Name …...ID …............................... Section …........Seat No..........

Sirindhorn International Institute of Technology
Thammasat University

Midterm Examination: Semester 1/2009

Course Title : ITS323 Introduction to Data Communications

Instructor : Dr Steven Gordon

Date/Time : Monday 27 July 2009; 13:30 – 16:30

Instructions:

- This examination paper has 19 pages (including this page).
- Condition of Examination
  Closed book (No dictionary; Non-programmable calculator is allowed)
- Students are not allowed to be out of the exam room during examination. Going to the restroom may result in score deduction.
- Turn off all communication devices (mobile phone etc.) and leave them under your seat.
- Write your name, student ID, section, and seat number clearly on the answer sheet.
- The space on the back of each page can be used if necessary.
- Assume bits are ordered from left to right: 1^{st} bit, 2^{nd} bit, 3^{rd} bit, …, n^{th} bit
- Unless otherwise stated in the question, assume the speed of transmission is $3\text{x}10^8$m/s
- Free space propagation path loss:

$$\frac{P_t}{P_r} = \frac{(4\pi d)^2}{G_t G_r \lambda^2}$$

- Antenna gain for parabolic antenna with area A:

$$G = \frac{4\pi A}{\lambda^2}$$

Part A – Multiple Choice Questions [22 marks]

Select the most accurate answer (only select one answer). Each correct answer is worth 2 marks. Incorrect answer is 0 marks. No answer is 0 marks.

1. Which technique is used for transmitting analog data as digital signals?
   a) Frequency Modulation
   b) Pulse Code Modulation
   c) Quadrature Amplitude Modulation
   d) Binary Frequency Shift Keying
   e) Manchester Encoding
   f) None of the above
2. Which of the following is an example of a network layer address?
   a) 00:17:31:7e:50:7d
   b) its323@ict.siit.tu.ac.th
   c) www.siit.tu.ac.th
   d) 72.103.16.5
   e) Port 22
   f) None of the above
3. What is the absolute bandwidth of the signal:

$$s(t)=15\sin(2000\pi t)+5\sin(6000\pi t)+3\sin(10000\pi t)+2\frac{1}{7}\sin(14000\pi t)$$

   a) 1000 Hz
   b) 2000 Hz
   c) 3000 Hz
   d) 6000 Hz
   e) 10000 Hz
   f) 12000 Hz
   g) 14000 Hz
   h) 28000 Hz
4. Which technique can be described as "vary the frequency of the output carrier signal as the amplitude of the input data changes":
   a) Frequency Modulation
   b) Frequency Shift Keying
   c) Quadrature Amplitude Modulation
   d) Bipolar AMI Encoding
   e) Phase Modulation
   f) Delta Modulation
5. Which of the following is incorrect?
   a) Optical fibre provides higher data rates than electrical cabling technologies
   b) With coaxial cable signals can be transmitted over a large distance than with twister pair.
   c) Unshielded twister pair is easier to install than shielded twisted pair
   d) Electrical cable technologies are designed to minimise the effects of interference from other sources on the transmitted signal
   e) The most common technology used in home telephone lines and in-building LANs is coaxial cable
   f) Optical fibre can be used over larger distances the twisted pair.
6. Quadrature PSK uses the phases: 45°, 135°, 225°, 315°. If the duration of each signal

element (that is, signal at one phase) is 200ns, then the data rate is:

a) 1 Mb/s
b) 2.5 Mb/s
c) 5 Mb/s
d) 10 Mb/s
e) 25 Mb/s
f) 50 Mb/s
g) 100 Mb/s

7. A transmission systems that provides half-duplex communications between A and B:
   a) Only allows A to send to B
   b) Only allows B to send to A
   c) If A is sending to B, then B cannot send to A at the same time
   d) If A is sending to B, then B can send to A at the same time
   e) Allows both A and B to transmit to each at the same time

8. If a signal has a period of 4us, then the wavelength of the signal is:
   a) 1200 m
   b) 2500 m
   c) 2.5 KHz
   d) 120000 m
   e) 120000 Hz
   f) 250000 Hz
   g) 2.5 MHz

9. A link with data rate of 3Mb/s has a received signal power of 6.3W and received noise level of 20dBm. What is the minimum bandwidth required?
   a) 20 KHz
   b) 300 KHz
   c) 500 KHz
   d) 600 KHz
   e) 1.5 MHz
   f) 3 MHz
   g) 6 MHz

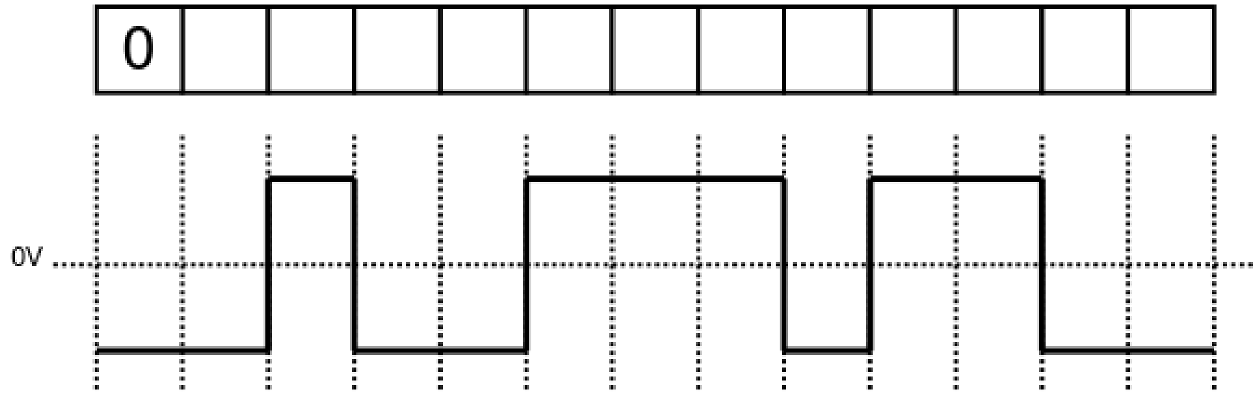
10. The User Datagram Protocol (UDP) is an example transport layer protocol. Normally it would be implemented as:
    a) Software in the operating system
    b) Part of a user application, such as web browser, email client or audio/video streaming application.
    c) Device drivers that control the LAN/WAN interface cards
    d) hardware on the LAN/WAN interface cards
    e) An application installed by users that want to use UDP

11. Which layer in the Internet stack has the role of delivering packets from source node to intermediate nodes and eventually to destination node when there are multiple links between source and destination?
    a) Physical
    b) Transport
    c) Application
    d) Hardware
    e) Data Link
    f) Network

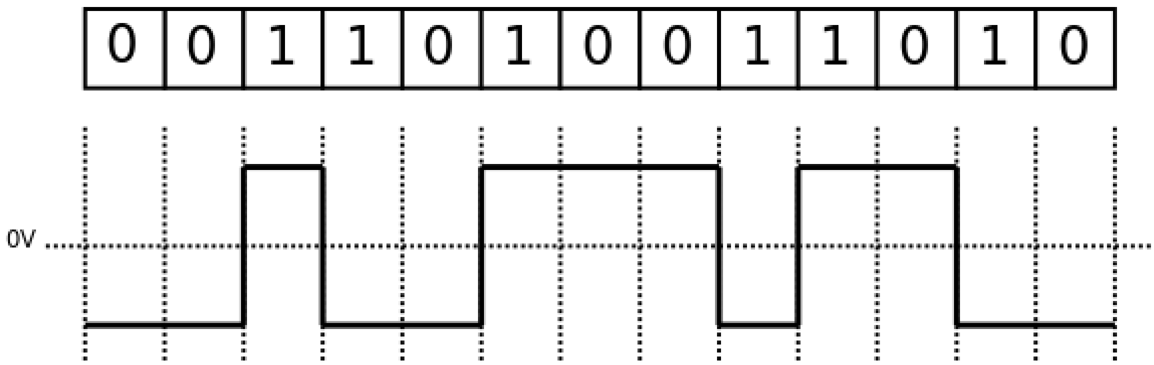
Part B – General Questions [78 marks]

Question 1 [12 marks]

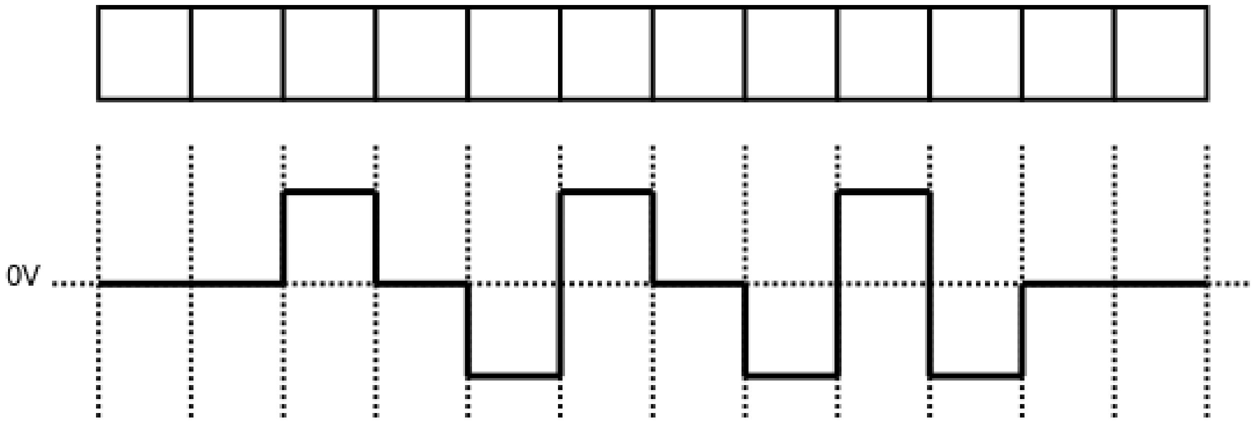
a) The following digital signal was encoded using NRZ-Invert. What is the digital data? (Fill in the boxes). [3 marks]



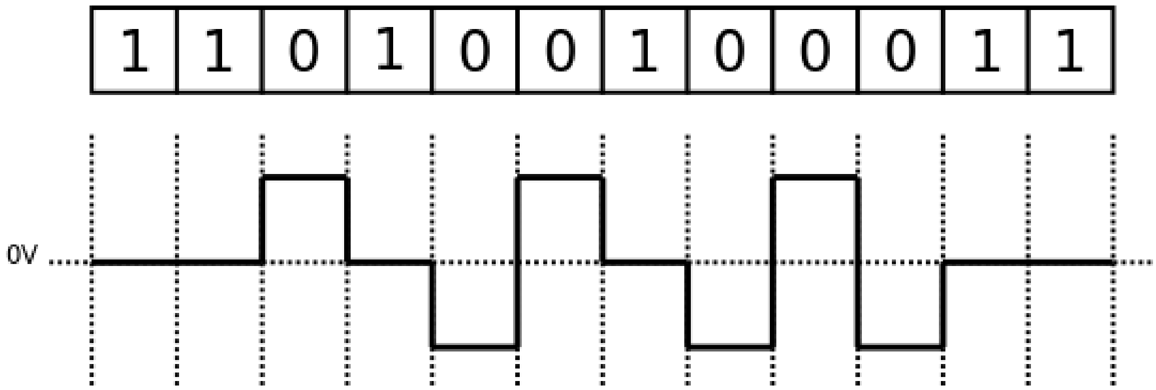
Answer



b) The Pseudoternary digital encoding scheme alternates between positive and negative voltage levels between successive bit 0's, and uses zero voltage for bit 1. Write the received bits in the boxes for the following received digital signal. [3 marks]



Answer



The following data is to be sent using a combination of FSK and ASK. There are 2 possible frequencies and 4 possible amplitudes.

010101001001010110110111

c) Select and describe a mapping of bits to signals (sinusoids) that uses all possible combinations of frequencies and amplitudes. [3 marks]

Answer

A possible mapping:

000	A=1	f=1
001	A=2	f=1
010	A=3	f=1
011	A=4	f=1
100	A=1	f=2

101	A=2	f=2
110	A=3	f=2
111	A=4	f=2

d) Using the mapping you selected in part (c), draw the analog signal to be transmitted. [3 marks]

Answer

Question 2 [11 marks]

Consider a point-to-point wireless communications system using two parabolic antennas:

- Transmit antenna diameter: 1 metre
- Receive antenna gain: 20dBi
- Signal frequency: 3GHz
- Distance between transmitter and receiver: 10km
- Receive power threshold: -80dBm

a) Assuming free space path loss, what is the minimum transmit power required? [6 marks]

Answer

Wavelength: $\lambda = \frac{3\times 10^8}{3\times 10^9} = 0.1\text{m}$

Gain of transmit antenna: $G_t = \frac{4\pi\pi\, 0.5^2}{0.1^2} = 986.9604...$

Absolute gain of receive antenna: $G_r = 10^{20/10} = 100$

Receive power: $P_r = 10^{-80/10} = 10^{-8}\, mW = 10^{-11}\, W$

Transmit power:

$$\begin{aligned} P_t &= \frac{P_r (4\pi d)^2}{G_t G_r \lambda^2} \\ &= \frac{10^{-11}\, 16\pi^2\, 10000^2}{986.9604 \times 100 \times 0.01} \\ &= 1.6\times 10^{-4}\, W = 160\, uW \end{aligned}$$

The transmit power required is 160uW.

In the free space path loss model, the absolute path loss (L) between the two antenna's can be written as:

$$L = \frac{(4\pi d)^2}{\lambda^2}$$

However, the free space path loss model does not consider obstructions or other environmental factors. Assume you have measured the real path loss between the two antennas to be L_{dB} = 130dB

b) Using the measured path loss (instead of free space path loss), what is the minimum transmit power required? [5 marks]

Answer

Absolute loss: $L = 10^{130/10} = 10^{13}$

Transmit power:

$$\begin{aligned} P_t &= \frac{P_r L}{G_t G_r} \\ &= \frac{10^{-11} 10^{13}}{986.9604 \times 100} \\ &= 1014\, uW \end{aligned}$$

Question 3 [9 marks]

A standard encoding format for digital telephony is Pulse Code Modulation.

a) Assuming the human voice has a spectrum of frequencies ranging from 200Hz to 4000Hz, what sampling rate should be used to retain all necessary information in the digital data? [1 mark]

Answer

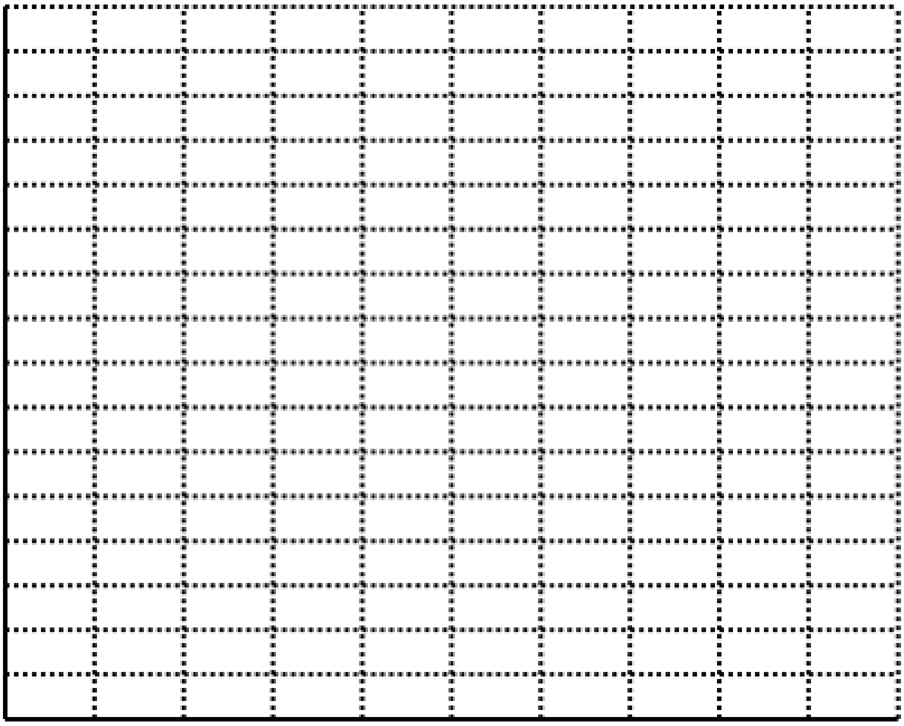
Using the Nyquist-Shannon sampling theorem the sampling rate should be twice the highest frequency component.

Sampling rate = 2 x 4000 = 8000 samples per second.

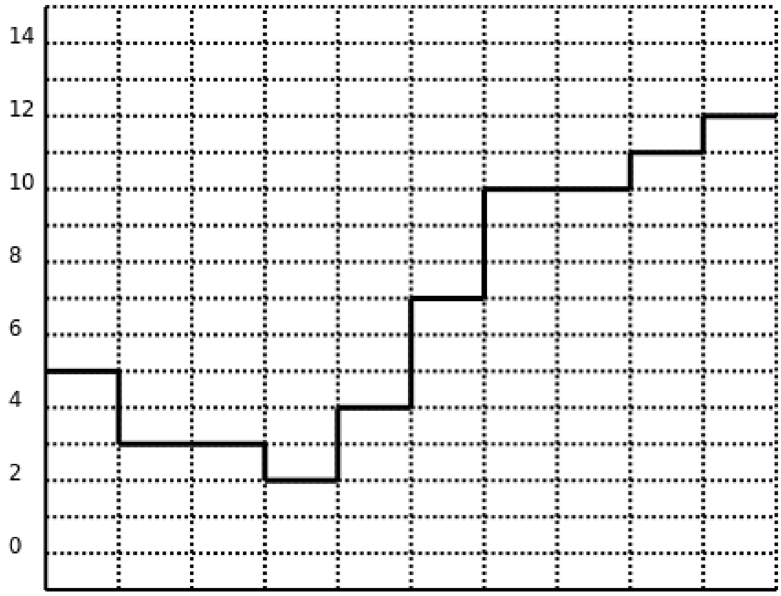
Assume that the number of code levels used for PCM in a telephone system is 16. The following sequence of bits are the PCM encoded digital data received by a destination telephone (assume no errors).

0101001100110010010001111010101010111100

b) On the figure below draw the output analog audio signal at the destination telephone. The horizontal dotted lines should be used as the levels, and the vertical dashed lines as the sample points. [3 marks]



Answer



In practice, the number of code levels used in telephone systems is normally 128 (instead of 16). Assume this is the case in the following parts.

c) If SIIT Bangkadi has a 1Mb/s link to SIIT Rangsit, how many PCM encoded voice calls can be sent from Bangkadi to Rangsit at the same time (ignore other overheads such as headers)? [3 marks]

Answer

8000 samples per second with each sample 7 bits gives a data rate required per voice call of 56kb/s. Therefore 17 voice calls can be transmitted at a time over a 1Mb/s link.

d) Some applications that transmit voice over the Internet (such as Skype, MSN) may use lower sampling rates and less code levels than above. Referring to part (c), that is the number of voice calls, explain an advantage and disadvantage of changing these values for voice applications [2 marks]

Advantage of lower sampling rates, less code levels

Answer

Leads to lower data rate required, meaning more voice calls can be sent over the same link/network at the same time.

Disadvantage of lower sampling rates, less code levels

Answer

Output generated at destination is lower quality, that is, low quality audio reproduction at the receiver.

Question 4 [16 marks]

Table 1 shows the list of codewords for a Hamming-distance based Forward Error Correction (FEC) scheme.

Data	Codeword
000	011011
001	100110
010	100111
011	010000
100	111100
101	001010
110	100101
111	001011

Table 1: Hamming-based FEC

a) For the following cases, explain the steps taken by the receiver (showing any calculations where necessary), and summarise the outcome by answering the 4 questions. [9 marks]

i. The data 010 is to be sent from transmitter to receiver. The 1st bit transmitted is in error (that is, the 1st bit transmitted is different from the 1st bit received).

Steps taken by receiver:

Codeword received by receiver: ___ ___ ___ ___ ___ ___

Error detected by receiver? YES NO

Data received: ___ ___ ___ (if applicable)

Is the correct data received? YES NO

Answer

Data 010 maps to codeword 100111. The codeword is transmitted, however because of the single bit error the receive codeword is 000111.

The receiver detects an error, and compares the received codeword to the valid codewords. The valid codeword with unique minimum Hamming distance to 000111 is 100111 (distance = 1). Therefore the receiver assumes the data received is 010. This is the correct assumption.

ii. The data 100 is to be sent from transmitter to receiver. The 1st and 2nd bits transmitted are in error.

Steps taken by receiver:

Codeword received by receiver: ___ ___ ___ ___ ___ ___

Error detected by receiver? YES NO

Data received: ___ ___ ___ (if applicable)

Is the correct data received? YES NO

Answer

Data 100 maps to codeword 111100. The codeword is transmitted, however because of the bit errors the receive codeword is 001100.

The receiver detects an error (the received codeword is invalid), and compares the received codeword to the valid codewords. There is no valid codeword with minimum Hamming distance (both 111100 and 001010 have distance of 2). Therefore error correction is not attempted, and the correct data is not received.

iii. The data 001 is to be sent from transmitter to receiver. The last bit transmitted is in error.

Steps taken by receiver:

Codeword received by receiver: ___ ___ ___ ___ ___ ___

Error detected by receiver? YES NO

Data received: ___ ___ ___ (if applicable)

Is the correct data received? YES NO

Answer

Data 001 maps to codeword 100110. The codeword is transmitted, however because of the bit error the receive codeword is 100111.

The receiver has received a valid codeword (and hence no error detected). It assumes the received data is 010, which is incorrect.

b) Assuming you must use a FEC with 3 bits of data and 6-bit codeword, explain how the scheme in Table 1 could be changed to reduce the possibility of single-bit errors being undetected. [3 marks]

Answer

Choose a different set of code words. For example, codewords for data 001 and 010 differ only by a single bit. Therefore if data 001 is transmitted, a single-bit error will be undetected if the received codeword matches that for 010. The codewords should have large Hamming distance between each other.

c) If using a link with data rate of 12Mb/s, what is the maximum possible throughput using the encoding scheme in Table 1? [2 marks]

Answer

To send 3 bits of data, 6 bits actually have to be transmitted, representing 50% efficiency.

Throughput is 6Mb/s.

d) Explain one advantage and one disadvantage of using an 8-bit codeword (instead of 6-bit codeword as in Table 1). [2 marks]

Advantage

Answer

Increases the chance to detect and correct errors

Disadvantage

Answer

Decreases the throughput

Question 5 [14 marks]

Consider a network with two links: A --- B --- C.

First consider the link from A to B with the following characteristics.

- DATA frame consists of 100 bits of header plus 9900 bits of data (total size 10,000 bits)
- ACK frame consists only of 100 bits of header
- Link data rate is 100Mb/s
- Link distance is 120km
- Link signal speed is 3×10^8 m/s

The Sliding Window flow control protocol is used on this link. The receiver (B) sends an ACK frame immediately after receiving a DATA frame (there is no processing delay). A 3-bit sequence number is used within the header of the DATA frame (and ACK frame).

a) What is the maximum number of DATA frames node A can send before having to wait for an ACK? [1 mark]

Answer

With a 3-bit sequence number the maximum window size is 2^3-1. Therefore 7 frames may be sent before having to wait for an ACK.

b) Assuming node A always has data ready to send, and it starts transmission of its 1st DATA frame at time 0s, at what time can node A start transmitting the 2nd DATA frame? [2 marks]

Answer

Immediately after the 1st DATA frame has been transmitted. The transmission time of a DATA frame is:

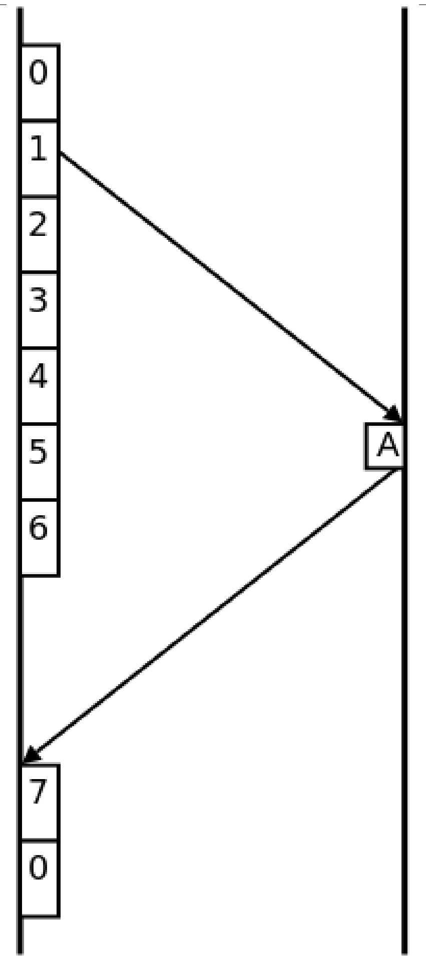
$$T_{DATA} = \frac{9900+100}{100 \times 10^6} = 100\text{us}$$

Therefore, 2nd DATA frame can be sent at time 100us.

c) At what time can node A start transmitting the 8th DATA frame? [4 marks]

Answer

As the maximum window size is 7, the transmitter must wait until at least the 1st DATA frame is ACKed before sending the 8th DATA frame. See the diagram:



The ACK transmission time and Propagation delay are:

$$T_{ACK} = \frac{100}{100 \times 10^6} = 1\text{us}$$

$$P_{AB} = \frac{120 \times 10^3}{3 \times 10^8} = 4 \times 10^{-4} = 400\text{us}$$

The time at which the ACK for the 1st DATA frame is received is: 901us

Note that the first 7 DATA frames are transmitted within 700us. Therefore the transmitter must wait until time 901us before sending the 8th frame.

d) What is the maximum throughput that can be achieved across the link from A to B? [3 marks]

Answer

7 DATA frames sent within 901us, gives a throughput of:

$$Throughput = \frac{7 \times 9900}{901 \times 10^{-6}} = 76.91\text{Mb}/s$$

Now consider the link from B to C with the following characteristics:

- DATA frame consists of 100 bits of header plus 9900 bits of data (total size 10,000 bits)
- ACK frame consists only of 100 bits of header
- Link distance is 1200m
- Link signal speed is 3×10^8 m/s

The Stop and Wait flow control protocol is used in this link.

e) What is the minimum data rate necessary for Link B to C such that the throughput from A to C is the same as calculated in part (d)? [4 marks]

Answer

At B, 7 DATA frames arrive every 901us. Therefore to achieve the same throughput from B to C, 7 DATA frames must be delivered to C every 901us. Considering Stop and Wait is used from B to C:

$$P_{BC} = \frac{1200}{3 \times 10^8} = 4\text{us}$$

Time for sending 7 DATA frames:

$$7 \times \left(\frac{10000}{Rate} + 2 \times P_{BC} + \frac{100}{Rate}\right) = 901\text{us}$$

$$Rate = \frac{10100}{\left(\frac{901}{7} - 8\right)} = 83.66\text{Mb}/s$$

The data rate necessary is 83.66Mb/s.

Question 6 [7 marks]

High Definition TV consists of a 1920 x 1080 pixel image, refreshed at a rate of 30 images per second. 24-bit colour is used – that is, each pixel is a 24-bit colour value.

a) What is the data rate required to transmit an uncompressed HDTV movie in real-time? [3 marks]

Answer

Data rate: 1920 x 1080 x 30 x 24 bits per second = 1492 Mb/s

b) Compression is often used to reduce the data rate necessary for sending video. Under some circumstances, HDTV can be compressed to 5% of its original size. What is the data rate required to transmit a compressed HDTV movie in real-time? [1 mark]

Answer

5% of 1492 = 75Mb/s

c) In a noise-free channel, what is the minimum number of signal levels necessary to transmit the compressed HDTV within 6MHz of bandwidth? (Note the number of signal levels must be a power of 2) [3 marks]

Answer

Using Nyquists theorem:

75Mb/s = 2 * 6MHz log (M)

M must be at least 128 (64 would be too small)

Question 7 [9 marks]

Table 2 shows a set of frames received by the Data Link layer of a computer (including the time when it is received). Each frame contains a header plus data. The amount of data in each frame is shown in the Data column. The Data Link layer header contains five fields:

1. A 16-bit timestamp, which indicates the time when the frame was sent
2. Address of the source, in the format of a 48-bit IEEE address
3. Address of the destination, in the format of a 48-bit IEEE address
4. A 32-bit sequence number
5. A 2 byte field to indicate the type of protocol used.

Time received [ms]	*Sequence number*	*Timestamp [ms]*	*Data [Bytes]*
7	0	0	100
11	1	3	120
14	2	6	150
16	3	9	125
22	4	12	100
23	5	15	125

Table 2: Frames Received

Answer the following questions considering only the frames in the table.

a) What is the average delay from source to destination? [2 marks]

Answer

Delay of packets are: 7, 8, 8, 7, 10, 8. Average delay is 48/6 = 8ms

b) What is the jitter between source and destination? [2 marks]

Answer

The difference between delays is: 1, 0, 1, 3, 2. Jitter is 7/5 = 1.4ms

c) What is the throughput for the received data? [2 marks]

Answer

There is 720 Bytes of data received over a period of 16ms. Throughput is 45000 Bytes/second or 360kb/s.

d) Consider the source sending the frames. Assume the Physical layer at the source adds an additional 80 bits to each frame. What is the average rate at which bits are sent by the source Physical layer? [3 marks]

Answer

Each frame contains data plus 20 byte Data Link layer header plus 10 byte Physical layer header. A total of 720 + 6x30 = 900 Bytes are sent over a period of 15ms, giving a sent rate of 60KB/s. Alternatively, we could say the frames are sent every 3ms. The average data sent per frame is 900/6 = 150B, therefore sent rate of 50KB/sec.