets have to be put in sequence at the destination.
- Message segmentation results in many smaller packets. Since header size is usually the same for all packets regardless of their size, with message segmentation the total amount of header bytes is more.

Additional 1: Please explain why we need layering protocol model and explain the hourglass philosophy of the TCP/IP model.

Solutions:

Most network software are organized as a stack of layers or levels, each one built upon the one below it. The advantages of layering model:

- To reduce design complexity, divide the communication problem into subpieces and to design a separate protocol for each subpiece, making each protocol easier to design, analyze and implement.
- Independence. Each layer could be designed, maintained and updated independently, as long as keep in mind the services the lower layer provides for it and the services it should provide for the upper layer.
- Flexibility. Allow subsets of protocols be used as needed and allow any one of the protocols be replaced or updated.

The "Hourglass" philosophy of Internet is: IP bridges different applications over different networks. If everybody just supports IP, can use many different applications over many different networks

Additional 2: For a 4kHz voice channel with signal-to-noise ratio 30dB. Is it possible to provide 56kbps data rate service?

Solutions:

According to Shannon theorem, the maximum data rate of this channel= $4k*\log_2(1+S/N)$= $4k*\log_2(1+1000)\approx 4k*10=40$kbps. So it's impossible to provide 56kbps data rate service on this channel.

Additional 3: 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?

Solutions:

The total delay of the circuit-switched network= circuit setup time + transmission delay + propagation delay =$s + x/b + k*d$
For the packet-switched network, let $x/p=n\ldots\ldots r$.
The total delay of the packet-switched network= the end-to-end delay of the first packet + transmission delay of all the packets except the first one = $x/b+ (k-1)p/b+k*d$
So compare these two delays, we can conclude that if $(k-1)p<b*s$, then the packet network has a lower delay.

Discussion4 (Don't submit): Skype offers a service that allows you to make a phone call from a PC to an ordinary phone. This means that the voice call must pass through both the Internet and through a telephone network. Discuss how this might be done.

For Your Information:

Skype uses VoIP over Internet with packet switching technology where the digitalized speech data is encapsulated in UDP header, IP header and frame header (Ethernet, WiFi or PPP) and transferred over the packet-switching Internet.
Skype uses circuit switching technology when the speech data transferred over Public Switched Telephone Network (PSTN). So we need a gateway between the Internet and PSTN to do the translation. The telephone network can also use VoIP technology to reduce the cost.