Benha University
Faculty of Engineering at Shoubra
Electrical Engineering Department
2nd Year (Communications & Electronics)

Final Term Exam
Date: Thursday 8/6/2017
Subject: Signals
Duration: 3 hours

- Answer all the following question
- Illustrate your answers with sketches when necessary.
- The exam consists of **Three** page

- No. of questions: 3
- Total Mark: 90 Marks
- Examiner: Dr. Michael Nasief

Question (1): Signal Manipulation: [30 Marks]

A. For the signal shown in figure (1) draw: **[10 Marks]**
   - The signal delayed 2 sec, atenuated by 2 and compressed by 2.
   - The signal DC shifted by (-1), advanced by 1 and expanded by 2.

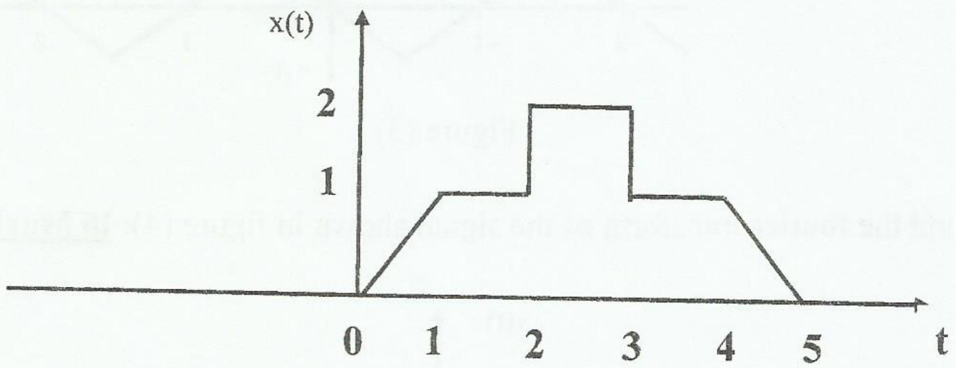

Figure (1)

B. Sketch and label the even and odd components of the signal shown in figure (1). **[8 Marks]**

C. For the signal x(t) shown in figure (1) find the size of: **[6 Marks]**
   - $x(t)\,[u(t) - u(t-1)]$.
   - $x(t)\,[u(t-1) - u(t-2)]$.

D. Show that the following system is Time-Invariant/Time-varying: **[3 Marks]**
   - $y(t) = \sin(t)\, x(t-2)$.

E. Show that the following system is Invertible or not: **[3 Marks]**
   - $y(t) = 3\,t\,x(t-2) - 2$.

Question (2): Convolution and Fourier: [30 Marks]

A. Convolve the signals shown in figure(2): **[8 Marks]**

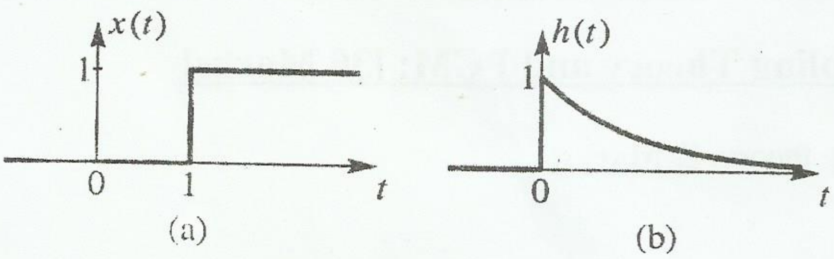

Figure (2)

Where:
$x(t) = u(t-1)$.
$h(t) = e^{-t} u(t)$.

B. Use direct integration to find expression for: **[4 Marks]**

$$y(t) = e^{-at} u(t) * e^{-bt} u(t).$$

C. Find the trigonometric Fourier series and compact trigonometric Fourier series for the triangular periodic signal shown in Fig. 3 over the interval $-1 \le t \le 1$. **[8 Marks]**

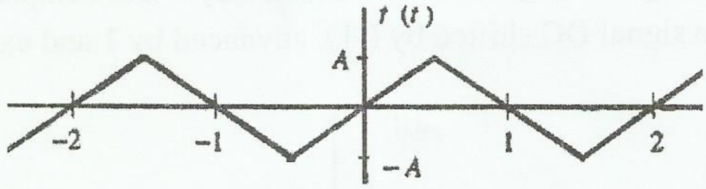

Figure (3)

D. Find the fourier transform of the signal shown in figure (4): **[6 Marks]**

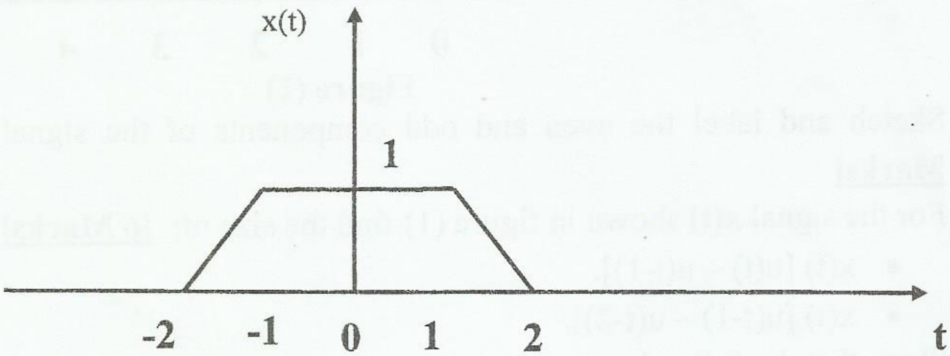

Figure (4)

E. Use Harmonics Tabular Method to find the Fourier series coefficients for the signal samples shown in the next table **(get only the Dc component and the first harmonic coefficients): [4 Marks]**

V(t)	10	20	30	35	40	40	35	30	20	10	5	0
wt	30	60	90	120	150	180	210	240	270	300	330	360

Question (3): Sampling Theory and PCM: [30 Marks]

A. State the sampling theory. **[2 Marks]**

B. Proof (mathematicaly) that the output spectrum of the sampled signal will be a duplicated version of the original spectrum. **[8 Marks]**

C. Define the Aliasing problem and show how you can avoid it. **[4 Marks]**

D. If x(t) is the input signal to PCM encoder (Sound card in your PC): **[8 Marks]**

x(t) = speech signal with peak to peak (5 v)

Find:

- Sampling frequency at Nyquist rate.
- If the step voltage between 2 consecutive quantum levels (Δv) = 10 mv:
    - What is the number of levels?
    - What is the number of bits per sample?
    - What is the bit rate?

E. Explain the PCM Encoder and Decoder **[6 Marks]**.

F. In your opinion what are the advantages and disadvantages of Delta Modulation? **[2 Marks]**

GOOD LUCK

DR. MICHAEL NASIEF

Q(1) Signal Manipulation [30 marks]

(A)

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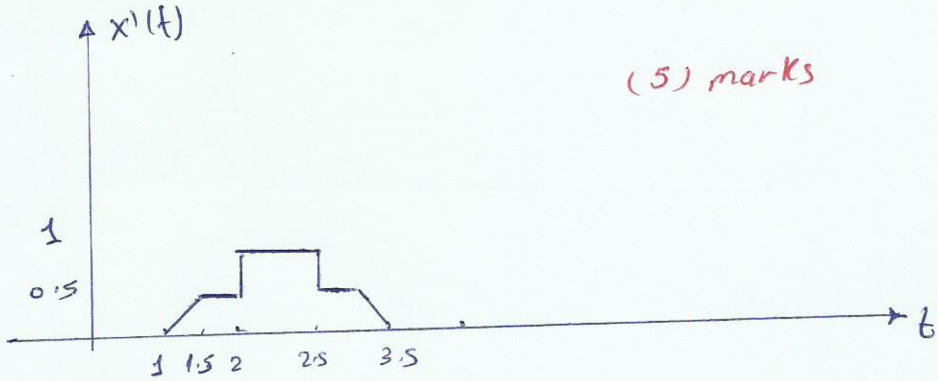
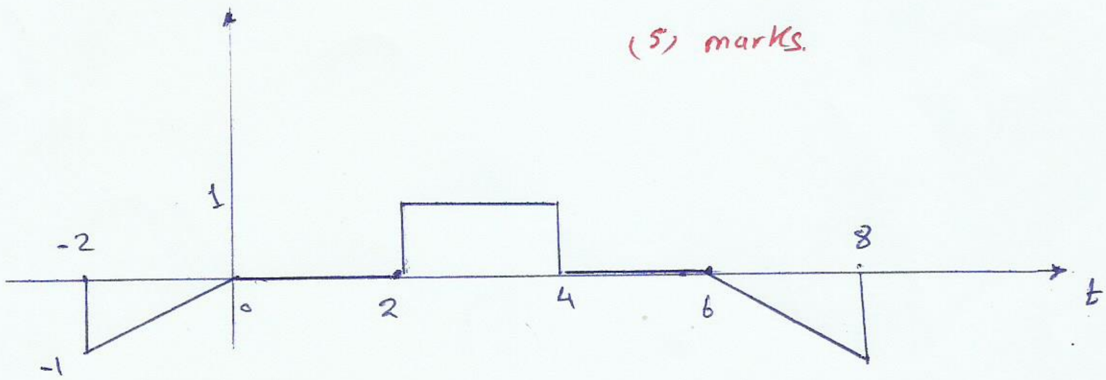


(5) marks

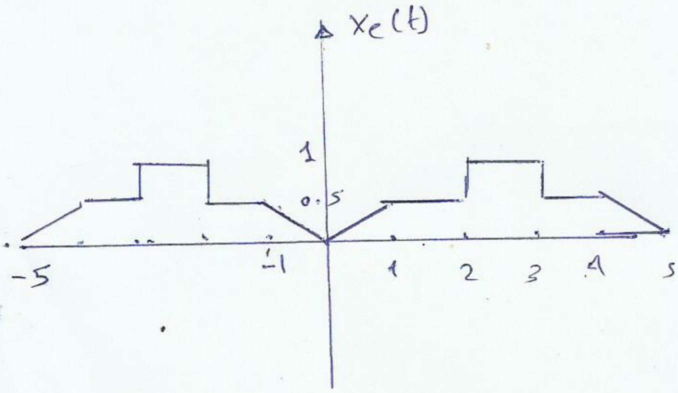
*



(5) marks.

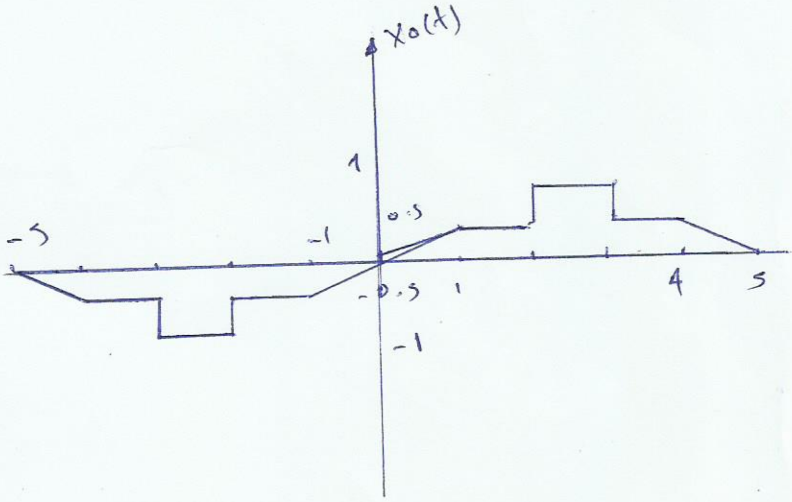
(B) $X_e = \frac{1}{2} x(t) + \frac{1}{2} x(-t)$

(4) marks



$X_o = \frac{1}{2} x(t) - \frac{1}{2} x(-t)$

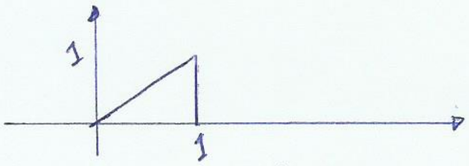
(4) marks



(c)

$x(t)\,[u(t)-u(t-1)]$ (3) marks

Energy signal

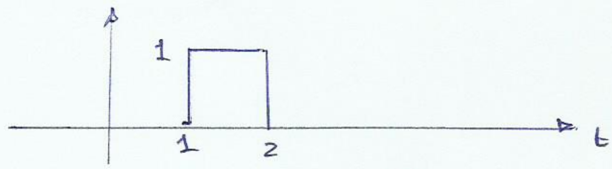


$$\int_{-\infty}^{\infty} x^2(t)\,dt = \int_0^1 t^2\,dt = \frac{t^3}{3}\Big]_0^1$$

$$= \frac{1}{3}$$

$x(t)\,[u(t-1)-u(t-2)]$

Energy signal (3) marks.



$$\int_1^2 1^2\,dt = 2-1 = 1$$

(D) Time varying system. (3) marks

$y_1(t) = x(t-t_0-2)\sin t$

$y_2(t) = x(t-t_0-2)\sin(t-t_0)$

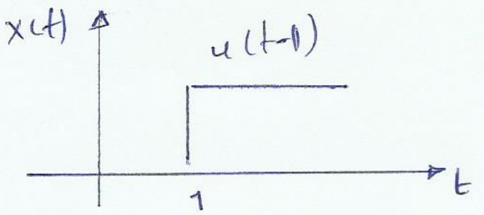
$y_1 \neq y_2 \therefore$ time varying system.

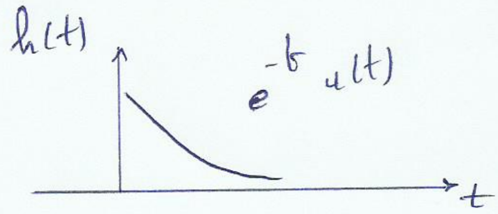
(3) marks.

(E) The sys is invertible Except for $t=0$ " we can say generally non-inv "

Q(2) Convolution and Fourier. [30 marks]

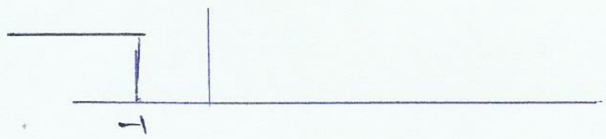
{A}





$x(-\tau)$

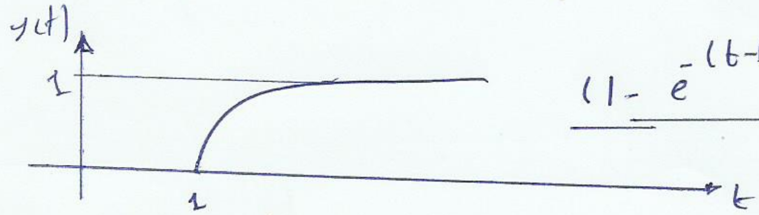
(8) marks



at $t-1<0$ no overlap $\therefore = 0$

at $t-1>0$

$$\int_0^{t-1} e^{-\tau}\, d\tau = -e^{-\tau}\Big]_0^{t-1} = 1 - e^{-(t-1)} \qquad t \geq 1$$



$(1 - e^{-(t-1)})\, u(t-1)$

{B}

$y(t) = e^{-at}u(t) * e^{-bt}u(t)$ (4) marks

$$= \int_{-\infty}^{\infty} e^{-a\tau} u(\tau)\, e^{-b(t-\tau)} u(t-\tau)\, d\tau$$

$$= \frac{e^{-at} - e^{-bt}}{b-a}\, u(t) \qquad t>0$$

$\{ \; 0 \quad t<0$

{C} period $T_0 = 2$ (8) marks

$$\omega_0 = \frac{2\pi}{T_0} = \pi$$

$$F(t) = a_0 + \sum_{n=1}^{\infty} a_n \cos n\pi t + b_n \sin n\pi t$$

$$F(t) = \begin{cases} 2At & |t| \leq \frac{1}{2} \\ 2A(1-t) & \frac{1}{2} \leq t \leq \frac{3}{2} \end{cases}$$

choose interval $-\frac{1}{2}$ to $\frac{3}{2}$

$$a_n = \frac{2}{2} \int_{-\frac{1}{2}}^{3/2} f(t) \cos n\pi t \, dt$$

$$= 0$$

$$b_n = \int_{-1/2}^{1/2} 2At \sin n\pi t \, dt + \int_{1/2}^{3/2} 2A(1-t) \sin n\pi t \, dt$$

$$b_n = \frac{8A}{n^2\pi^2} \sin\left(\frac{n\pi}{2}\right)$$

#

$$c_n = \sqrt{a_n^2 + b_n^2} = b_n \quad \text{since } a_n = 0$$

D

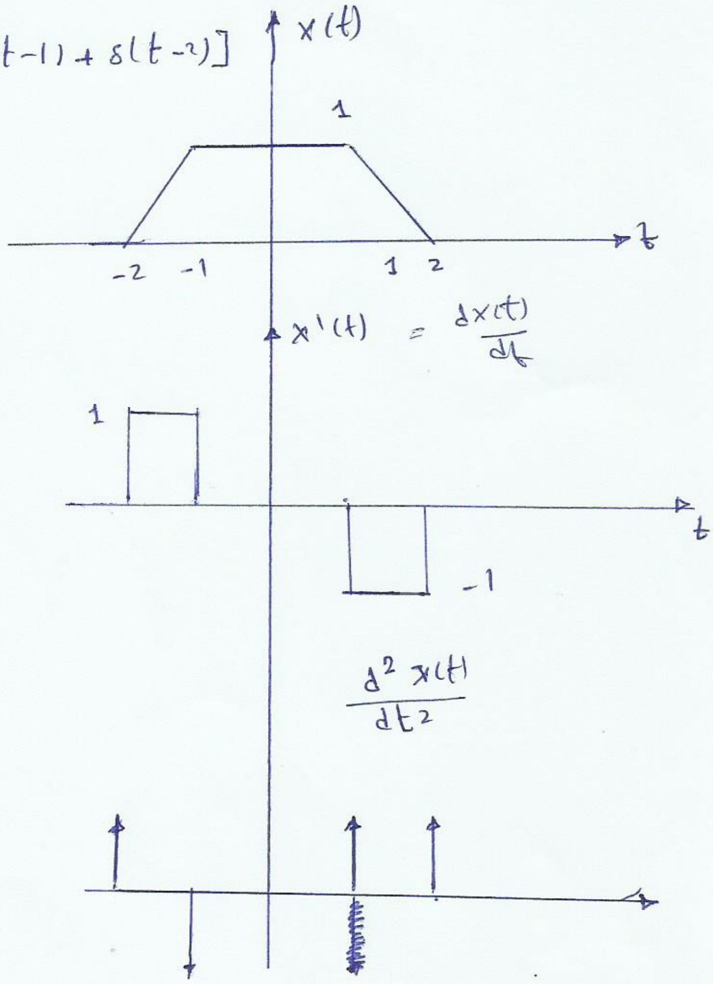
$$F\left(\frac{d^2 x(t)}{dt^2}\right) = \frac{1}{2-1}\left[\delta(t+2) - \delta(t+1) - \delta(t-1) + \delta(t-2)\right]$$

$$-\omega^2 F(\omega) = \frac{1}{2-1}\left[e^{j\omega 2} - e^{j\omega} - e^{j\omega} + e^{-j\omega 2}\right]$$

$$= [\cos 2\omega - \cos\omega]$$

$$-\omega^2 F(\omega) = \cos 2\omega - \cos\omega$$

$$\therefore F(\omega) = -\frac{(\cos 2\omega - \cos\omega)}{\omega^2}$$



(6) marks

E

(4) marks

$$a_0 = \frac{275}{12} = 22.91$$

$$a_1 = \frac{-109.5}{6} = -18.25$$

$$b_1 = \frac{28.173}{6} = 4.695$$

Q(3) Sampling Theory

30 marks

(A) A cont. time signal $x(t)$ with Freq. no higher than f_{max} Hz can be reconstructed Exactly from its samples $x[n] = x(nT_s)$ if its samples taken at a rate $F_s = \frac{1}{T_s} \geq 2F_{max}$

(2) marks

(B) The sampled version can be expressed as:

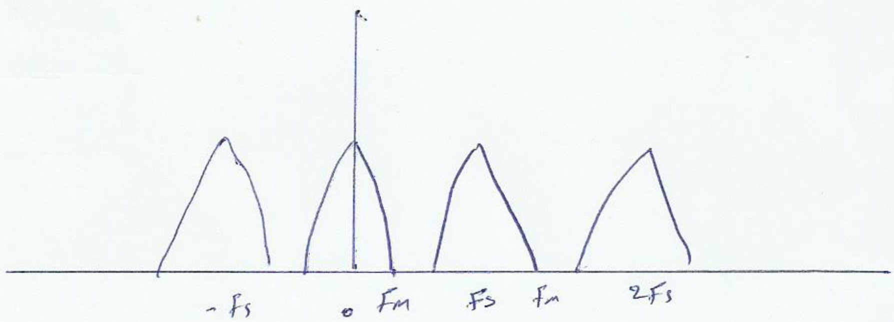
(8) marks

$$\bar{x}(t) = x(t)\,\delta_{T_s}(t) = \sum_n x(nT_s)\,\delta(t - nT_s)$$

$$\delta_n(t) = \frac{1}{T_s}\left[1 + 2\cos\omega_s t + 2\cos 2\omega_s t \cdots\right] \quad \& \quad \omega_s = \frac{2\pi}{T_s}$$

Since $2x\cos\omega_s t \Leftrightarrow X(\omega - \omega_s) + X(\omega + \omega_s)$

$$\therefore \bar{X}(\omega) = \frac{1}{T_s} \sum_{-\infty}^{\infty} X(\omega - \omega_s)$$

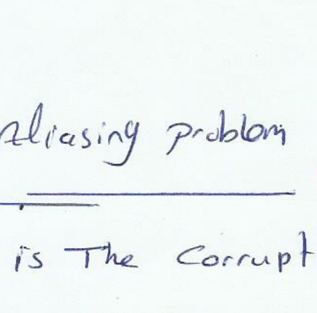


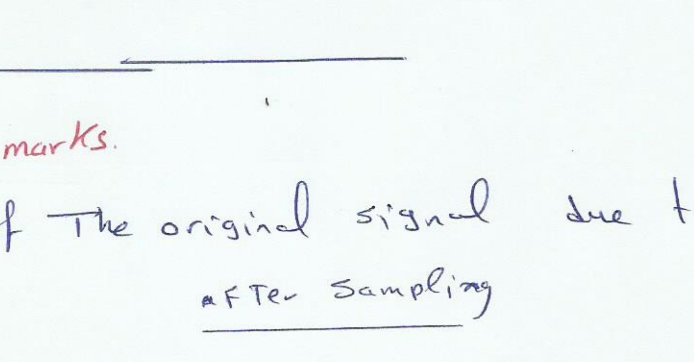
(C) Aliasing problem

(4) marks

is The corruption of The original signal due to after Sampling

B < Fs/2





to avoid aliasing problem apply Low pass Filter before Sampling

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D) $x(t)$ speech $\therefore$ 4 KHz (8) marks

* $F_s = 2F_m = 8K$ Samples /sec
* $L = \frac{5V}{10mV} = 500 \rightsquigarrow 512$ levels
* $n = \log_2 L = 9$
* $F_b = n F_s = 9 * 8k = 72$ kbps

E) (6) marks

1) Sampling : $F_s \geq 2F_m$
2) Quantizing : to The nearst Quantum level
3) Encoding : Convert to 0,1 $L = 2^n$

F) (2) marks

Adv

Low bit rate

dis adv

Low quality
