King Fahd University of Petroleum & Minerals

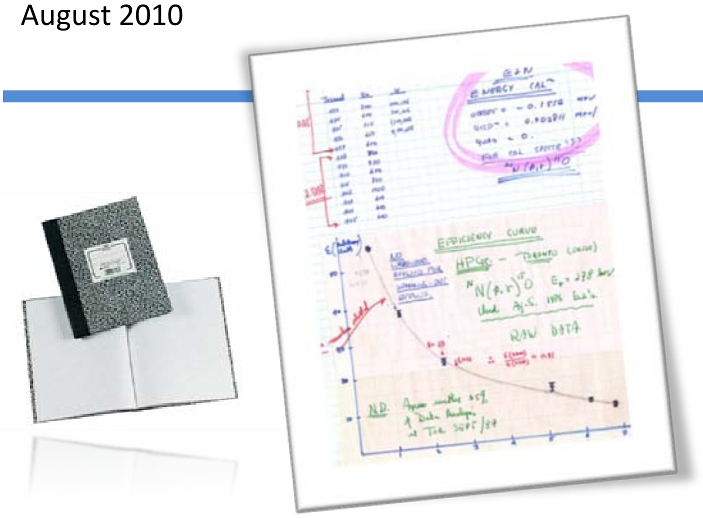
Electrical Engineering Department

EE370 Communications Engineering

LAB Sheets

Dr. Maan A. Kousa & Dr. Ali H. Muqaibel

August 2010



Contents

Lab Sheet 1: Getting Familiar with the Laboratory Equipment

Student ID		Student Name	
Date		Instructor Signature	

1. Which of the following equations can be implemented using the ADDER and which cannot? Explain you answer for each.

$-2\cos(2\pi\ 2\text{x}10^6 t) - 1.5\cos(2\pi\ 2\text{x}10^5 t)$

$-1.3\cos(2\pi\ 2\text{x}10^4 t)x(t) - 0.5\sin(2\pi\ 2\text{x}10^3 t)$

$-2.5\cos(2\pi\ 2\text{x}10^4 t)x(t) - 10.5\sin(2\pi\ 2\text{x}10^3 t)$

$1.3\cos(2\pi\ 2\text{x}10^4 t)x(t) + 0.5\sin(2\pi\ 2\text{x}10^3 t)$

2 & 3. Fill in the table for the signals of The MASTER SIGNALS module.

Signal	Measured frequency	Amplitude
100 kHz sin		
8.3 kHz Clock		
2 kHz TTL		
2 kHz sinusoid		

4. Draw the block diagram that generates the signal $3\cos(2\pi x 2x10^3 t) + 6$ V

Instructor Verification of waveform_____________

5. Observe and plot the spectra of each of the four signals of the MASTER SIGNALS module.

   a. Do the spectra plots coincide with your expectations? Explain.

   b. How far is the noise level below the signal level?

6. Explore the following features and settings of the picoscope. Tick the one you were successful to do.

Feature	Tick
Switch between oscilloscope and frequency analyzer on the same view	
Display one or both channels on the same view (window)	
Separate the two channels on the same view so that they are non-overlapping (do it manually and auto)	
Change the setting of the axes.	
Take a snap shot or continuous scan	
Zoom in a specific segment of the graph	
Display measurements of DC value, frequency, period, …	
Use horizontal and vertical markers	
Set the oscilloscope on external triggering	
Create time view and spectrum view and save them	

Other Comments and Observations

Lab Sheet 2: Simulation of Communication Systems Using MATLAB

Student ID		Student Name	
Date		Instructor Signature	

What would m.*c produce? What would happen if you remove the dot?

How many arguments would a BPF require? What are they?

Change m(t) to 2+ sin(2π 1000t) and c(t) to cos(2π 10^4) and the cutoff frequency of the filter to 2 kHz. Write the m-file for the given system.

Other Comments and Observations

Lab Sheet 3: Representation of Signals & Systems

Student ID		Student Name	
Date		Instructor Signature	

Fill in the missing blocks in the table

Property	Time	Frequency
Modulation	$g(t)cos(\omega_0 t)$	
Time Scaling	$g(at)$	

Part I: Verification of Fourier Transform Properties

Sketch equivalent module diagram and connection for the block diagram in the manual.

Comment on the noise level and harmonics in the spectrum plot of m(t).

Describe the effect of varying the frequency of x(t) in light of the time scaling property.

What is the property we are trying to prove in step 8?

Compare the spectrum of $m(t)$ and $z(t)$ and comment on the modulation property.

Part II: Filtering of periodic Signals

Observe the input and the out of the filter in both time and frequency domain. Are they similar? Why?

Explain the effect of varying the filter cutoff frequency on the output waveform.

Part III: System Identification

Instructor Inspection for part 4 ___________________

Why the spectrum consists of spectral lines and not continuous curve? What controls the spacing between the spectral lines?

What is the type of the filter in each case? Estimate the 3dB bandwidth of the filters.

Setting	Filter Type	Bandwidth
1		
2		
3		

Other Comments and Observations

Lab Sheet 4: Speech Signals

Student ID		Student Name	
Date		Instructor Signature	

Part I: Audible Range of our Hearing System

Find the lowest frequency and highest frequency over which you can hear the sound.

Lowest Frequency	
Highest Frequency	

Part II: Spectrum of Speech Signals

Estimate the bandwidth of the speech signal on CHANNEL 1. (Estimate the noise floor, and consider the spectrum above the noise floor).

Noise Floor	
Bandwidth	

How is the spectrum of CHANNEL 2 different from that of CHANNEL 1?

Part III: Filtering the Speech Signal

Comment on the effect of varying the LPF cutoff on the spectrum and the sound of the signal on CHANNEL 2.

Part IV: Frequency Translation (modulation)

Comment on the effect of varying the frequency of the signal generator on the spectrum and sound of CHANNEL 1.

Other Comments and Observations

Lab Sheet 5: DSBSC Modulation & Demodulation

Student ID		Student Name	
Date		Instructor Signature	

Part I: DSBSC Generation

Sketch the module diagram to generate DSBSC.

Part II: DSBSC Demodulation

Is the spectrum before the LPF filter in the demodulator circuit what you expected? Explain.

Find the range of acceptable values for best recovery of the message.

Lowest acceptable cutoff	
Highest acceptable cutoff	

Increase the cutoff frequency of the LPF beyond the range of good recovery. What happens to the recovered signal? Why?

Part III: Effect of Phase Mismatch

Describe the effect on phase mismatch on the recovered signal.

Part IV: Effect of Frequency Mismatch

Describe the effect of frequency mismatch on the single-tone message.

Describe the effect of frequency mismatch on the speech signal.

Other Comments and Observation:

Lab Sheet 6: AM and QAM

Student ID		Student Name	
Date		Instructor Signature	

Part I: AM

Comment on the quality of single-tone message recovered by RC filter and the HEADPHONE AMPLIFIER filter (steps 4 and 5).

Comment on the quality of speech recovered by RC filter and the HEADPHONE AMPLIFIER filter. What is the lesson learned?

Part II: QAM Modulation and Demodulation

Instructor Inspection of the modulator and demodulator: __________________________

Part III: Effect of Phase error on QAM

Describe the effect on phase error on the recovered signal.

Other Comments and Observations

Lab Sheet 7: FM Modulation

Student ID		Student Name	
Date		Instructor Signature	

Part I: Sensitivity and Linearity of VCO

V_{in}	**-2**	**-1.5**	**-1**	**-0.5**	**0**	**0.5**	**1**	**1.5**	**2**
Freq with 25%									
Freq with 60%									

Plot the obtained frequency versus the input voltage for each setting. (You can use Excel or MATLAB). You can include a hard copy of the plot in your report, but for now looking at the soft copy:

- Which of the above settings results in a more linear performance in the given range of V_{in}? 25 % or 60 %

- Determine the linear range for the second case (60% setting)

- Using the table only, estimate the frequency of the VCO when the DC input is 1.75 V for both settings? Which setting results in easier interpolation? Why?

Part II: Setting the Frequency Deviation

Measured frequency with +2 V input ___________

Part III: FM Generation

With the AUDIO AOSCILLATOR connected, why the frequency counter is still at 100 kHz?

Vary the frequency of the AUDIO AOSCILLATOR. Explain the change in the modulated signal.

Vary the GAIN of VCO. Explain the change in the modulated signal.

Part IV: Spectrum Analysis and Bandwidth Estimation

Vary the message frequency and describe the impact on the spectrum of the FM signal.

Reset the frequency of the message to 2 kHz, and vary the deviation ratio (by varying the GAIN in the VCO). Describe the effect on the spectrum of the FM signal.

Explain the obtained spectra in light of Carson's Rule for bandwidth estimation.

Other Comments and Observations

Lab Sheet 8: FM Demodulation

Student ID		Student Name	
Date		Instructor Signature	

Part I: FM Demodulation Using PLL

Draw the module diagram corresponding to the block diagram of PLL in Figure (1).

Examine the output of the PLL VCO and compare it with the original message.

Effect of varying the frequency f_0 of the PLL-VCO on the quality of the received speech.

Effect of varying the GAIN of the PLL-VCO on the quality of the received speech.

Part II: Frequency Discriminator

Demonstrate the operation of the frequency discriminator to your Lab instructor.

Instructor Approval ___________________________

Other Comments and Observations

Lab Sheet 9: PCM Encoding

Student ID		Student Name	
Date		Instructor Signature	

Part I: PCM Frame Structure

frame duration	Bit duration	codeword duration

What is the sampling rate of the PCM ENCODER? ___________________

Is it appropriate to sample a speech signal with this rate? Why?

Part II: Quantizing levels for 4-bit linear encoding

Describe the binary sequence of the PCM DATA for three consecutive frames. Give reasons for such sequence

Gradually increase the amplitude of the DC input signal until you notice a change to the PCM output. Record the binary sequence of the new digital word, and the input amplitude at which the change occurred.

DC								
Binary								
DC								
Binary								

Input-out characteristics of the linear quantizer.

Part III: Non-Linear Quantization

Gradually increase the amplitude of the DC input signal until you notice a change to the PCM output. Record the binary sequence of the new digital word, and the input amplitude at which the change occurred.

DC								
Binary								
DC								
Binary								

Input-out characteristics of the non-linear quantizer.

Part IV: Time-Varying Messages

Observe the PCM DATA output over consecutive frames. How it is different from the DC input case?

Other Comments and Observations

Lab Sheet 10: PCM Decoding

Student ID		Student Name	
Date		Instructor Signature	

Part I: System Setup

Alternating '0' and '1' bits (for remote FS) in the LSB position should be 1920 ms apart. Verify by measurement and calculation.

Part II: Quantization Effects

As the input to the encoder moves continuously, the output from the decoder moves in discrete jumps. Explain this behavior.

Reset the coding scheme on both modules to 7-bit. Notice the 'granularity' in the output is almost unnoticeable compared with the 4-bit case. *Comment*.

Part III: Signal Recovery

Measure the delay between the input and the output.

How many samples are taken in one period of the input signal?______________

Explain how the delay affects the error.

Change the coding scheme from 4-bit to 7-bit, explain how this affects the error.

Find the cutoff frequency for best possible recovery of the rectangular signal of 20 Hz.

Adjust the cutoff frequency of the TUNABLE LPF to 50Hz, then 500Hz. Comment on the type of distortion in the output signal.

Part IV: Companding

Examine all the possible scenarios in the table below in terms of quality, recoverable/non-recoverable, loud/low with reference to the first row. First guess the output, and then verify your guess experimentally.

Encoder	Decoder	Your Expectation	Experimental Observation
4-bit linear	4-bit linear		
4-bit non-linear	4-bit non-linear		
7-bit linear	7-bit linear		
4-bit linear	4-bit non-linear		
4-bit non-linear	4-bit linear		
7-bit linear	4-bit linear		

Was the speech signal recovered satisfactorily in any case? Justify

Other Comments and Observations

Lab Sheet 11: Line Coding

Student ID		Student Name	
Date		Instructor Signature	

Part I: Line Codes Definitions and Spectra

Observe the different Line codes on the scope. Explain what each code does (the law of transformation).

Line Code	Law of Transformation (Definition)
NRZ-L	
NRZ-M	
UNI-RZ	
BIP-RZ	
RZ-AMI	
BiØ-L	
DICODE-NRZ	

Observe the spectra of the following two line codes: NRZ-L and BiØ-L (Manchester). Compare them in terms of DC content and first null bandwidth.

Line Code	DC Content	First Null Bandwidth
NRZ-L		
BiØ-L		

How is this BW related to the time domain waveform?

Part II: LINE Decoding

Are the generated and recovered waveforms identical?

The introduced delay________________

Part III: Signal Inversion Test

Identify the codes that are insensitive to channel inversion.

Line Code	NRZ-L	NRZ-M	UNI-RZ	BIP-RZ	RZ-AMI	BiØ-L	DICODE-NRZ
Sensitive (Y/N)							

Other Comments and Observations

Lab Sheet 12: Digital Modulation: FSK

Student ID		Student Name	
Date		Instructor Signature	

Part I: Generation Continuous Phase FSK, (CPFSK)

Is the phase at the bit-transition continuous? ____________________

Toggle Switch Low		Toggle Switch High	
f_1	f_2	f_1	f_2

Part II: General Method of FSK Generation

Vary f_2. How does that affect the waveform and the spectrum of the modulated signal?

Set f_s to minimum then to maximum. Which setting generates a better modulated signal? Why?

Zoom in to a transition between any two bits. Is the phase continuous? Why?

How does that affect the bandwidth?

Set the VARIABLE DC to zero. What type of modulated signal is generated? Why?

How can you use the above system to generate PSK?

Other Comments and Observations