1. With $n_{eff} = 1.5$ in silicon waveguide, calculate the waveguide length difference ΔL.
2. Explain how you can build a 4×4 multiplexer from a 2×2 multiplexer. Determine the waveguide length difference of each stage.

Question (4)

(a) Explain with drawing the wavelength division multiplexing (WDM) system. State the advantages of the WDM.

(b) Consider a simplex linear bus network consisting of $N = 4$ stations. Using the largest distance power budget:

1. Draw the network.
2. Derive the expression of the output power $P_{1,N}$ in terms of the input power P_o if the output power is taken from the final coupler tap (N=4).

Question (5)

(a) Draw the block diagram of the OFDM modulator and demodulator. Explain the multicarrier modulation process. Draw the OFDM signal spectrum.

(b) Define the channel delay spread and the inter-symbol interference (ISI). Explain how the OFDM overcomes the ISI problem.

(c) State the disadvantages of the OFDM.

With my best wishes

Tanta University
Faculty of Engineering
Electronics and Electrical Communications Dept.

Final exam (2012/2013)
Optical Communications

3rd year

Answer the following five questions: **(Time: 3 hours)**

Question (1)

(a) **Briefly explain the following points:**

1. Base line wander problem of NRZ line coding format. Explain how to solve this problem using one technique.
2. Single mode fiber is used to solve the modal noise problem, but the presence of splices may generate a modal noise. (Explain).

(b) Consider an optical link consists of a LED with output power of -13dB coupled into a fiber flylead. A silicon pin receiver with a responsivity of $R = 0.65\,A/W$ and the minimum generated photocurrent $I_P = 41\mu A$. Two connectors at the ends each has a loss of $0.75dB$. A single splice at the middle of the fiber link has a loss of $0.5dB$. The fiber attenuation is $\propto_f = 3.5dB/km$. The system margin is $6dB$.

1. Draw the system model.
2. Determine the *pin* receiver sensitivity in dB.
3. Find the length of the transmission path.
4. Represent the link power budget graphically.

Question (2)

(a) Draw the block diagram of the multichannel amplitude modulation system. Explain the operation of each block. State the disadvantages of the multichannel transmission.

(b) Consider an SCM system having 120 channels, each modulated at 2.3 %. The link consists of 12 km of single mode fiber having a loss of 1 dB/km, plus a connector having a 0.5 dB loss on each end. The laser source couples 2 mW of optical power into the fiber and has $\mathrm{RIN} = -135\ \mathrm{dB/Hz}$. The pin photodiode receiver has a responsivity of 0.6 A/W, front face reflection coefficient of 0.5, bandwidth $B = 5GHz$, $\mathrm{I_D} = 10\ \mathrm{nA}$, $\mathrm{R_{eq}} = 50\Omega$, and $\mathrm{F_t} = 3\ \mathrm{dB}$. Find the carrier-to-noise ratio of the system.

Question (3)

(a) Explain with drawing the external writing technique used to fabricate the Bragg reflection filter.

(b) How to use the Bragg grating filter as a de-multiplexer.

(c) Assume that the input wavelength of 2×2 silicon MZI are separated by $10\,GHz$ ($\Delta\lambda = 0.08nm$ at $\lambda = 1550nm$):

Question 3: (Answer two points only)

a- Explain a speech encryption algorithm. Show how speech enhancement, deconvolution, signal separation, watermarking, and encryption can be used to achieve a high level of security with a good performance in automatic speaker identification systems.

b- Derive the mathematical model for blind separation of speech signals.

c- What are the main properties of the SVD? Why is it used for watermarking? In your opinion, can it be used for speaker identification? Why?

Question 4: (Answer two points only)

a- What is meant by the directivity and directivity index of an acoustic source? Explain how they are measured. Give examples for a directive and non directive sources and calculate the directivity and directivity index for these sources.

b- Explain the following phenomena for sound; transmission through three media, reflection, refraction, diffraction, scattering, and interference.

c- Explain the process of sound filtering. How are the different types of filters constructed? Can this process be used as pre-processing step for speaker identification systems? Why?

Question 5: (Answer two points only)

a- Explain the reverberation phenomenon and how it creates a reverberation chamber. What are the benefits and drawbacks of this phenomenon?

b- Explain the requirements for the design of acoustic rooms and studios.

c- What are the mechanisms for noise insulation and reduction in buildings?

Question 6: (Answer two points only)

a. Explain with aid of sketches the construction of the loudspeaker and the function of each component in it.

b. Explain with aid of sketches the construction of the carbon microphone and the function of each component in it. Show also how ultrasonic signals can be used for underwater applications.

c. In your opinion, can facial expressions be used with speech features in security systems? Why?

Best wishes.

Tanta University (Acoustics and Ultrasonics)
Faculty of Electronic Engineering 3rd Year
Dept. of Electronics and Electrical Comm. Date: 14-6-2012
Time allowed : 3 hours

Answer the following questions:

Question 1: (Answer two points only)

a- Sketch the block diagram of the adaptive Wiener filter for speech enhancement and explain the mathematical model for this filter. Why is this filter performance better than the spectral subtraction and Wiener filtering method?

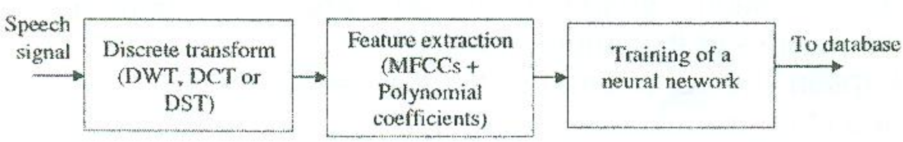
b- What are the metrics that can be used for quality evaluation of audio signals? State the physical meaning of each of these metrics.

c- Why are polynomial shape coefficients used in addition to the MFCCs for speaker identification? Explain how MFCCs and polynomial coefficients are extracted.

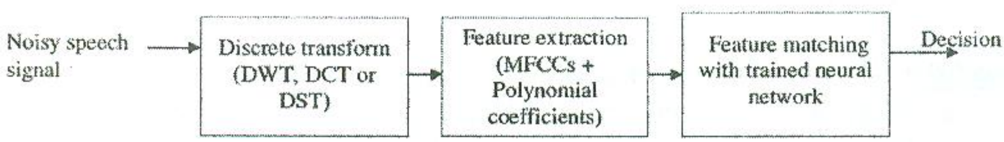
Question 2: (Answer two points only)

a- What is the difference between speech enhancement and speech deconvolution? Do they have an effect on automatic speaker systems? Why? Explain mathematically a speech deconvolution algorithm.

b- The following block diagram is for an automatic speaker identification system. What is the function of each block?



(a) Training Phase.



(b) Testing phase.

c- Explain the steps of the SVD audio watermarking algorithm? Why is the watermark embedding in the segments of the audio signal preferred to watermark embedding in the signal as a whole?

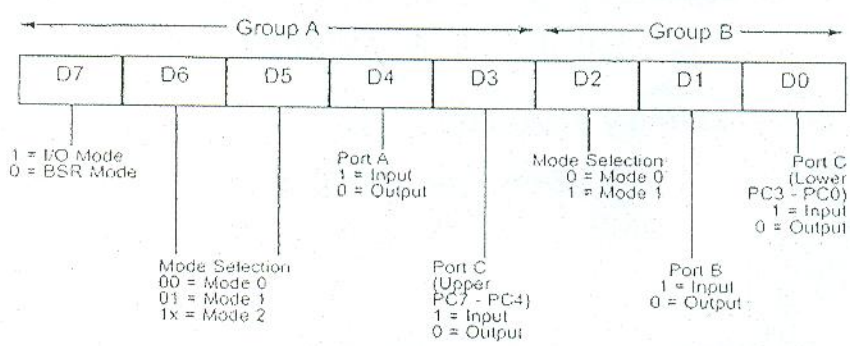
Assume that the keyboard is connected to port A and the printer is connected to port B. Design the hardware interface and write a 8086 assembly language program to accomplish this. Using the full decoding technique, decode the port addresses to be as follows: PortA → 60H, PortB → 68H, PortC → 70H, Control Register → 78H. Use the XLAT instruction.

__Third Question:__ *(25 marks)*

__(a)__ Identify each of the following 8086 instructions as aligned or misaligned transfer. Briefly explain for each case how data will be transferred via which data pins (D15- D0) by the 8086: MOV AL, [4007H] - MOV CX, [005EH].

__(b)__ Using the polled technique to handle interrupts from 4 Analog to Digital Converter devices that share one interrupt line, design a circuit that achieves interrupt priorities using only 2 bits of port A and 2 bits of port B. Use a decoder and an encoder if required, and explain the operation of the circuit.

__(c)__ An 8086-based voltmeter is designed to measure DC voltage signals in the range 0 to 8V and display the result in three decimal digits: One integer part and two fractional parts. The microcomputer is required to start the A/D converter at the falling edge of a "start conversion" pulse. When the conversion is completed, the A/D's "conversion complete" signal will go to HIGH. Then, the microcomputer will send a LOW on the A/D converter's "OUTPUT ENABLE" line and input the 8-bit output from A/D via port A and display the voltage (0 to 8 V) in three decimal digits (one integer and two fractional) via port B on three TIL311 displays. Design the hardware interface and write the required assembly language program for this voltmeter.



Specialization: Communications Dept.	Tanta University Faculty of Engineering	Academic Year: 2011/2012
Form: Third Year		Final Exam
Subject: Microprocessor Applications		Date: 10 - 06 - 2012
Examiner: Dr. Zeiad El-Saghir		Time: 180 min

Questions for Final Written Examination

No of Questions: 3 — No of Pages: 2

Answer the Following Questions:

First Question: (25 marks)

(a) What is meant by "foldback" in linear decoding?

(b) What is the basic difference between cycle-stealing and interleaved DMA?

(c) Interface a microprocessor with 16-bit address pins and 8-bit data pins to two 512 x 8 EPROM chips and four 256 x 8 RAM chips to obtain the following memory map:

EPROM chips	0000H - 03FFH
RAM chips	0400H - 04FFH

Assume that both EPROM and RAM chips contain two enable pins; CE0' and OE' for EPROM chips, CE' and WE' for RAM chips.

(d) Write an 8086 assembly program to move 200 words from a source memory block starting at offset 0250H in ES to a destination memory block starting at offset 0200H in the same extra segment. Use string instructions.

Second Question: (25 marks)

(a) What is the purpose of the 8086 MN/MX' pin?

(b) Determine the addressing modes for the following 8086 instructions: CLC - MOV AX, DX - ADD [SI], BX - SUB [BX][DI], BX.

(c) Write an 8086 assembly program to solve the equation $Z = [2Y^2 - X^2]/34$, where Y and X are two signed 8-bit numbers stored in data segment at offsets NUM1 and NUM2. Store the 16-bit result in extra segment at offset RESULT.

(d) A 8086/8255-based microcomputer is required to input a number from 0 to 9 from a Binary keyboard interfaced to it and output to an ASCII printer.

4- a- Write the general expression for the far field radiation from the long wire antenna then, obtain an expression for the far field pattern of $\lambda/2$ dipole.

b- Plot the radiation pattern and evaluate the directivity, radiation resistance , effective length and beam width of the following linear antennas:

i- A dipole of length 50 cm, operating at 300 MHz

ii- A monopole of height 75 m, operating at 1 MHz

iii- A folded dipole of length 5 cm, operating at 300 MHz.

5- a- Write down an expression for the far field radiation from a circular loop antenna that placed in the x-y plane assuming constant current , then plot the far field patterns and evaluate R_{rad} ,D and beam width for the following cases:

i- A loop antenna of radius 30 cm operating at 1500 MHz.

ii- A loop antenna of radius 10 cm operating at 300 MHz.

b- Write down the far field radiation pattern of a traveling wave antenna with length L, and then, Plot its pattern if its **length is** $L=4\lambda$ tabulating the directions of nulls, peaks and relative peak amplitudes..

" ربي اشرح لي صدري ويسر لي أمري"

Dr. Abdel-Fattah A. Abu-Hashem

محمد كار كروه نتائج ليبيه

الاتصالات

TANTA UNIVERSITY
FACULTY OFENGINEERING
ELECTRONICS&COMMUNICATIONS DEP.

JUNE 2012
TIME ALLOWED: 3 HOURS
3rd YEAR COMMUNICATION

FINAL EXAMINATION
SUBJECT: WAVE PROPAGATION II

1-a- Compare the received signal levels (in dB) of the surface wave component at a distance of 50 Km assuming the ground parameters as $\sigma = 0.01$ *and* $\varepsilon_r = 36$ if frequency is 1.2 MHz.

b- Write down an expression for and sketch the space-wave attenuation function due to ground reflection in the UHF frequency limit, then, derive the expression for the maximum line-of-sight (MLOS) distance due to earth's curvature, indicating clearly the correction made to the obtained MLOS distance due to the effect of the tropospheric refraction.

c- Write down short notes about the standard atmosphere, Ducting and critical refraction modes indicating the meteorological conditions affecting each of them.

d- A 600 MHz radio wave link is operating over a distance of 60 km using antennas heights of 25 m at transmitter and 36 m at receiver. Check the availability of the link in the following cases $i-\frac{dN}{dh} = -0.08 \quad m^{-1}$ $\quad ii-\frac{dN}{dh} = -01 \quad m^{-1}$, if not show how to ensure the system availability., then estimate both the path and space losses factors in dB if the transmitting antenna gain is 1000 and that of the receiving one is 35 dB

2-a- Define the critical frequency, Fc , and the maximum usable frequency , MUF , of an ionospheric layer. What is the range on earth over which the waves can be received due to the reflection from an ionospheric layer

b- If the ionization profile of the ionosphere can be approximated by:

$$N(h) = 4x10^{12}\, tri\frac{h-250}{200} + 5x10^{12}\, tri\frac{h-250}{100}$$ where h is the height in Km,

i- Sketch the ionization profile $N(h)$.

ii- calculate F_c ,and the maximum usable frequency, MUF of the layer for **normal incidence**.

iii- If a plane wave is incident at an angle $\theta_i = 60^0$, check if the following frequencies can be reflected or not. If they can, calculate the height of reflection and the range on earth over which the waves can be received, and if they can't, estimate another suitable value for θ_i

$f_1 = 20\,MHz$, $f_2 = 40\,MHz$ *and* $f_3 = 60\,MHz$.

3- a- Show briefly the main parameters affecting the satellite communication system, then, show why the microwave band is selected for its operation.

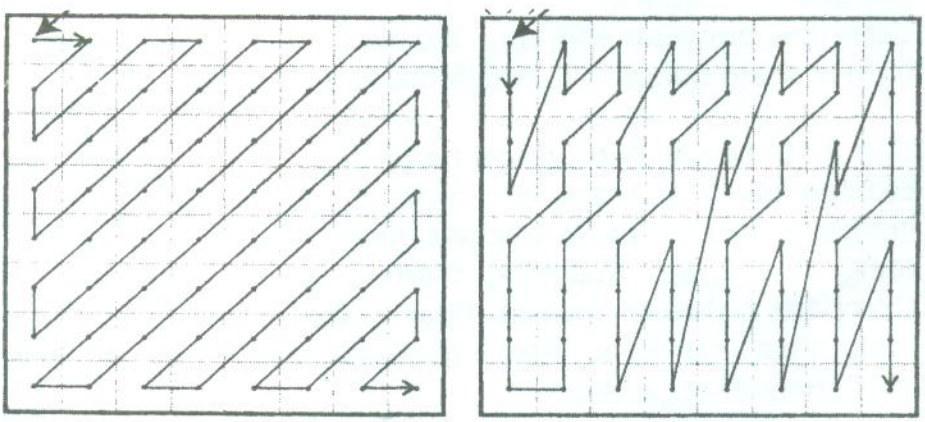
b- For the INTELSAT IV system, calculate the required transmitted power to ensure the reception of 10^{-14} watts through the third transponder with the following parameters: the gain of the transmitting earth station $G_1 = 30\,dB$, the gain of the receiving earth station $G_2 = 25\,dB$, the gain of the receiving antenna of the satellite is $G_{sr} = 20\,dB$, the gain of the transmitting antenna of the satellite is $G_{st} = 20\,dB$ and the effective gain of the satellite system $G_0 = 40\,dB$

150	80	40	14	4	2	1	0
92	75	36	10	6	1	0	0
52	38	26	8	7	4	0	0
12	8	6	4	2	1	0	0
4	3	2	0	0	0	0	0
2	2	1	1	0	0	0	0
1	1	0	0	0	0	0	0
0	0	0	0	0	0	0	0

(a)

1	1	2	4	8	16	32	64
1	1	2	4	8	16	32	64
2	2	2	4	8	16	32	64
4	4	4	4	8	16	32	64
8	8	8	8	8	16	32	64
16	16	16	16	16	16	32	64
32	32	32	32	32	32	32	64
64	64	64	64	64	64	64	64

(b)



(c) (d)

Fig. (1).

b- Sketch the block diagram of the process of super resolution reconstruction of images. If we have a discrete sequence $f(x_k)$ of length N as shown in Fig. (2-a), and this sequence is filtered and down sampled by 2, we get another sequence $g(x_n)$ of length N/2 as shown in Fig.(2-b). The interpolation process aims at estimating a sequence $l(x_k)$ of length N as shown in Fig.(2-c), which is as close as possible to the original discrete sequence $f(x_k)$. Assume bilinear interpolation with:

$$\beta(x) = \begin{cases} 1-|x| & |x| < 1 \\ 0 & 1 \le |x| \end{cases}$$

Tanta University (Selective Course: Signal Processing)
Faculty of Electronic Engineering 3rd Year
Dept. of Electronics and Electrical Com. Date: 17-6-2012
Time allowed : 3 hours

Answer the following questions:

Question 1: **(Answer two points only)**

a- What is the difference between homomorphic filtering and median filtering? Explain both of them.

b- Compute the impulse response $h(n)$ for n = 0, 1, 2, 3, 4 of the digital system defined by the following I/O equation:

$$y(n) = 2x(n) - 2x(n-1) + 0.5x(n-2)$$

Find also the transfer function and frequency response. Is this system stable or not? Why?

c- Compare between FIR and IIR filters. If we can implement both of them adaptively, which one is preferred? Why?

Question 2: **(Answer two points only)**

a- Explain the blur types and their origins. Why is a blurred image simulated by a lowpass filtered image? Explain the residual spectral matching method for blur identification. Compare between inverse filter, LMMSE, and regularized restoration methods. Also, compare between single and multichannel restoration.

b- For the filter with transfer function $H(z) = \left(\frac{1}{2}\right)(1 + z^{-1})$, find the impulse, magnitude, and phase responses, and determine its type whether lowpass or highpass. Does this filter achieve a linear phase? Why?

c- Explain how the image super resolution is performed based on multichannel restoration.

Question 3: **(Answer two points only)**

a- Fig.(1.a) shows the dct coefficients of an 8 × 8 image block. If the dc value of the previous block is 130, use the quantization table in Fig.(1.b) to find the quantized dct coefficients. What is the dc value of this block? Use the two scanning methods in Figs.(1.c) and (1.d) to find the transmitted coefficients. Compare between these two scanning methods? Which one is better? Why?

TANTA UNIVERSITY FACULTY OF ENGINEERING DEPARTMENT OF ELECTRONICS & ELECTRICAL COMMUNICATION EXAMINATION (*third* YEAR)			
COURSE TITLE: Digital communication systems			COURSE CODE: EEC 3220
DATE: 7/6/2012	TERM: SECOND	TOTAL ASSESSMENT MARKS:90	TIME ALLOWED: 3 HOURS

Answer the following questions

PROBLEM # ONE

I. Compare between natural and flat top sampling from time domain signal, signal spectrum, bandwidth then discuss how baseband signal could be recovered from both of them.

II. Discuss different types of noise appear in PCM then show how you can reduce each of them.

III. A PCM system uses a uniform quantizer followed by a 7-bit encoder. The bit rate of this system is equal to 50×10^6 b/s.

a) What is the maximum message bandwidth for which the system operates satisfactorily?

b) Determine the signal to noise ratio when a full-load sinusoidal modulating wave of frequency 1 MHz is applied to the input.

PROBLEM # TWO

I. Determine the 12-bit linear code, 8-bit code, recovered 12-bit code hence error occurred for 0.01v resolution and analog sample voltage of 0.05v, 0.32v and 10.23v.

II. Derive an expression for the power spectral density of AMI code then discuss how filling sequence is used to maintain code transparency.

III. Draw the block diagram for both differential PCM and differential coding showing their main usage in communication systems.

PROBLEM # THREE

I. Draw block diagram of 16QAM coding then explain why M-ary coding is extensively used in today's communication systems.

II. What are the main conditions for minimum shift keying and what its main advantages?

PROBLEM # FOUR

I. For PN code generator consists of three flip flops which have initial value of 100

a. Find and sketch the PN code generator signal and its power spectral density.

b. If this code generator was used in DS-SS system estimate the approximate PSD hence, deduce signal bandwidth.

II. Draw block diagram for FH-SS transmitter and receiver.

Good Luck,

Dr. Salwa Serag Eldin