L-3/T-1/CSE **Date : 19/10/2019**

BANGLADESH UNIVERSITY OF ENGINEERING AND TECHNOLOGY, DHAKA

T-3/T-1 B. Sc. Engineering Examinations 2018 – 2019

Sub : **CSE 311** (Data Communication)

Full Marks : 210 Time : 3 Hours

The figures in the margin indicate full marks.

USE SEPARATE SCRIPTS FOR EACH SECTION

SECTION – A

There are **FOUR** questions in this section. Answer any **THREE**.

1. (a) A signal $g(t)$ band-limited to 50 Hz is uniformly sampled at intervals of 0.01 sec apart such that the following sampled values are found: **(21)**

$$g(0 \text{ sec}) = 1, g(\pm 0.01 \text{ sec}) = g(\pm 0.02 \text{ sec}) = g(\pm 0.03 \text{ sec}) = \ldots.. = 0$$

Find and draw the signal $g(t)$ and its Fourier transform (FT). Write down the characteristics of $g(t)$. Can you justify the FT of a Dirac delta function from $g(t)$ and its FT?

(b) Characterize a distortionless transmission system. Justify whether an All-Pass transmission system is distortionless. **(14)**

2. (a) Justify the relationship between the sampling frequency and the quality of a delta modulator. Why and how does a delta modulator replace the delay component with an integrator? How does the threshold of coding have impacts on slope overload and granular noise in delta modulation? **(20)**

(b) Show that the coefficients of the exponential Fourier series of an even symmetric and an odd symmetric periodic signal are real and imaginary, respectively. Also show that the coefficients can be calculated by integrating the signal over the half cycle only. **(15)**

3. (a) The imput stream to a 4B/5B block coder is 1100 0000 0000 0000 0011. What is the output stream? What are the lengths of the longest consecutive sequence of 0's in the input and the output streams? What will be the results of scrambling the input stream using B8ZS and HDB3 techniques? Assume that the last non-zero signal level has been positive and the number of non-zero pulses is odd after the last substitution. (Please see **Table for Q. No. 3(a)**). **(15)**

(b) Explain the main cause of aliasing which occurs during sampling a signal? What problems does aliasing introduce to a sampled signal? How are they compensated? **(20)**

4. (a) Establish the equation which shifts the spectrum of a signal $g(t)$ to a different frequency band. Explain how it is used to transmit multiple signals simultaneously over a shared channel. Also briefly introduce a dual technique that does the same job. **(18)**

(b) Prove that for an input signal $x(t)$, the output of a linear time invariant system with impulse response $h(t)$ is given by $y(t) = x(t)*h(t)$. Use ONLY time domain variables and equations. **(17)**

Contd P/2

Data Sequence	Encoded Sequence	Control Sequence	Encoded Sequence
0000	11110	Q (Quiet)	00000
0001	01001	I (Idle)	11111
0010	10100	H (Halt)	00100
0011	10101	J (Start delimiter)	11000
0100	01010	K (Start delimiter)	10001
0101	01011	T (End delimiter)	01101
0110	01110	S (Set)	11001
0111	01111	R (Reset)	00111
1000	10010		
1001	10011		
1010	10110		
1011	10111		
1100	11010		
1101	11011		
1110	11100		
1111	11101		

Table for Q. No. 3(a)

SECTION – B

There are **FOUR** questions in this section. Answer any **THREE**.

5. (a) An Amplitude Modulated (AM) wave is represented by the expression: **(10)**

$$v = 5[1 + 0.6\cos(6280t)].\sin(2\pi \times 10^4 t)$$

Calculate (i) Modulation depth, (ii) modulating frequency, (iii) period of carrier wave, (iv) peak instantaneous value of the modulated wave.

(b) Consider that one input to an AM DSBFC modulator is an 800 KHz carrier with an amplitude of 40 V. The second input is a 25 KHz modulating signal whose amplitude is sufficient to produce a ± 10V change in the amplitude of the envelope. Calculate (i) upper and lower side band frequencies, (ii) Modulation coefficient and percentage modulation, (iii) Maximum and minimum positive amplitude of the envelope. In addition, draw the envelope and the output frequency spectrum. **(15)**

(c) Show that in amplitude modulation, maximum average power transmitted by an antenna is 1.5 times the carrier power. Also, show that for 100% modulation, each sideband contains one-sixth of the total power. **(10)**

6. (a) Why is it possible to exceed 100 percent modulation in frequency modulation, but not in amplitude modulation? Derive the mathematical expression for single tone frequency modulation. **(12)**

CSE 311

Contd..... Q. No. 6

(b) What is the bandwidth required for an FM signal if the modulating frequency is 1 KHz and the maximum deviation is 10 KHz. What is the bandwidth required for a DSBFC amplitude modulation? **(8)**

(c) What is the Carson's rule? Use Carson's rule to compare the bandwidth that would be required to transmit a baseband signal with a frequency range from 300 Hz to 3 KHz using (i) NBFM with maximum deviation of 5 KHz (ii) WBFM with maximum deviation of 75 KHz. **(7)**

(d) In an FM system, the audio frequency is 1 KHz and audio voltage is 2V. The deviation is 4 KHz. If audio voltage is now increased to 8 volts and audio frequency is dropped to 500 Hz, find the modulation index in each case and the corresponding bandwidth. **(8)**

7. (a) Draw the constellation diagram of quadrature phase shift keying (QPSK). Write the mathematical expression of a QPSK signal. Explain the working operation of an offset QPSK (0 QPSK) transmitter. **(10)**

(b) Draw the geometric representation of M-ary PSK signals and find the distance between signal points. Also find the bandwidth of M-ary PSK. **(10)**

(c) Given an 8-level PSK signal that employs two different amplitudes as shown in figure 7(c). Calculate the distance between two nearest signal points and compare with the minimum distance for standard 8-level PSK. Which scheme is better in terms of error probability? **(10)**

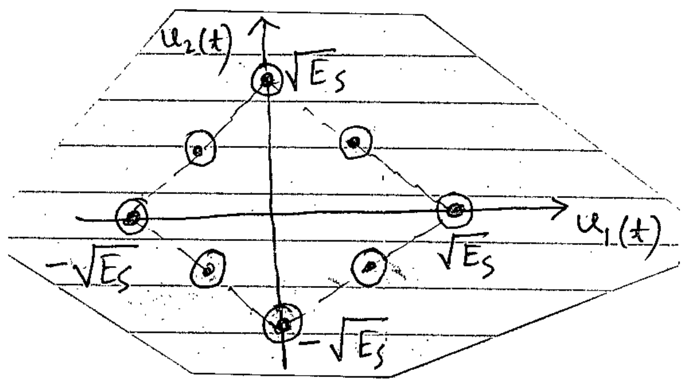


Figure for Q. 7(c)

(d) Assume that you are required to transmit f_b = 90 Mb/S data in an authorized bandwidth of 20 MHz using M-ary PSK. Find the number of bits per symbol. **(5)**

8. (a) Write the mathematical expression of amplitude shift keying (ASK) waveform. Why is bandpass filter used in generation of ASK waveform? **(6)**

(b) What are the desirable characteristics of a line coding scheme? Describe the Manchester and the differential Manchester line coding techniques. **(10)**

(c) Consider a binary sequence: 11111011111. Draw the waveforms for (i) Unipolar NRZ, (ii) Bipolar NRZ, (iii) AMIRZ.

Discuss the advantages and disadvantages of the three signaling formats. **(10)**

(d) Explain why BPSK signaling scheme is better than BFSK signaling in terms of error probability? **(9)**

L-3/T-1/CSE **Date: 01/09/2018**

BANGLADESH UNIVERSITY OF ENGINEERING AND TECHNOLOGY, DHAKA

L-3/T-1 B. Sc. Engineering Examinations 2017-2018

Sub: **CSE 311** (Data Communication)

Full Marks: 210 Time: 3 Hours

The figures in the margin indicate full marks

USE SEPARATE SCRIPTS FOR EACH SECTION

SECTION – A

There are **FOUR** questions in this section. Answer any **THREE**.

All the symbols have their usual meanings unless explicitly mentioned.

1\. (a) Why are we interested in frequency domain representations of signals? How do we obtain frequency spectrum of a signal from its time domain representation? Write the corresponding formula. What is the difference between frequency spectrum of a periodic signal and frequency spectrum of an aperiodic signal? Explain graphically using the examples of a rectangular pulse and a periodic rectangular pulse train. Does being periodic or aperiodic have any effect on the bandwidth of a signal? Explain using the same examples, i.e., a rectangular pulse and a pulse train of the same pulse duration, τ. **(4+3+6+5)**

(b) Let $G(f)$ denotes the Fourier transform of the rectangular pulse $g(t)$ in Figure (a) below. Now, using properties of Fourier transform, express the Fourier transform of the triangular pulse $h(t)$ in Figure (b) in terms of $G(f)$. Show all steps of calculation. **(10)**

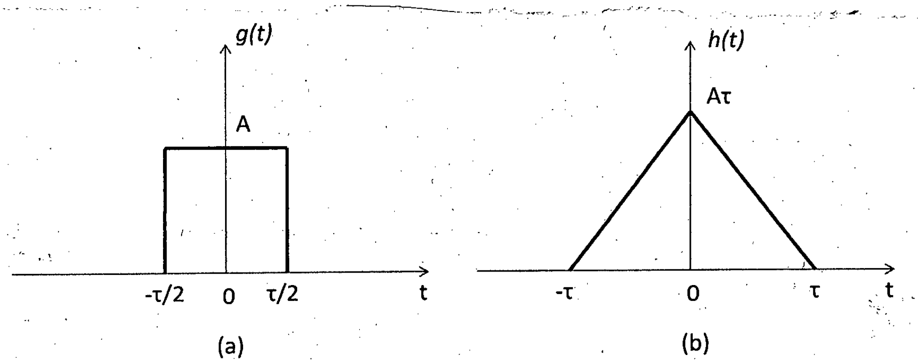


Figure for Question 1(b)

(c) Write the transfer function of an ideal low-pass filter. From the transfer function, derive the impulse response, in time domain, of the ideal low-pass filter. **(2+5)**

2\. (a) What is power-bandwidth trade-off? Demonstrate power-bandwidth trade-off with an example. **(4+6)**

(b) Show the Signal-to-Noise Ratio (SNR) of a uniformly spaced quantizer is proportional to input signal power. Assume all values in the amplitude range of the quantizer are equally likely. **(10)**

Contd P/2

CSE 311

Contd... Q. No. 2

(c) A PCM system implements μ-law companding where μ = **255**. Total number of levels in the quantizer is L = **256**. Now, for each of the following sample values: **–2.4, 0.6 and 5**, what will be the value received after expansion. The μ-law (for positive amplitude) is given by: **(15)**

$$y = \frac{1}{\ln(1+\mu)} \ln\left(1 + \frac{\mu m}{m_p}\right), \qquad o \le \frac{m}{m_p} \le 1$$

Assume $m_p = 5$. Show detailed calculation steps.

3. (a) Draw a block diagram of the transmitter of a DPCM (Differential Pulse Code Modulation) system. Delta Modulation (DM) system is a special case of DPCM. What are the special features of a DM system with respect to a general DPCM system? What is the advantage of using a DM system? **(5+3+3)**

(b) Given a sequence of samples $\{g_0, g_1, \cdots g_{N-1}\}$ of an analog signal *g(t)*, the Discrete Fourier transform (DFT) of the sequence is defined as: **(10+8+6)**

$$G_k = \sum_{n=0}^{N-1} g_n e^{-j\frac{2\pi}{N}kn}, \quad k = 0, 1, \ldots, N-1$$

Here the sequence $\{G_0, G_1, \cdots G_{N-1}\}$ is called the transform sequence.

Now formulate a *decimation-in-frequency* FFT (Fast Fourier Transform) algorithm to efficiently computer the DFT of a finite data sequence. Draw the complete *signal-flow* graph corresponding to the computation of an 8-point DFT (i.e., N = 8) using your algorithm. If *in-place computation* is used for space efficiency in such an algorithm, then the samples of the transform sequence G_k are stored in a *bit-reversed order*. Explain using the *signal-flow* graph drawn, what *bit-reversed order* means.

4. (a) Describe two causes of asynchronicity among incoming channels of a TDM (Time Division Multiplexing) multiplexer. Describe how this problem of asynchronicity is resolved in a TDM system. **(6+9)**

(b) What is 'Digital Line Coding' technique? Describe six desirable characteristics of a line coding technique. For each of the characteristics, give examples: names and encoding rules of two line coding schemes such that one is better than the other with respect to that characteristic. Justify your examples with proper explanation. **(2+18)**

Contd P/3

CSE 311

SECTION – B

There are **FOUR** questions in this section. Answer any **THREE**.

5. (a) Mention the transmission bandwidths required for DSB, QAM and SSB modulation techniques. **(3+5+5+5+5=23)**

   For a base signal $m(t) = \cos 100t \times \cos 200t$

   (i) Draw the spectrum of $m(t)$.

   (ii) Draw the spectrum of DSB-SC for $2m(t)\cos 100t$.

   (iii) Draw the spectrum of DSB+C for $2(1+m(t))\cos 1000t$.

   (iv) Draw the spectrum of LSB-SC from the figure you draw for Question (ii)

   (b) How does the value of RC affect Envelope Detection in DSB-WC? Mention the condition imposed on the value of RC. **(8+4=12)**

6. (a) Define Tone. What do you mean by Tone Modulation? **(3)**

   (b) Draw a detailed block diagram for SSB Modulation (Upper Side band only). **(15)**

   (c) Write the modulation equations for Frequency Modulation and Phase Modulation. Which one is better in practice? Draw the block diagrams for converting a Frequency Modulator into a Phase Modulator and vice versa. **(8+2+7=17)**

7. (a) Define PSK and DPSK. What is the difference between the two? **(6+2=8)**

   (b) For Vestigial Sideband Modulation, derive the following equation for an equalizer filter: $H_o(f) = \dfrac{1}{H_i(f+f_c) + H_i(f-f_c)}$ (for an input filter $H_i(f)$, and carrier frequency f_c). Draw the figures necessary for derivation. **(27)**

8. (a) Draw a detailed block diagram for a Nonlinear DSB-SC Modulator. **(15)**

   (b) Mention the bandwidth requirements for NBFM and WBFM. Draw the block diagram for constructing an NBFM Modulator using a DSB-SC Modulator. **(5+15=20)**

L-3/T-1/CSE **Date: 08/08/2017**

BANGLADESH UNIVERSITY OF ENGINEERING AND TECHNOLOGY, DHAKA

L-3/T-1 B. Sc. Engineering Examinations 2016-2017

Sub: **CSE 311** (Data Communication I)

Full Marks: 210 Time: 3 Hours

USE SEPARATE SCRIPTS FOR EACH SECTION

The figures in the margin indicate full marks

SECTION – A

There are **FOUR** questions in this section. Answer any **THREE**.

1. (a) Define bandwidth of a signal. Sketch two time-domain signals such that bandwidth of one signal is greater than the other. Justify your answer using the definition of bandwidth **(4+5=9)**

(b) Prove that for a real-time signal, the amplitude spectrum is an even function and the phase spectrum is an odd function. **(6)**

(c) Let $\Delta(\frac{t}{\tau})$ represents a triangular pulse as shown in the figure below. The Fourier transform of $\Delta(\frac{t}{\tau})$ is given as: $\Delta(\frac{t}{\tau}) \Leftrightarrow \frac{\tau}{2}\operatorname{sin}c^2(\frac{2\pi f\tau}{4})$. Now, using duality property, find the Fourier transform of the time function: $\operatorname{sinc}^2(t)$ in terms of the function $\Delta(.)$. Also, draw the spectrum. **(6)**

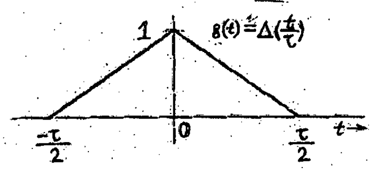


Figure for Question 1(c)

(d) Give a proof for why an ideal low pass filter is physically unrealizable. **(8)**

(e) Consider a Linear Time Invariant (LTI) system that functions as a differentiator, i.e., given a signal g(t) as input, its output will be $\frac{d}{dt}$ g(t). Derive and write down the transfer function of the system. **(6)**

2. (a) List three important advantages digital communication over analog communication. **(5)**

(b) State and prove Nyquist's sampling theorem. In light of the proof, explain why in practice – (i) a signal is passed through a band-Limiting filter before sampling, and (ii) sampling rate is higher than the Nyquist rate. **(10+6=16)**

CSE 311

Contd ... Q. No. 2

(c) Describe how companding achieves non-uniform quantization without actually changing the uniform level spacing in the quantizer. Explain intuitively how a quantizer with companding causes Signal-to-Noise Ratio (SNR) to be independent of input power. **(9+5=14)**

3. (a) Let G (f) be the Fourier transform of a real-time signal g(t). In discrete Fourier Transform (DFT), we compute G(f) at some finite frequencies from the samples of g(t). Let g(t) be sampled over the duration $0 \leq t \leq T_0$ at uniform intervals of T_s seconds resulting in N_0 samples. Then the formula for computing DFT is as follows: **(4+5+5+6)**

$G_q = \sum_{k=0}^{N_0-1} g_k e^{-jq\Omega_0 k}$, where $g_k = T_s g(kT_s)$, $G_q = G(qf_0)$, $\Omega_0 = 2\pi f_0 T_s$

Now answer the following.

(i)Derive the above formula from the expression of G(f).

(ii) Prove that, $f_0 = \dfrac{1}{T_0}$.

(iii) Compute DFT for the following sample values of g(t). Write down the values of sample frequencies, f and computed values of G(f).

t	0	1	2	3
g(t)	1	2	3	4

(iv) What considerations should be made while selecting vales for T_s and T_o? Explain.

(b) How does a good 'predictor' design improve the performance of a Differential Pulse Code Modulation (DPCM) system? Explain. **(6)**

(c) Consider a signal, m(t) = Acosωt is input to a Delta Modulation (DM) system. What should be the step size, E such that no slope overload occurs in the system for m(t)? How the step size can be made adaptive in a DM system? **(5+4)**

4. (a) Give a real life example where Time Division Multiplexing (TDM) is used. Why pulse/bit stuffing may be required in a TDM multiplexer? Explain the difference between positive pulse stuffing and negative pulse stuffing? How does he receiver detect a stuffed bit in each case of positive and negative pulse stuffing? **(2+5+6+4)**

(b) Why does 'Inter-symbol Interference (ISI)' occur in a digital transmission system? Give an example of a pulse of bandwidth B Hz that can be used to transmit 2B pulses per second in presence of ISI,. Explain how it works. How such a pulse can be generated using a transversal filter? Explain using a block diagram and necessary illustrations. **(4+8+6)**

CSE 311

SECTION – B

There are **FOUR** questions in this section. Answer any **THREE**.

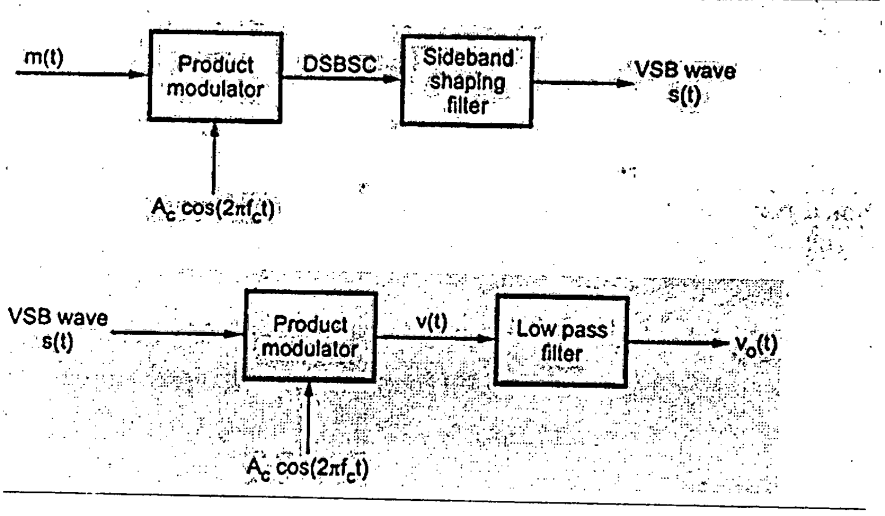
5. (a) Suppose, an AM signal is described by $[A+m(t)]\cos\omega_c t$, Where $A \gg m(t)$. Show that, the message m(t) can be recovered at the receiver from this AM signal by squaring it and then passing the resulting signal through a low-pass filter. **(15)**

   (b) Explain the necessity of Costas phase-locked loop in the demodulation of DSB-SC signals with a suitable example. **(10)**

   (c) Explain the operation of Costas phase-locked loop with approximate block diagram. **(10)**

6. (a) Derive the time domain representation of SSB signal with appropriate figures. Then based on this representation, design a SSB modulation technique using block diagram. **(15)**

   (b) The modulator and demodulator for VSB signal are shown below: **(20)**



   Now derive the required characteristics of the transfer function $H(f)$ of the sideband shaping filter of modulator with appropriate figures

7. (a) Derive the bandwidth estimation of WBFM signal with appropriate figures using staircase approximation approach. **(15)**

   (b) Consider a FM signal $\cos\left[\omega_c t + k_f \int_0^t m(\alpha)d\alpha\right]$.

   Draw a block diagram showing necessary components to convert the signal into

   $\cos\left[\omega_c t + 3k_f \int_0^t m(\alpha)d\alpha\right]$. **(20)**

   And also for each component, clearly mention its function along with corresponding input and output signals.

CSE 311

8. (a) What is 'Digital Line Coding'? Describe using examples how digital line coding techniques determine the following characteristics of transmitted signals. **(15)**

(i) Bandwidth consumption

(ii) Power consumption

(iii) Synchronization Capability

(iv) DC value suppression capability

(b) Mathematically prove that a FM modulator can be constructed using a PM modulator. Draw the block diagram of such FM modulator. **(10)**

(c) How can we improve the spectral efficiency of amplitude modulation using Quadrature Amplitude Modulation (QAM) technique? Explain with the block diagram of appropriate modulator and demodulator. **(10)**

BANGLADESH UNIVERSITY OF ENGINEERING AND TECHNOLOGY, DHAKA

L-4/T-2 B. Sc. Engineering Examinations 2014-2015

Sub : **CSE 311** (Data Communication)

Full Marks: 210 Time : 3 Hours

The figures in the margin indicate full marks.

USE SEPARATE SCRIPTS FOR EACH SECTION

SECTION – A

There are **FOUR** questions in this section. Answer any **THREE**.

Give your answer to the point avoiding unnecessary writings.

Try to draw as many figures as possible and explain with block diagrams.

1. (a) Briefly explain the steps required for "Echo Cancellation"? Explain how "Inter-Symbol Interference (ISI)" can occur? **(7+6)**

   (b) Briefly explain the procedure of 4B/5B encoding scheme and intentions behind it. **(10)**

   (c) Show with necessary figures how 4-bit information is passed at a time with 16-QAM (Quadrature Amplitude Modulation). **(6)**

   (d) For a baseband signal of $m(t) = \cos 100t + \cos 400t$ **(6)**

   (i) Draw basic spectrum of $m(t)$

   (ii) Draw spectrum of the modulated signal using Double Side Band with Carrier (DSB+C), where the carrier signal is $2(1 + m(t))\cos 1000t$. **(6)**

2. (a) Draw digital signals of the corresponding binary sequence of 1011001 for the following line encoding schemes, **(9+6)**

   (i) Pseudo-Ternary

   (ii) Differential Manchester

   (iii) Multi-transition (MLT)-3

   Summarize each of these in terms of Bandwidth, Baseline wandering, and DC component issues.

   (b) Explain the incoherent demodulation procedure of FM and FSK. Explain how modulations of Phase Shift Keying (PSK) and Quadratic Amplitude Modulation (QAM) relate to each other? **(5+3)**

   (c) Explain how offset QPSK (Quadrature Phase Shift keying) reduces maximum phase shift from π to $\frac{\pi}{2}$ without reducing bit rate. Explain how phase continuation is maintained during MSK (Minimum shift keying). **(6+6)**

CSE 311

3. (a) Show that, for a real time $f(t)$, the signal $f_{USB}(t) = f(t)\cos\omega_c t - f_h(t)\sin\omega_c t$ represents an upper sideband SSB-SC signal, where $f_h(t)$ represents Hilbert Transform of $f(t)$ and ω_c is the carrier frequency in rad/sec. **(13)**

(b) What is orthogonal signal? Explain the motivations behind making each m-FSK signals orthogonal to each other? Show that the minimum frequency difference to make signal orthogonal is, $\delta_f = \frac{1}{2T_b} Hz$. **(2+4+6)**

(c)What was the fallacy behind the estimation of the bandwidth of Frequency Modulation (FM)? Show that the bandwidth requirement of Wide Band Frequency Modulation (WBFM) is $\approx 2(\Delta f + 2B)\, Hz$, where Δf refers to peak frequency deviation. **(4+6)**

4. (a) Explain with diagram how Phase Locked Loop (PLL) maintains Phase and Frequency synchronization between source and generated carrier signals. **(15)**

(b) For a Vestigial Side Band (VSB) modulation with suppressed carrier, derive following relation between input and output filter, $H_o = \frac{1}{H_i(\omega+\omega_c)+H_i(\omega-\omega_c)}$. **(8)**

(c) Explain with necessary figures and diagrams, the modulation and demodulation techniques of On Off Keying (OOK), Frequency Shift Keying (FSK), and Differential Phase Shift Keying (DPSK). **(4+4+4)**

SECTION-B

There are **FOUR** questions in this section. Answer any **THREE**.

5. (a) There are two principal forms of digital switching in communication networks, **(10)**

   (i) Circuit switching

   (ii) Packet switching

Big website companies like Facebook and Google store their data in data centers where thousands of computers act together as a single server. Also, they establish multiple data centers in geographically distance locations. That means when you load your Facebook home page, it may be the case that the picture of Aashik Salam, one of your computer science teachers, was downloaded from the Facebook data center at Singapore and the picture of Himel Sen, another of your computer science teachers, was downloaded from the data center at California. However, it is not just the case that only clients communicate with the data center servers, data centers communicate with each other a lot as well. What form of digital switching do these data centers use to communicate with each other?

(b) State and prove the dilation property of the Fourier transform. **(5+5)**

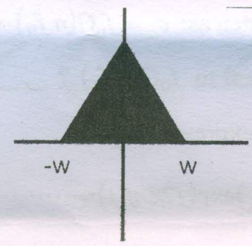
(c) If the Fourier Transform of the signal $g(t)$ is $G(f)$, then determine the Fourier Transform of the signal, $\sin^2(2\pi f_c t)g(t)$. **(10)**

(d) Derive the Fourier Transform of the Dirac Delta Function. **(5)**

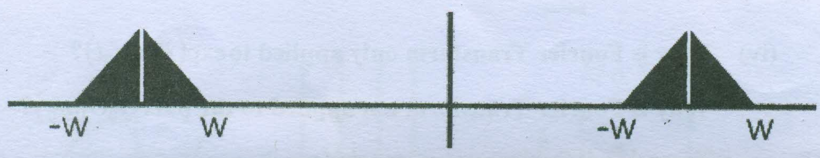
CSE 311

6. (a) Determine the bandwidth of the following two signals. The frequency domain representation of the signals are drawn as follows. Also tell what will be the bandwidth if the diagrams drawn are for time domain representation. **(3+3+4)**

(i) Signal One

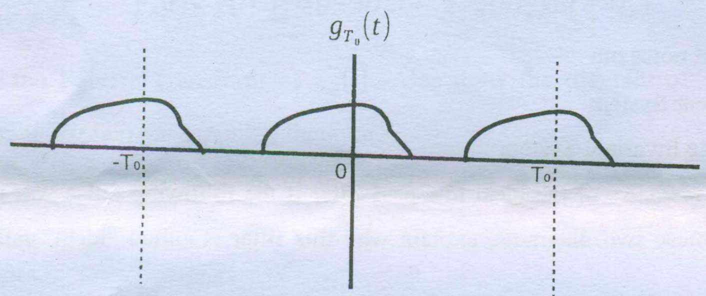


(ii) Signal Two



(b) Derivation of the Fourier Transform of periodic signals are given below. Read the derivation given below and write answer the questions: **(20)**

Let $g_{T_o}(t)$ be a periodic signal with period T_o. The diagram and the definition of $g_{T_o}(t)$ regarding complex exponential Fourier Series are given below,



$$g_{T_o}(t) = \sum_{n=-\infty}^{\infty} c_n \exp(j2\pi n f_0 t) \cdots\cdots\cdots (1)$$

where c_n is defined as

$$c_n = \frac{1}{T_0}\int_{-\frac{T_0}{2}}^{\frac{T_0}{2}} g_{T_0}(t)\exp(-j2\pi n f_0 t)\,dt \cdots\cdots\cdots (2)$$

Now, a pulse like function $g(t)$ is defined in terms of $g_{T_0}(t)$ as,

$$g(t) = \begin{cases} g_{T_0}(t) & ; -\frac{T_0}{2} \le t \le \frac{T_0}{2} \\ 0 & ; \text{elsewhere} \end{cases} \cdots\cdots\cdots (3)$$

Contd... Q. No. 6(b)

(i) **Why do we need to introduce this new function $g(t)$?** [Given: the Fourier Transform of $g(t)$ if $G(f)$]

(ii) **Also, Draw the time domain diagram of $g(t)$.**

Now c_n can be written as: $c_n = f_0 G(n f_0)$ $\cdots\cdots\cdots\cdots$ (4)

(iii) **How can c_n be equal to $f_0 G(n f_0)$?**

Now $g_{T_0}(t)$ can be expressed as:

$$g_{T_0}(t) = \sum_{n=-\infty}^{\infty} f_0 G(nf_0) \exp(j2\pi n f_0 t) \cdots\cdots\cdots\cdots (5)$$

and the Fourier Transform of $g_{T_0}(t)$ would be,

$$F\left[g_{T_0}(t)\right] = \sum_{n=-\infty}^{\infty} f_0 G(nf_0) F\left[\exp(j2\pi n f_0 t)\right] \cdots\cdots\cdots\cdots (6)$$

(iv) **Why is Fourier Transform only applied to $\exp(j2\pi n f_0 t)$?**

Why is Fourier Transform not applied to the part $f_0 G(nf_0)$?

From this definition we can conclude,

$$F\left[g_{T_0}(t)\right] = f_0 \sum_{n=-\infty}^{\infty} G(nf_0) \delta(f - nf_0) \cdots\cdots\cdots\cdots (7)$$

(v) **How can we deduce Eq. 7 from Eq. 6?**

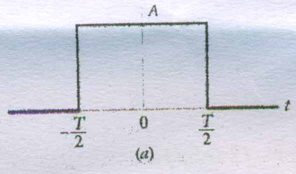
(c) Why is non-uniform quantization needed? What are the problems in case of uniform quantization? **(5)**

7. (a) Write short notes on- **(5+5)**

(i) Linear System

(ii) Time Invariant System

(b) Draw the response of the ideal low-pass filter if the following pulse was given as input. From these two diagrams, explain why this filter is called "ideal" rather than "real"? **(10)**

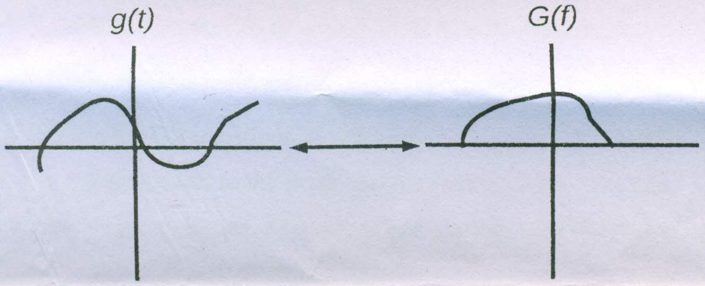


(a)

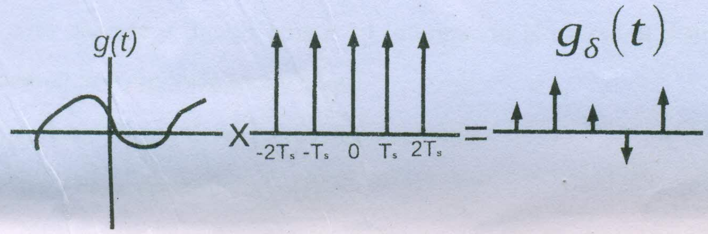
(c) Deduce the recurrence relationship for the Fast Fourier Transform. **(15)**

CSE 311

8. (a) Suppose a Fourier Transform pair signal is given as follows: **(6+3+6)**



The signal is sampled as described in the following figure,



The sampled signal is described as,

$$g_\delta(t) = g(t)\sum_{n=-\infty}^{\infty}\delta(t-nT_s) = \sum_{n=-\infty}^{\infty} g(nT_s)\delta(t-nT_s)$$

Find the Fourier Transform of $g_\delta(t)$. Also draw the diagram of $G_\delta(f)$. From the diagram, explain the sampling theorem.

(b) Draw the block diagram of the regenerative repeater component of Pulse code modulation. Also explain how each of the sub-components work in the regenerative repeater. **(10)**

(c) What are the two types of error in case of delta modulation? Illustrate each of them with figures. Also, describe how these two type of errors can be overcome. **(10)**

L-3/T-1/CSE **Date : 16/01/2016**

BANGLADESH UNIVERSITY OF ENGINEERING AND TECHNOLOGY, DHAKA

L-3/T-1 B. Sc. Engineering Examinations 2014-2015

Sub : **CSE 311** (Data Communication I)

Full Marks : 210 Time : 3 Hours

The figures in the margin indicate full marks.

USE SEPARATE SCRIPTS FOR EACH SECTION

SECTION – A

There are **FOUR** questions in this section. Answer any **THREE**.

All the symbols have their usual meanings unless explicitly mentioned.

1. (a) How is the bandwidth of a signal calculated? Accordingly calculate the bandwidth of the unit impulse function, δ(t) defined below. **(5+7)**

$$\delta(t) = 0 \qquad t \neq 1$$

$$\int_{-\infty}^{\infty} \delta(t)\, dt = 1$$

(b) Explain 'Linearity' and 'Time invariance' properties of a 'Linear Time Invariant (LTI)' system. **(10)**

(c) A low-pass filter transfer function H(f) is given by **(13)**

$$H(f) = \begin{cases} (1 + k\cos 2\pi fT)\, e^{-j2\pi f t_d} & |f| < B \\ 0 & |f| > B \end{cases}$$

A pulse g(t) band-limited to B Hz is applied at the input of the filter. Show that the output

$$y(t) = g(t - t_d) + \frac{k}{2}[g(t - t_d - T) + g(t - t_d + T)$$

2. (a) List three advantages of digital communication over analog communication. **(5)**

(b) A signal band limited to 4000 Hz is transmitted by a PCM system. The signal is sampled at twice the Nyquist rate and the sampled pulses are passed through a uniformly spaced quantizer with sixteen levels. Finally, the quantized samples are encoded as binary signals. Calculate the output data rate (in bits/sec) of the PCM system. **(8)**

(c) Explain the 'Aliasing effect' that occurs during sampling of practical signals. How can 'Aliasing effect' be eliminated? **(8+2)**

(d) Consider a PCM system that implements μ-law companding with $\mu = 255$. Total number of levels in the quantizer is $L = 256$. Now, what will be the total root-mean-square (rms) quantization error for the following set of sample values of some signal: {–2.4, 0.6, 5}. Assume the maximum and the minimum values of the signal are 5 and –5, respectively. Show detailed calculation steps. **(12)**

Contd P/2

CSE 311

3. (a) Draw the block diagrams of transmitter and receiver of a Differential PCM (DPCM) system. Draw the block diagram of a linear predictor for DPCM. **(8+5)**

   (b) Draw the block diagrams of the modulator and the demodulator of a Delta Modulation (DM) system that use integrator-amplifiers and comparators. **(8)**

   (c) What is 'slope overload noise' in a Delta Modulation (DM) system? How to determine the step size of a DM modulator such that 'slope overload' does not occur for a given input signal m(t)? **(5+9)**

4. (a) What is Time Division Multiplexing (TDM)? Describe briefly how the following two issues are addressed in a system using TDM: (i) Synchronization between the sender (multiplexer) and the receiver (demultiplexer), (ii) asynchronicity among incoming channels in the multiplexer. **(3+6+6)**

   (b) Why does 'Inter-Symbol Interference(ISI)' occur in a digial transmission system? Explain how Nyquist first criterion for pulse shaping can be used to eliminate ISI. Does the pulse $p(t) = \text{sinc}(\pi R_b t)$ satisfy the criterion? If so, what is the maximum pulse rate achievable with p(t)? Explain. Accordingly, how much bandwidth (in Hz) would be required to transmit 4000 pulses per second using p(t). Given, bandwidth of p(t) is $\frac{R_b}{2}$. **(5+6+4+3+2)**

SECTION – B

There are **FOUR** questions in this section. Answer any **THREE**.
Give your answer to the point avoiding unnecessary writings.
Try to draw as many figures as possible and explain with block diagrams.

5. (a) For a base signal of $m(t) = \cos 100t \times \cos 200t$ **(3+3+3+3+3=15)**

   (i) Draw spectrum of $m(t)$

   (ii) Draw spectrum of DSB-SC for $2m(t)\cos 100t$

   (iii) Draw spectrum of DSB+C for $2(1 + m(t))\cos 1000t$

   (iv) Draw spectrum of LSB-SC from the figure of (ii).

   Just mention the transmission bandwidth required for DSB-SC, DSB+C and VSB+C modulation techniques.

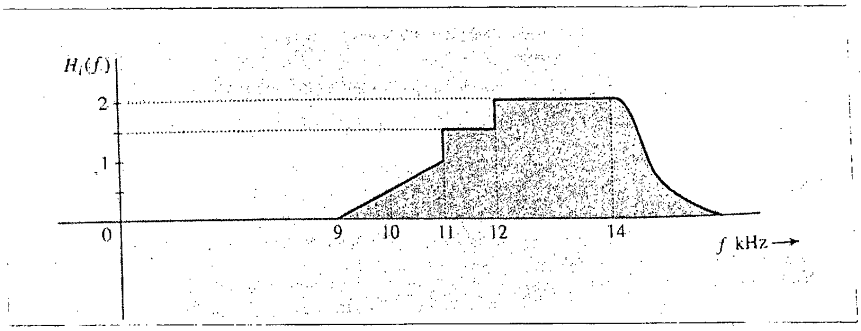
   (b) Show that the bandwidth requirement of WBFM is $\approx 2(\Delta f + 2B)$ *Hz*, where Δf refers to peak frequency deviation. What was the fallacy behind bandwidth estimation of FM? **(8+5)**

   (d) Explain the modulation and demodulation procedure of Differential Phase Shift Keying (DPSK)? **(7)**

Contd P/3

CSE 311

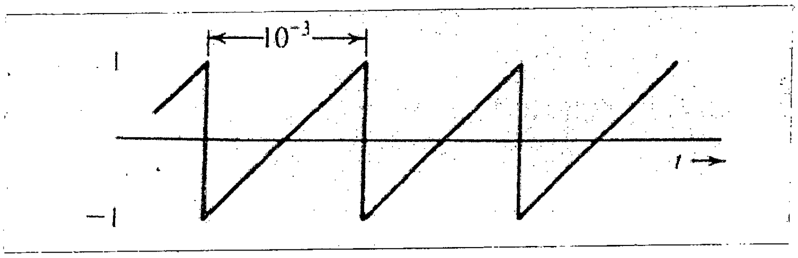
6. (a) A vestigial filter $H_i(f)$ has a transfer function shown below. The carrier frequency is $f_c = 10$ kHz and the baseband signal bandwidth is 4 kHz. Draw the corresponding transfer function of the equalizer filter, $H_o(f)$. [Hint: $H_o(f) = \dfrac{1}{H_i(f+f_c)+H_i(f-f_c)}$.] **(10)**



(b) Show that for M-ary FSK, the minimum frequency separation to ensure orthogonality among the FSK symbols is: $\delta f = \dfrac{1}{2T_b}$, where T_b denotes the symbol duration. **(11)**

(c) Show that, for a real time function f(t), the signal, $f_{LSB}(t) = f(t)\cos\omega_c t + f_h(t)\operatorname{sing}\omega_c t$ represents a lower sideband SSB-SC signal $f_h(t)$ represents Hilbert Transform of f(t) and ω_c is the carrier frequency in rad/sec. **(14)**

7. (a) A baseband signal m(t) is shown below. Sketch the $\varphi_{FM}(t)$ and $\varphi_{PM}(t)$ for this signal m(t) if $\omega_c = 2\pi \times 10^6$, $k_f = 2000\pi$ and $k_p = \dfrac{\pi}{2}$. (Mention only the maximum and minimum frequency. No need to put extra effort on drawing.) **(10)**



(b) What is Pilot Carrier? Explain with diagram how Phase Locked Loop (PLL) maintains Phase and Frequency synchronization? **(2+10)**

(c) Explain with diagram how 4-bit information is passed at a time with 16-QAM? **(10)**

(d) Describe a trivial demodulation procedure of FM signal? **(3)**

Contd P/4

CSE 311

8. (a) Draw the signals obtained after encoding the bit stream '10010110' by each of the following digital line coding techniques, **(6+9)**

   – Polar NRZ-L

   – Manchester

   Analyze these techniques in terms of the following characteristics.

   (i) Synchronization capability

   (ii) Bandwidth requirement

   (iii) DC value suppression

   (b) Explain how offset QPSK reduces maximum phase shift from π to $\frac{\pi}{2}$ without reducing bit rate? Explain how phase continuation is maintained during MSK, what are the prerequisites for carrier frequency to work with MSK. **(6+8)**

   (c) What is baseline wandering? Describe how 'Scrambling' solves the synchronization problem with Bipolar-AMI. **(2+4)**

L-3/T-1/CSE Date : 04/08/2015

BANGLADESH UNIVERSITY OF ENGINEERING AND TECHNOLOGY, DHAKA

L-3/T-1 B. Sc. Engineering Examinations 2013-2014

Sub : **CSE 311** (Data Communication I)

Full Marks : 210 Time : 3 Hours

The figures in the margin indicate full marks.

USE SEPARATE SCRIPTS FOR EACH SECTION

SECTION – A

There are **FOUR** questions in this section. Answer any **THREE**.

All the symbols have their usual meanings unless explicitly mentioned.

1. (a) What are the advantages of using digital signals over analog signals? Explain. **(5)**

(b) (i) How is the bandwidth of a signal calculated? **(4+8)**

(ii) The time-scaling property of Fourier Integral is as follows:

If $g(t) \Leftrightarrow G(f)$, then for any real constant a, $g(at) \Leftrightarrow 1/|a|\, G(f/a)$.

Now which of $g(t)$ and $g(at)$ has higher bandwidth when $a > 1$? Explain your answer

(c) (i) For a Linear Time Invariant (LTI) continuous time system, an input $x(t) \Leftrightarrow X(f)$ and the corresponding output, $y(t) \Leftrightarrow Y(f)$ are related as follows: $Y(f) = H(f)X(f)$ where H(f) is called the transfer function of the system. Now given an LTI system, how would you empirically find out the transfer function of the system? Explain your answer. **(8)**

(ii) For a 'Distortionless' LTI system, the amplitude response $|H(f)|$ must be constant and the phase response $\theta(f)$ must be a linear function of f going through the origin, $f = 0$ Now, under what conditions, the RC circuit in Figure 1(c) can be approximated as a Distortionless system. **(10)**

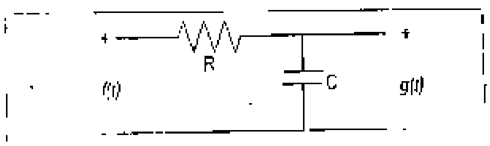


Figure 1(c)

2. (a) State and prove Nyquist's sampling theorem. **(3+10)**

(b) What are the practical issues in signal sampling and reconstruction according to the proof you have described above? How are those issues handled in practice? **(3+7)**

(c) Explain how quantization with non-uniform spacing of levels can be achieved without actually changing the uniform level spacing in the quantizer of a PCM system? **(12)**

Contd P/2

3. (a) What are the special features of a Delta Modulation (DM) system in terms of (i) sampling rate, (ii) number of levels in quantizer, and (iii) predictor implementation? **(3)**

(b) Draw the block diagrams of the modulator and the demodulator of a Delta Modulation system. **(4+4)**

(c) The following set of sampled values of transmitted through the modulator of the DM system you have just drawn above. Compute the values received (at the demodulator) for each of the samples. Also compute the total RMS noise. Assume, step value, E = 0.5. Make any other assumptions required **(6+4)**

Sampled Values: 0.2, 0.3, 0.4, 0.4, 0.8, 1.2, 1.6

(d) What is the motivation behind making the step value adaptive in a DM system? Explain. How such adaptation can be made. **(10+4)**

4. (a) (i) Give an example where Time Division Multiplexing (TDM) is required. **(5+12)**

(ii) A DM1/2 multipmexer combines four DS1 signals, each of rate 1.544 Mbps and generates a time multiplexed signal of rate 6.312 Mbps. The frame format for such a DM1/2 multiplexer is given in Figure 4(a). Now explain why such framing is required. Particularly, describe the functions of the overhead bits: M. C and F's.

M_0	[48]	C_A	[48]	F_0	[48]	C_A	[48]	C_A	[48]	F_1	[48]
M_1	[48]	C_B	[48]	F_0	[48]	C_B	[48]	C_B	[48]	F_1	[48]
M_1	[48]	C_C	[48]	F_0	[48]	C_C	[48]	C_C	[48]	F_1	[48]
M_1	[48]	C_D	[48]	F_0	[48]	C_D	[48]	C_D	[48]	F_1	[48]

Figure: 4(a)

(b) When does 'Inter-Symbol Inference (ISI)' occur in a digital transmission system? Explain how ISI can be eliminated using pulse shaping. **(6+12)**

SECTION – B

There are **FOUR** questions in this section. Answer any **THREE**.

You must give your answer to the point avoiding unnecessary writings.

Try to draw as many figures as possible and give explanation with block diagrams.

Symbols in questions have the usual meanings.

5. (a) Why do you need to modulate baseband signal? **(3)**

(b) Show a comparison between DSB-SC, DSB+C (With Carrier), LSB-SC for the following **(2+3+3+3+3)**

(1) Modulation for a tone of $m(t) = \cos 100t$

(i) Draw basic spectrum of $m(t)$

(ii) Draw DSB-SC for $2m(t) \cos 100t$

(iii) Draw DSB+C for $2(1 + m(t))\cos 1000t$

(iv) Draw LSB-SC from figure of (ii) by suppressing USB

(2) Bandwidth used for these modulation techniques.

Contd P/3

CSE 311

Contd ... Q. No. 5

(c) Draw the signals obtained after encoding the bit stream '101011001' by each of the following digital line coding techniques: (i) Polar NRZ-1 (ii) Differential Manchester. **(6+12)**
Analyze these techniques in terms of the following characteristics.

(i) Bandwidth requirement
(ii) Synchronization capability
(iii) DC value suppression
(iv) Error detection capability

6. (a) Show the modulation procedure with block diagram, demodulation procedure with block diagram and bandwidth value only for the following.

(i) PSK (Phase shift keying) **(3+3+1)**
- Modulation
- Coherent Demodulation
- Bandwidth

(ii) FSL (Frequency shift keying) **(3+4+1)**
- Modulation
- Non-coherent Demodulation
- Bandwidth

(iii) DPSK (Differential Phase shift keying) **(3+4+1)**
- Modulation
- Non-coherent Demodulation
- Bandwidth

For modulation case you must give an example binary sequence and draw its modulated signal (approximately).
For Demodulation case explain with either diagram of as few lines as possible.
(b) Draw block diagram for a simple demodulation technique for FM (Frequency Modulation). **(4)**
(c) Three signals, each covering the range 10-15 KHz is to be transmitted simultaneously using Frequency Division Multiplexing (FDM). The resultant signal should cover the range 80-95 KHz. Describe how to generate the desired frequency multiplexed signal. **(8)**

7. (a) (i) Explain the relationship between FM (Frequency Modulation) and MP (Phase Modulation) in term of angle and instantaneous frequency of carrier signal. **(4+7+2+2)**
(ii) Show that the bandwidth of FM signal is infinite.
(iii) Under what condition FM is called Narrowband FM (NBFM).
(iv) Applying the above condition on the above expression of an FM signal, show that a narrowband FM signal can be approximated as:

$$f_{NBFM}(t) \approx A\left[\cos\omega_c t - k_f a(t)\sin\omega_c t\right] \qquad \text{where } a(t) = k_f \int_{-\infty}^{t} f(a)\,da$$

Contd P/4

CSE 311

Contd ... Q. No. 7

(b) Show with figure how phase continuation is maintained during MSK, what are the prerequisites for carrier frequency to work with MSK. **(8+4)**

(c) On the process of recovering incoming carrier explain the following method briefly. **(3+5)**

(i) Pilot carrier

(ii) Phase locked loop (PLL) (Draw Block diagram)

8. (a) Show that, for a time function $f(t)$, the signal, $f_{SSB}(t) = f(t)\cos\omega_c t + f_h(t)\sin\omega_c t$ represents a lower sideband SSB-SC signal where $f_h(t)$ represents Hilbert Transform of $f(t)$ and ω_c is the carrier frequency in rad/sec. **(14)**

(b) Describe how Quadrature Amplitude Modulation (QAM) is performed. Explain why QAM is bandwidth efficient than DSB-SC modulation. **(4+5)**

(c) For Vestigial Sideband (VSB), derive that output filter $H_0(\omega) = \dfrac{1}{H_i(\omega + \omega_c) + H_i(\omega - \omega_c)}$ is used to reform the actual transmission signal that is distorted by transmitter input filter $H_i(\omega)$. [Show only the necessary equation] **(8)**

(d) Describe how 'Scrambling' solves the synchronization problem with Bipolar-AMI. **(4)**

L-3/T-1/CSE **Date : 31/05/2014**

BANGLADESH UNIVERSITY OF ENGINEERING AND TECHNOLOGY, DHAKA

L-3/T-1 B. Sc. Engineering Examinations 2012-2013

Sub : **CSE 311** (Data Communication I)

Full Marks : 210 Time : 3 Hours

The figures in the margin indicate full marks.

USE SEPARATE SCRIPTS FOR EACH SECTION

SECTION – A

There are **FOUR** questions in this section. Answer any **THREE**.

1. (a) (i) In Bipolar NRZ-L line coding , 1 is represented by 5V and 0 is represented by –5V during bit duration. Now, explain why Bipolar NRZ-L suffers from synchronization problems for both long run of 1's and 0's. **(4+4+4+4+4)**

(ii) With proper explanation, give an example of a line coding technique that uses the same bandwidth as Bipolar NRZ-L but solves the synchronization problem for long run of 1's (though synchronization problem for long 0's still exists).

(iii) With proper explanation, give an example of a line coding technique that solves the synchronization problems for both long run of 1's and 0's but requires twice the bandwidth of Bipolar NRZ-L.

(iv) Describe how 'Block Coding' attempts to solve the synchronization problem with "NRZ-I". Does "Block Coding" affect the bandwidth of the resulting signal? Explain with an example.

(b) Five signals, each covering the range 20-80 KHz are to be transmitted simultaneously using Frequency Division Multiplexing (FDM). The resultant signal should cover the range 300-700 KHz where frequency gap (guard band) of 25 KHz is provided between each pair of consecutive channels for interference reduction. See Figure 1(b) for an illustration. Now, describe how to generate the desired frequency multiplexed signal. **(15)**

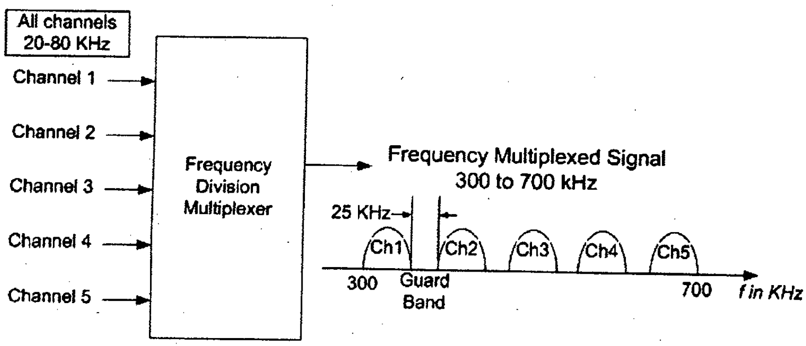


Figure 1(b)

Contd P/2

CSE 311

2. (a) For each the following three modulation techniques for binary digital data, determine the bandwidth of the modulated signal when the baseband bandwidth of the modulating signal is B Hz. **(12)**

   (i) Amplitude Shift Keying (ASK), (ii) Frequency Shift Keying (FSK), and (iii) Phase Shift Keying (PSK).

   (b) Draw the block diagrams for – (i) Noncoherent (asynchronous) detection of FSK, and (ii) coherent (synchronous) detection of FSK. **(4+4)**

   (c) (i) Explain with an example, how M-ary signaling achieves higher data rate than binary signaling using the same bandwidth but at the cost of more power. **(7+8)**

   (ii) Show that for M-ary FSK, the minimum frequency separation to ensure orthogonality among the FSK symbols is: $\delta f = 1/2T_b$ where T_b denotes the symbol duration.

3. (a) Explain how the circuit in Figure 3(a) functions as an envelope detector for AM (Amplitude Modulation) signal. Describe how to choose the constants 'R' and 'C' in the circuit for envelope detection of a given AM signal. **(8+4)**

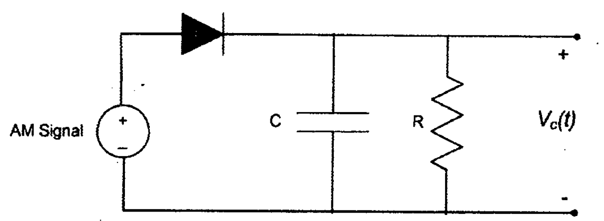


Figure 3(a)

   (b) Show that, for a real time function $f(t)$, the signal, $f_{SSB}(t) = f(t)\cos\omega_c t + f_h(t)\sin\omega_c t$ represents a lower sideband SSB-SC signal where $f_h(t)$ represents the Hilbert Transform of $f(t)$ and ω_c is the carrier frequency in rad/sec. **(14)**

   (c) Describe how Quadrature Amplitude Modulation (QAM) is performed. Explain why QAM is bandwidth efficient than DSB-SC modulation. **(4+5)**

4. (a) A baseband signal $f(t)$ of bandwidth B is modulating the frequency of a carrier at frequency ω_c (in rad/sec). Show that, the resulting frequency modulated signal can be expressed as – **(5+2+8+5+3)**

$$f_{FM}(t) = A\cos\left[\omega_c t + k_f \int_{-\infty}^{t} f(a)\,da\right]$$ where, k_f is a constant.

Contd P/3

CSE 311

Contd ... Q. No. 4(a)

Under what condition, an FM signal is called Narrowband FM (NBFM). Applying the condition on the above expression of an FM signal, show that a narrowband FM signal can be approximated as:

$$f_{NBFM}(t) \approx A\left[\cos \omega_c t - k_f\, a(t) \sin\omega_c t\right] \quad \text{where } a(t) = k_f \int_{-\infty}^{t} f(a)\, da$$

Draw the block diagram of a narrowband FM generator according to this expression. What is the problem of generating NBFM signals in this way?

(b) 'Angle modulation is preferable over Amplitude modulation for systems where nonlinear amplifiers are used'. Justify the statement by showing how Angle Modulation is immune to nonlinearities whereas Amplitude Modulation suffers from distortion during nonlinear amplification. **(12)**

SECTION – B

There are **FOUR** questions in this section. Answer any **THREE**.

5. (a) Define unit impulse function. Derive the sampling property of unit impulse function. **(8)**

(b) The convolution of two functions *g(t)* and *w(t)*, denoted by *g(t)*w(t)*, is defined by the integral: $g(t)*w(t) = \int_{-\infty}^{\infty} g(\tau) w(t-\tau)\, d\tau$. Now, prove that the convolution of two signals in the time domain becomes multiplication in the frequency domain and vice versa. **(14)**

(c) The Fourier transform of a function *g(t)* is expressed as: $G(f) = \int_{-\infty}^{\infty} g(t) e^{-j2\pi ft} dt$.

Find the Fourier transform of $g(t) = \Pi(t/\tau)$ in Figure 5(c). **(13)**

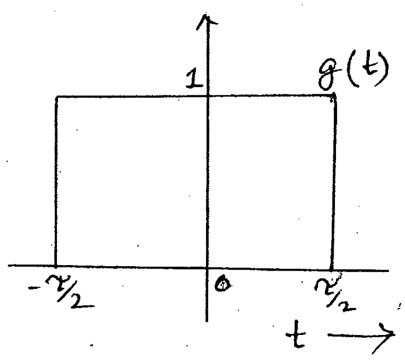


Figure: 5(c)

CSE 311

6. (a) Describe how DPCM system works with suitable block diagrams of both transmitter and receiver. **(12)**

   (b) State and proof the Time-Shafting property and the Frequency Shifting property of Fourier Integral. **(10)**

   (c) How does 'Inter-Symbol Inference (ISI)' occur in digital transmission system? Explain how ISI can be eliminated using "pulse shaping". **(13)**

7. (a) Show that for a distortionless system, the amplitude response $|H(f)|$ must be a constant, and the phase response $\theta_h(f)$ must be a linear function of f going through the origin. **(8)**

   (b) Describe how TDM (Time Division Multiplexing) Digital Hierarchy works. Why bit stuffing and framing are used in TDM? **(14)**

   (c) A low-pass filter (Figure 7(c.1) transfer function $H(f)$ is given by **(13)**

$$H(f) = \begin{cases} (1 + k\cos 2\pi fT)e^{-j2\pi f t_d}, & |f| < B \\ 0, & |f| > B \end{cases}$$

A pulse $g(t)$, band-limited to B Hz (Figure 7(c.2), is applied at the input to this filter. Show that the output is $y(t) = g(t - t_d) + \frac{k}{2}[g(t - t_d - T) + g(t - t_d + T)]$

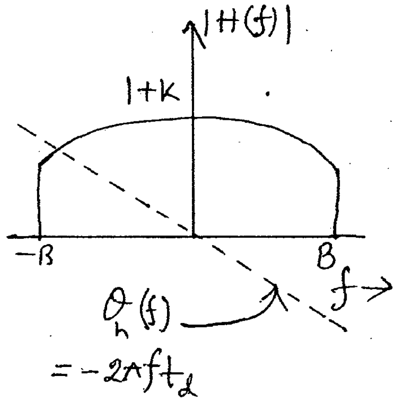


figure: 7 (c.1)

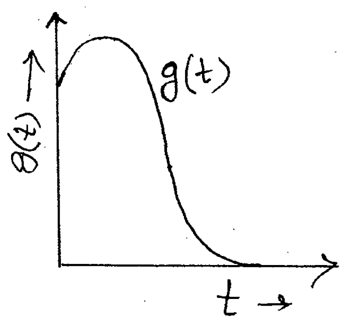


figure 7(c.2)

8. (a) Prove that the minimum sampling frequency for perfect signal recovery is $f_s = 2B$ Hz. Assume conventional meaning of the symbol. **(15)**

   (b) How quantization is done in a PCM system? Why non-uniform quantization is suitable for voice signal? How does companding achieve non-uniform quantization? **(6+6+8=20)**

Saifur 6/10/13

L-3/T-1/CSE **Date : 06/10/2013**

BANGLADESH UNIVERSITY OF ENGINEERING AND TECHNOLOGY, DHAKA

L-3/T-1 B. Sc. Engineering Examinations 2011-2012

Sub : **CSE 311** (Data Communication I)

Full Marks : 210 Time : 3 Hours

The figures in the margin indicate full marks.

USE SEPARATE SCRIPTS FOR EACH SECTION

SECTION – A

There are **FOUR** questions in this Section. Answer any **THREE**.

1. (a) Draw the signals obtained after encoding the bit stream- '010100110' by each of the following two digital line coding techniques: (i) Polar NRZ-I and (ii) Bipolar-AMI. Analyze these two techniques in terms of the following characteristics. **(5+10+5)**

   (i) Bandwidth requirement

   (ii) Synchronization capability

   (iii) DC value suppression

   (iii) Error detection capability

   Describe how 'Scrambling' solves the synchronization problem with Bipolar-AMI.

   (b)(i) Ten voice sources, each covering the frequency range from 20 Hz – 20 KHz are each to be transmitted using PCM technique as shown in Figure 1(b). Each signal is sampled at a rate of 44,000 samples/sec. Each sample is quantized and coded into 12 bits. The 10 PCM streams are then time-multiplexed into one composite PCM signal. (Neglect control and framing bits in the TDM frame that might be required). The composite PCM stream is fed to a 16-QAM modulator with carrier frequency, f_c = 100 MHz. Sinusoidal roll-off shaping with r = 0.1 is used. Find the transmission bandwidth at the output of the modulator. **(10)**

   (ii) Consider again the problem in part (i) above. This time, a frame at the output of the time-division multiplexer is defined to be 1.25 ms long. 10 control bits are added per frame. What is the required transmission bandwidth at the output of the modulator? **(5)**

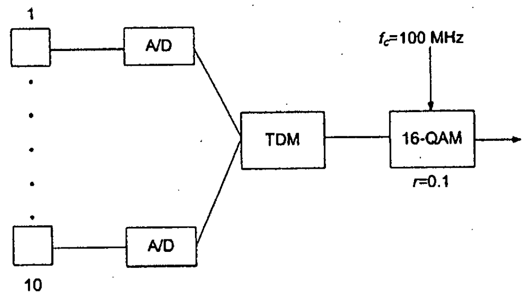


Figure 1(b)

Contd P/2

CSE 311

2. (a) Consider the second level of a two-level FDM hierarchy as shown in Figure 2(a). At this level, four groups (from level 1, not shown in the figure) each in the range 80 to 96 kHz are combined to generate a supergroup in the range 312 to 376 kHz using SSB (upper sideband) modulation. Explain the supergroup generation process using suitable diagram. Specify the frequency values that must be used at multiplier for each group/channel. **(10)**

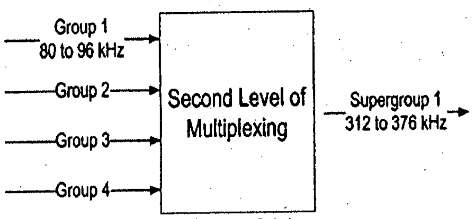


Figure 2(a)

(b) Explain how 'Frequency Modulation (FM)' and 'Phase Modulation (PM)' are related. Define narrowband FM and wideband FM. An FM signal can be demodulated by passing it through an ideal differentiator followed by an envelope detector. Explain why this method works. **(8+5+12)**

3. (a) (i) One of the methods of coherent/synchronous detection of OOK signals is to multiply the incoming signal by the carrier frequency, as locally generated at the receiver, and then low pass-filtering the resultant multiplied signal. What happens when the local carrier is not exactly at the same frequency and same phase as the incoming carrier? Explain.
(ii) There is another method for synchronously demodulating OOK signals that does not require exact frequency and phase synchronization. In this method, the incoming signal is sampled at the carrier frequency rate and then the sampled signal is passed through a low-pass-filter. Explain why this method works. **(12+8)**
(b) Draw the diagram of an envelope detector circuit. Why envelope detection cannot be used with PSK signals? Describe an asynchronous/non-coherent demodulation technique for PSK signals. **(3+3+9)**

4. (a) Consider a 'Quaternary PSK (QPSK)' signal where the four phase angles are: $\pm\pi/4$, $\pm3\pi/4$. What is the maximum phase-shift at symbol transition point of the given QPSK signal? Describe how each of the following two techniques reduces the value of maximum phase-shift: (i) Offset QPSK and (ii) $\pi/4$- shifted QPSK. **(3+8+8)**
(b) Describe how the following modulation techniques differ: (i) Normal AM, (ii) DSB-SC (Double-Sideband Suppressed-Carrier), (iii) SSB-SC (Single-Sideband Suppressed-Carrier) and (iv) VSB-SC (Vestigial-Sideband Suppressed-Carrier). Describe a scenario where Normal AM is required. **(8+8)**

Contd P/3

CSE 311

SECTION – B

There are **FOUR** questions in this Section. Answer any **THREE**.

5. (a) The Fourier integral F(ω) of a non-periodic function *f*(t) is defined as: **(8)**

$$F(\omega)=\int_{-\infty}^{\infty} f(t)e^{-j\omega t}\,dt$$

Accordingly, find the Fourier integral of the function *f*(*t*) shown in Figure 5(a).

(b) Find the Fourier integral of function *g*(*t*) shown in the Figure 5(b) from the Fourier integral of function *f*(*t*) shown in Figure 5(a), and then the Fourier integral of function *h*(*t*) shown in Figure 5(c) from the Fourier integral of function *g*(*t*) using suitable properties of Fourier integral. **(8+6)**

(c) How is the bandwidth of a real time signal calculated? Accordingly which of the three functions *f*(*t*), *g*(*t*), and ~~*i*(*t*)~~ h(t) shown in Figures 5(a), 5(b), and 5(c) respectively, has the lowest bandwidth? Explain your answer. **(5+8)**

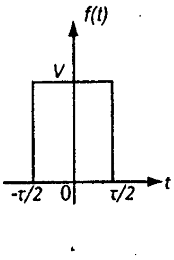


Figure 5(a)

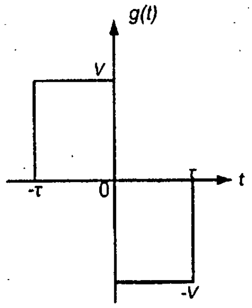


Figure 5(b)

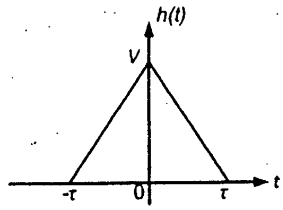


Figure 5(c)

6. (a) Explain the 'Linearity' and 'Time invariance' properties of a 'Linear Time Invariant (LTI)' system. Let *g*(*t*) be output response of an LTI system with transfer function $H(\omega)$ to an input signal f(t). Let $F(\omega)$ and $G(\omega)$ be the Fourier transform of f(*t*) and g(*t*) respectively. Show that, $G(\omega) = H(\omega)F(\omega)$. **(9+9)**

(b) An LTI system is called 'distortionless' when for any input function *f*(*t*), the output of the system *g*(*t*) is expressed as: $g(t) = Af(t - t_d)$, where A and t_d are constants. According to this definition, find out the transfer function $H(\omega)$ of a distrotionless LTI system. Under what conditions, the RC circuit shown in Figure 6(b) can be approximated as a distortionless system. **(9+8)**

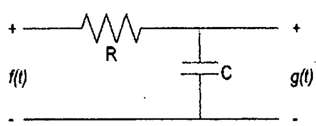


Figure 6(b)

7. (a) In a PCM system, the input signal is limited to ± V volts. After sampling, the sampled signals are quantized into M uniformly spaced levels. For this system, show that, the mean power output SNR, $S_0/N_0 = M^2-1$. **(12)**

(b) In a PCM system, explain how quantization with non-uniform spacing of levels can be achieved without actually changing the uniform level spacing in the quantizer. **(8)**

(c) Describe how 'Delta Modulation (DM)' system works with suitable block diagrams of both transmitter and receiver. **(15)**

8. (a) How does 'Inter-Symbol Interference (ISI)' occur in a digital transmission system? Explain how ISI can be eliminated using pulse shaping. **(5+8)**

(b) Describe briefly how the following three issues are resolved in a Time Division Multiplexing (TDM) system. (i) Synchronization, (ii) Difference in the bit rates of the incoming channels, and ~~(ii)~~ (iii) Small variation in bit rate of an incoming channel. **(9)**

(c) Consider a DS3 multiplexer as shown in Figure 8(c) that combines seven DS2 digital streams into one DS3 stream. Each incoming DS2 stream has a nominal bit rate of 6.312 Mbps. The bit rate of the output DS3 signal is 44.736 Mbps. Each DS3 frame contains 4704 data bits and 56 control bits. What is the maximum fractional increase over the nominal rate for each input channel that can be accommodated by the multiplexer? DS3 multiplexer allows one bit per frame to be stuffed per input signal channel. Is this sufficient to accommodate expected variation in the input rate? Explain your answer. **(8+5)**

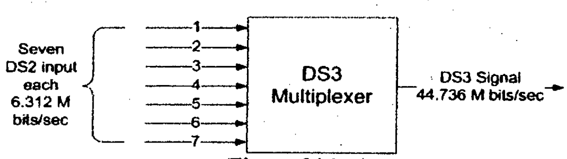


Figure 8(c)

L-3/T-1/CSE **Date : 23/02/2012**

BANGLADESH UNIVERSITY OF ENGINEERING AND TECHNOLOGY, DHAKA

L-3/T-1 B. Sc. Engineering Examinations 2010-2011

Sub : **CSE 311** (Data Communication I)

Full Marks: 210 Time : 3 Hours

The figures in the margin indicate full marks.

USE SEPARATE SCRIPTS FOR EACH SECTION

SECTION – A

There are **FOUR** questions in this Section. Answer any **THREE**.

1. (a) Consider the four signals in Figure 1(a), 1(b), 1(c) and 1(d) that are generated by encoding the bit-stream '0011100011' using four different digital line coding techniques. For each of these four line coding techniques identify and write the bit encoding rules (i.e., how 0 and 1 are represented). Compare the four techniques with respect to the following characteristics. **(4+16=20)**

(i) Bandwidth requirement (ii) Synchronization capability

(iii) DC value suppression (iv) Error detection capability

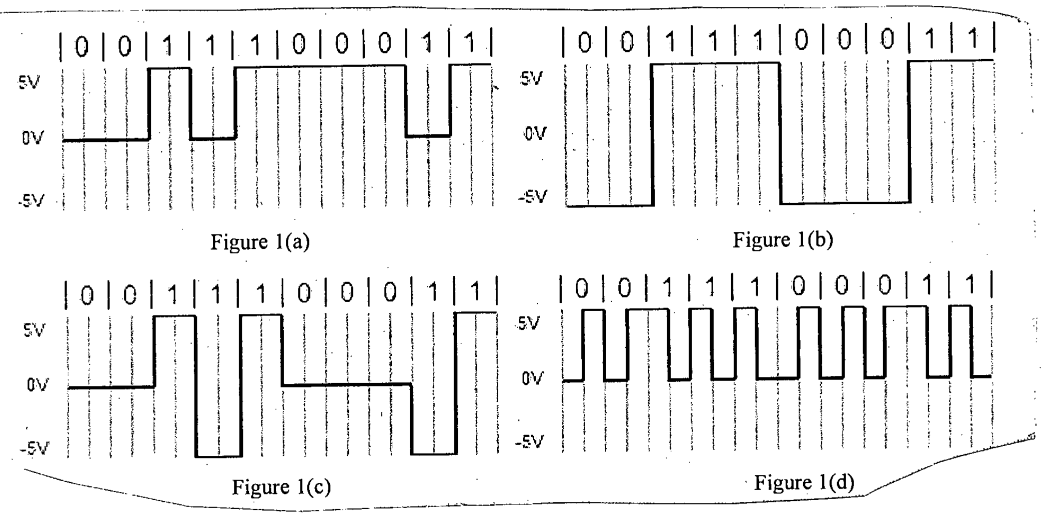


Figure 1(a) Figure 1(b)

Figure 1(c) Figure 1(d)

(b) Three signals, each covering the range 10-15 KHz are to be transmitted simultaneously using Frequency Division Multiplexing (FDM). The resultant signal should cover the range 80-95 KHz. Describe how to generate the desired frequency multiplexed signal. **(8)**

(c) FSK transmission is used to transmit 1200-bit/s digital signals over a telephone channel. The FSK signals are to fit into the range 500 to 2900 Hz. The carrier frequencies are 1200 and 2200 Hz respectively. Find the baseband bandwidth required for the binary signals. Assuming sinusoidal roll-off shaping, what roll-off factor is required? **(7)**

2. (a) Explain with an example the difference between bit rate and pulse rate and thus show the advantage of using multisymbol signaling. Describe how multi-symbol signal can be generated using QPSK. Also draw the block diagram of a QPSK signal generator. **(8+8+4=20)**

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CSE 311

Contd ... Q. No. 2

(b) How do Frequency Modulation (FM) and phase modulation (PM) differ? Show that the bandwidth of narrowband FM signal is 2B Hz when baseband bandwidth of the modulating signal is B Hz. **(5+10=15)**

3. (a) Show that for a real time signal $f(t)$, the signal $f_{SSB}(t) = f(t)\cos\omega_c t + f_h(t)\sin\omega_c t$ represents a lower sideband SSB-SC signal where $f_h(t)$ represents the Hilbert Transform of $f(t)$. **(15)**

(b) There are two methods for generating SSB-SC signal: (i) Passing a DSB-SC signal through a band-pass filter; (ii) Using a circuit according to the expression for $f_{SSB}(t)$ as mentioned in Question 3(a). Now identify the practical difficulties involved in both the methods. Give example of a signal which is suitable for SSB-SC modulation using the first method. **(8+5=13)**

(c) Why Vestigial-sideband (VSB) signal is called so? Describe how VSB signal is generated. **(3+4=7)**

4. (a) What is the maximum number of pulses that can be transmitted over a bandwidth of B Hz? Explain your answer. **(15)**

(b) The amplitude spectrum for a sinusoidal roll-off filter is given as follows: **(6+6+8=20)**

$$|H(\Delta\omega)| = \tfrac{1}{2}(1-\sin(\pi/2)(\Delta\omega/\omega_x)) \qquad |\Delta\omega| < \omega_x$$
$$= 0 \qquad |\Delta\omega| > \omega_x$$
$$= 1 \qquad -\omega_c < \Delta\omega < -\omega_x$$

Sketch the amplitude spectrum. Does this filter satisfy the Nyquist Criterion of odd symmetry for generating wave-shape with zero intersymbol interference (ISI)? Explain your answer. Show that for the given sinusoidal roll-off filter $B = (1 + r)/2T$, where the symbols carry their usual meanings in the given context.

SECTION – B

There are **FOUR** questions in this Section. Answer any **THREE**.

5. (a) How is the bandwidth of a real time signal calculated? Let $f'(t)$ be the derivative of a real time signal $f(t)$. Which of $f(t)$ and $f'(t)$ has higher bandwidth? Explain your answer. **(6+8=14)**

(b) The Fourier Transform of the triangular pulse f(t) in Figure 5(a) is given as- **(15+6=21)**

$$F(\omega) = 1/\omega^2\,(e^{j\omega} - j\omega e^{j\omega} - 1)$$

Using this information, and the time-shifting and time-scaling properties, find the Fourier Transforms of the signal $g(t)$ shown in Figure 5(b). Does the inverse-time bandwidth relation hold for this signal?

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CSE 311

Contd ... Q. No. 5(b)

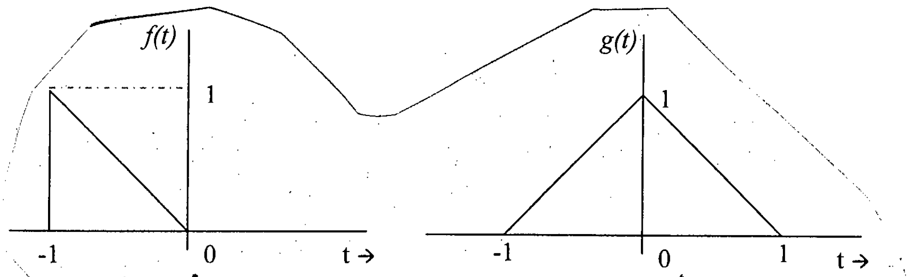

Figure 5(a) Figure 5(b)

6. (a) Find out and sketch the impulse response of an ideal low-pass filter with linear phase characteristic. Using the result explain why such an ideal filter in physically unrealizable. **(10+5=15)**

   (b) Let the signal $g(t)$ as shown in Figure 5(b) be input to a linear time invariant system with transfer function, $H(\omega) = j\omega$. Find out and sketch the output (in time domain) of the system. **(8)**

   (c) Let $f(t)$ be a real time signal band-limited to B Hz. Let $f_s(t)$ be the signal obtained by periodically sampling $f(t)$ at a rate of 2B samples per second. Show that if $f_s(t)$ is passed through an ideal low pass filter of bandwidth B Hz, then the output will be proportional to $f(t)$. **(12)**

7. (a) How does a Quantizer with uniform spacing of levels work in a PCM system? When does quantization with non-uniform spacing result in better signal -to-quantization noise ratio than quantization with uniform spacing? Explain with an example. **(5+10=15)**

   (b) Explain how step size k' affects 'quantization noise' and 'overload noise' in a Delta Modulation (DM) system. How does adaptive delta modulation adjust k' to reduce both 'quantization noise' and 'overload noise'? **(14+6=20)**

8. (a) What is Time Division Multiplexing (TDM)? What is the advantage of using TDM hierarchy? Explain why the output data rate of a multiplexer at any level in a TDM hierarchy is usually kept higher than the total of data rates of all its incoming channels. **(6+6+6=18)**

   (b) What is the advantage of using Differential Pulse Code Modulation (DPCM) system over Pulse Code Modulation (PCM) System? Discuss briefly on designing the predictor of a DPCM system. **(5+12=17)**
