

ANALOG & DIGITAL COMMUNICATION

LAB MANUAL



DEPARTMENT OF ELECTRICAL ENGINEERING,
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Lab Manual of Analog & Digital Communication

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Table of Contents

Sr. No.	Description	Page No.
1	List of Equipment	4
2	Experiment No.1, Generation of noise and observations of its effect on a sinusoidal signal	5
3	Experiment No.2, Generation of AM signals	7
4	Experiment No.3, Demodulation of AM signals	9
5	Experiment No.4, Frequency modulation	11
6	Experiment No.5, Introduction to phase locked loop	20
7	Experiment No.6, FM Demodulators	31
8	Experiment No.7, Digital Modulation: FSK	34
9	Experiment No.8, Pre-emphasis & De-emphasis	36
10	Experiment No.9, Pulse code modulation	39
11	Experiment No.10, MATLAB basic for communication system design	41
12	Experiment No.11, Communication Signals :Generation and Interpretation	48
13	Experiment No.12, Communication Signals: Operations	57
14	Experiment No.13, Introduction to amplitude modulation (SIMULINK implementation)	66
15	Experiment No.14, Introduction to amplitude modulation (MATLAB implementation)	84
16	Appendix A: Lab Evaluation Criteria	95
17	Appendix B: Safety around Electricity	96
18	Appendix C: Guidelines on Preparing Lab Reports	98

List of Equipment

Sr. No.	Description
1	RIMS Communication trainer DEV-2786
2	Function generator
3	Oscilloscope
4	Digital Multi-meter
5	Power supply
6	IC XR-2206
7	IC CD4046
8	IC LM565
9	Capacitors
10	Resistors
11	Diode
12	Probes

EXPERIMENT # 1

Generation of noise and observations of its effect on a sinusoidal signal

Objective

- Familiarize students with the contents of the experiment and give them hands on experience regarding the experiment
- To learn the effect of noise on sinusoidal signal
- Observe the filter output

Apparatus

- Communication trainer DEV-2786
- Oscilloscope
- Probes
- Connecting wires

Theory

The unwanted signal that gets introduced in a signal when it passes through any communication system is termed as Noise. Generally Noise is classified with respect to its origin as internal or external noise; the internal noise is generated by the components of the communication system itself while the external noise is added to the signal due to the external fields developed due to other communication systems, power lines or even due to human interference. With proper care the external noise can be minimized and can be even removed, similarly with proper care the internal noise can be minimized but can never be eliminated. Noise is one of the basic factors that limit the communication systems in terms of their performance. One of the important parameters to observe is Signal to Noise ratio (S/N) which plays very important role in any communication system. Shannon capacity defines maximum possible data rate for systems with noise and distortion. In white Gaussian noise channels, the capacity of the channel can be given by $C = B \log(1 + S/N)$.

Noise can also be classified in terms of its spectrum. Thermal noise one of the most common sources of noise is also known as white noise because its spectrum is flat over the range of frequencies. White noise analysis can be done stochastically and white noise is modeled as a random variable with probability density function (pdf) which could be

- Gaussian
- Uniform

Procedure

1. Generate a sinusoidal wave from the signal generation block. Check its amplitude, set it to 2 V p-p by using the amplifier block.
2. Now go to Main menu of trainer and select “Noise” then select the type of noise from the menu, this will enable the noise generation part of the trainer.
3. There are two outputs and one input on the noise generation block, noise can be separately observed from “Noise output block” where as total output after the addition of noise to the signal can be extracted from “signal + noise output” block. Remember to connect the sinusoidal signal that you generated in step1 to the “signal input block”.
4. Get the signal from the noise block and observe it on the oscilloscope.
5. Also observe the output from the signal plus noise block.

6. Using the filter module on the trainer use different filters to remove noise from the signal.

Graphical Analysis:

Sketch the output waveforms of the active and passive low-pass filters as seen on the oscilloscope for both uniform and Gaussian noises. Also mention the time/div and volts/div for each channel.

Design a simple RC low pass filter and compare the output with your earlier observations.

POST LAB

1. Derive the cutoff frequency expression for a first order RC low pass filter?
2. What will happen if we use high order low pass filter?

EXPERIMENT # 2

Generation of AM signals

Objective

- Familiarize students with the contents of the experiment and give them hands on experience regarding the experiment
- To observe the effect of modulation index in amplitude modulation

Apparatus

- Communication trainer DEV-2786
- Oscilloscope
- Function generator
- Probes
- Connecting wires

Theory

An amplitude modulated signal can be obtained by simply multiplying the message signal with the carrier signal; that is, if $m(t)$ is the message signal and $c(t)$ is the carrier signal, then we can write the modulated AM signal as

$$\varphi(t) = m(t) \cdot c(t)$$

or

$$\varphi(t) = m(t) \cdot A_c \cos \omega_c t$$

Therefore we need a simple multiplying unit for amplitude modulation. However there are different types of modulators which include:

1. Multiplier Modulators

These modulators contain a simple analog multiplier circuit whose output is directly proportional to the product of both signals

2. Non-linear modulators

These modulators use nonlinear devices such as semi-conductor diode or a transistor for the purpose of modulation.

3. Switching modulators

These modulators use the principle of switching to obtain the approximate product of carrier and the message signal, usually these modulators use square wave as carrier. Ring modulator is an example of such a modulator.

Procedure

Using RIMS Trainer

1. Generate a 2Vp-p, 100 Hz sinusoidal signal from the function generator block of communication trainer.
2. Generate the sinusoidal carrier signal of 2Vp-p, 5 KHz from function generator.

3. Plug in the signal generated in step 1 to the input of DSB board, block labeled as “Modulating signal input”.
4. Plug in the carrier signal generated in step 2 to the input of DSB board, block labeled as “Carrier Signal Input”.
5. Get the output from the “modulated signal output” block of DSB board and observe it on oscilloscope. Using the following three controls achieve the desired modulation.
 - Amplitude of Modulating Signal (Trainer)
 - Offset of Modulating Signal (Trainer)
 - Carrier Null Setting (DSB Board)

Graphical Analysis:

Sketch the waveforms of modulating signal, carrier signal and modulated signal. Also mention the time/div and volts/div for each channel.

Using Function Generator:

1. Generate a 2Vp-p, 1 kHz sinusoidal signal from the function generator block of RIMS communication trainer.
2. Generate the sinusoidal carrier signal of 2Vp-p, 10 KHz from function generator.
3. Connect the modulating signal generated in step 1 to the input of “MOD IN-OUT” of function generator.
4. To achieve the amplitude modulation using modulation block of function generator, keep the following settings
 - a. Set the on/off button to “on”
 - b. Set the Int/Ext button to “Ext”
 - c. Set the AM/FM button to “AM”
 - d. Keep modulation knob at “Max”
5. Get the modulated output from the function generator and observe it on oscilloscope.
6. Change the voltage level of message signal m_p to see the effect of modulation due to increasing modulation index μ , as per given table

Table 2.1. Modulating Index

Modulation Index $\mu = \frac{m_p}{A_c}$	1	0.75	0.5	0.25	1.25
$A_c(V)$	1	1	1	1	1
$m_p(V)$					

7. Also decrease the frequency of the carrier signal gradually and observe the effect on the modulated signal.

Graphical Analysis:

Sketch the modulated waveforms for all cases of modulation index. Also mention the time/div and volts/div for each channel.

POST LAB

3. What is the difference between DSB-WC and DSB-SC?
4. What is the bandwidth of DSB-SC signal and AM signal?

EXPERIMENT # 3

Demodulation of AM signals

Objective

- Familiarize students with the contents of the experiment and give them hands on experience regarding the experiment
- To demodulate the modulated wave using envelope detector

Apparatus

- Communication trainer DEV-2786
- Resistor
- Capacitor
- Diode
- Oscilloscope
- Function generator

Theory

Demodulation of the amplitude modulated signal can be achieved by using the modulators which are the circuits used for modulation can also be used for demodulation. However we have to make following changes:

1. We have to connect the modulated signal in the place of message signal.
2. Instead of using band pass filter we will use low pass filter

The carrier input is also required. We can classify demodulators as either coherent or non-coherent demodulators. Coherent demodulators require the carrier signal in addition to the modulated signal for demodulating AM signal however non-coherent demodulators do not require carrier signal as input. An example of such a demodulator is known as envelope detector which usually comprises of a diode followed by a RC circuit.

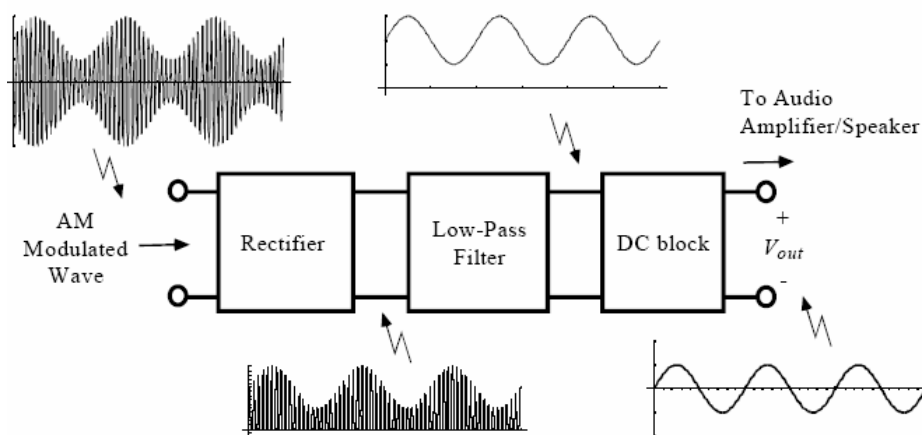


Figure 3.1. Block Diagram of AM demodulation

Procedure

Coherent Demodulation

1. Generate an AM signal as done in part 'a' of previous experiment.
2. Using the mixer module on the trainer, demodulate the signal with carrier wave; note that the carrier should be the same as used in the modulation process.
3. Compare the demodulated signal from the module with the original message signal.

Graphical Analysis:

Sketch the modulated signal and demodulated signal (mixer module) observe on the oscilloscope. Also mention the time/div and volts/div for each channel.

Non-coherent Demodulation

1. Generate an AM signal as done in part 'b' of previous experiment
2. Wire a simple envelope detector on the bread board as shown in circuit; you may skip the Op-Amp part. Note that for an envelope detector

$$RC \leq \frac{1}{\omega_m} \left(\frac{\sqrt{1 - \mu^2}}{\mu} \right)$$

3. Change different parameters of the modulated signal particularly modulation index and observe the effect on the response of Envelope Detector.

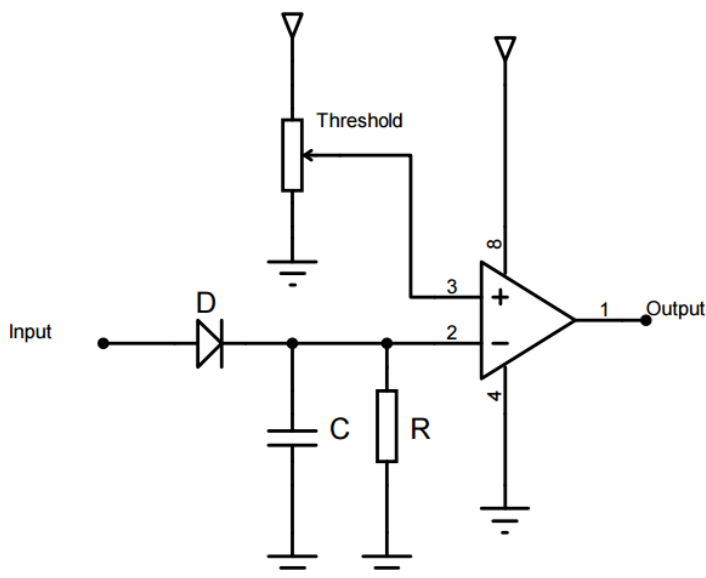


Figure 3.2. Envelope Detector and Threshold Circuit

Graphical Analysis:

Sketch the AM signal (for $\mu = 1$) and the demodulated signal obtained from envelope detector. Also mention the time/div and volts/div for each channel.

POST LAB

1. Explain the difference between coherent and non-coherent detection of AM signals.
2. Can DSB-SC be detected non-coherently with an envelope detector? Explain why.

EXPERIMENT # 4

Frequency Modulation

Objective

- To gain a good understanding of frequency modulation
- Learn to generate frequency modulated wave using RIMS trainer
- Learn to design FM circuit using IC XR2206

Apparatus

- Communication trainer DEV-2786
- Resistor
- Capacitor
- IC XR2206
- Oscilloscope
- Function generator

Theory

The requirement of frequency modulation is to vary the frequency of the high frequency carrier signal according to the amplitude of the message signal. Frequency modulated signal can be generated by using

- Direct method
- Indirect method

For direct method the following techniques are usually used

1. The output frequency of a voltage controlled oscillator is directly proportional to the input voltage. So if a message signal is supplied as input then the frequency of the output signal will contain the amplitude changes of input signal in its frequency. This method is used commonly with a feedback system which can generate an error voltage if the output frequency deviates far from the centre frequency.
2. The frequency can be varied with respect to some message signal if the reactance of an LC circuit is varied in proportion to the magnitude of the message signal
3. A Varactor diode can be used whose reactance changes with the input signal and hence output frequency can be varied

For indirect method, the signal is first integrated and then phase modulated to get the required FM output. However this type of modulation gives rise to Narrow Band FM signal which is then converted to required range and bandwidth by using frequency multipliers and converters.

The modulation index is defined as the ratio of the maximum frequency deviation to the modulating frequency. The maximum frequency deviation is the shift from center frequency f_c when the amplitude of the modulating signal is maximum.

By Carlson's rule $BW = 2(\Delta f + f_m(max))$

Where Δf = Maximum frequency deviation and $f_m(max)$ = Maximum modulating frequency

Procedure

Using RIMS Trainer

1. Obtain a 2V, 1 kHz sinusoidal signal from the function generator block.
2. Select the carrier frequency to 5 KHz.
3. Plug in the signal generated in step 1 to the input of modulator block labeled as “FM-IN”.
4. Get the output from the modulator block which will be frequency modulated.
5. Increase the voltage level of the message signal and observe the effect on modulated signal.
6. Also decrease the frequency of the carrier frequency gradually and observe the effect on the modulated signal.
7. Repeat the experiment for triangular and square wave message signals.
8. Repeat step 7 for carrier signal.

Graphical Analysis:

Draw the message signal, carrier signal and frequency modulated signal as seen on the oscilloscope. Also mention the time/div and volts/div for each channel.

Using XR-2206

Design a Frequency modulator for carrier frequency of 45 KHz using VCO of XR2206. Data sheet of IC XR2206 is attached with this manual. You may not use the other sections of the chip. Observe the frequency modulated signal on oscilloscope and note down the readings in the given table. Find the frequency Deviation and calculate the Modulation index.

Graphical Analysis:

Draw the message signal, carrier signal and frequency modulated signal as seen on the oscilloscope. Also mention the time/div and volts/div for each channel.

Experimental Results:

Table 4.1

	Amplitude	Frequency	
Message signal			
Carrier Signal			
FM Signal		$f_{\max} =$	$f_{\min} =$

$$\Delta f = \underline{\hspace{2cm}}$$

$$\beta = \underline{\hspace{2cm}}$$



XR-2206

Monolithic
Function Generator

June 1997-3

FEATURES

- Low-Sine Wave Distortion, 0.5%, Typical
- Excellent Temperature Stability, 20ppm/°C, Typ.
- Wide Sweep Range, 2000:1, Typical
- Low-Supply Sensitivity, 0.01%V, Typ.
- Linear Amplitude Modulation
- TTL Compatible FSK Controls
- Wide Supply Range, 10V to 26V
- Adjustable Duty Cycle, 1% TO 99%

APPLICATIONS

- Waveform Generation
- Sweep Generation
- AM/FM Generation
- V/F Conversion
- FSK Generation
- Phase-Locked Loops (VCO)

GENERAL DESCRIPTION

The XR-2206 is a monolithic function generator integrated circuit capable of producing high quality sine, square, triangle, ramp, and pulse waveforms of high-stability and accuracy. The output waveforms can be both amplitude and frequency modulated by an external voltage. Frequency of operation can be selected externally over a range of 0.01Hz to more than 1MHz.

The circuit is ideally suited for communications, instrumentation, and function generator applications requiring sinusoidal tone, AM, FM, or FSK generation. It has a typical drift specification of 20ppm/°C. The oscillator frequency can be linearly swept over a 2000:1 frequency range with an external control voltage, while maintaining low distortion.

ORDERING INFORMATION

Part No.	Package	Operating Temperature Range
XR-2206M	16 Lead 300 Mil CDIP	-55°C to +125°C
XR-2206P	16 Lead 300 Mil PDIP	-40°C to +85°C
XR-2206CP	16 Lead 300 Mil PDIP	0°C to +70°C
XR-2206D	16 Lead 300 Mil JEDEC SOIC	0°C to +70°C

XR-2206

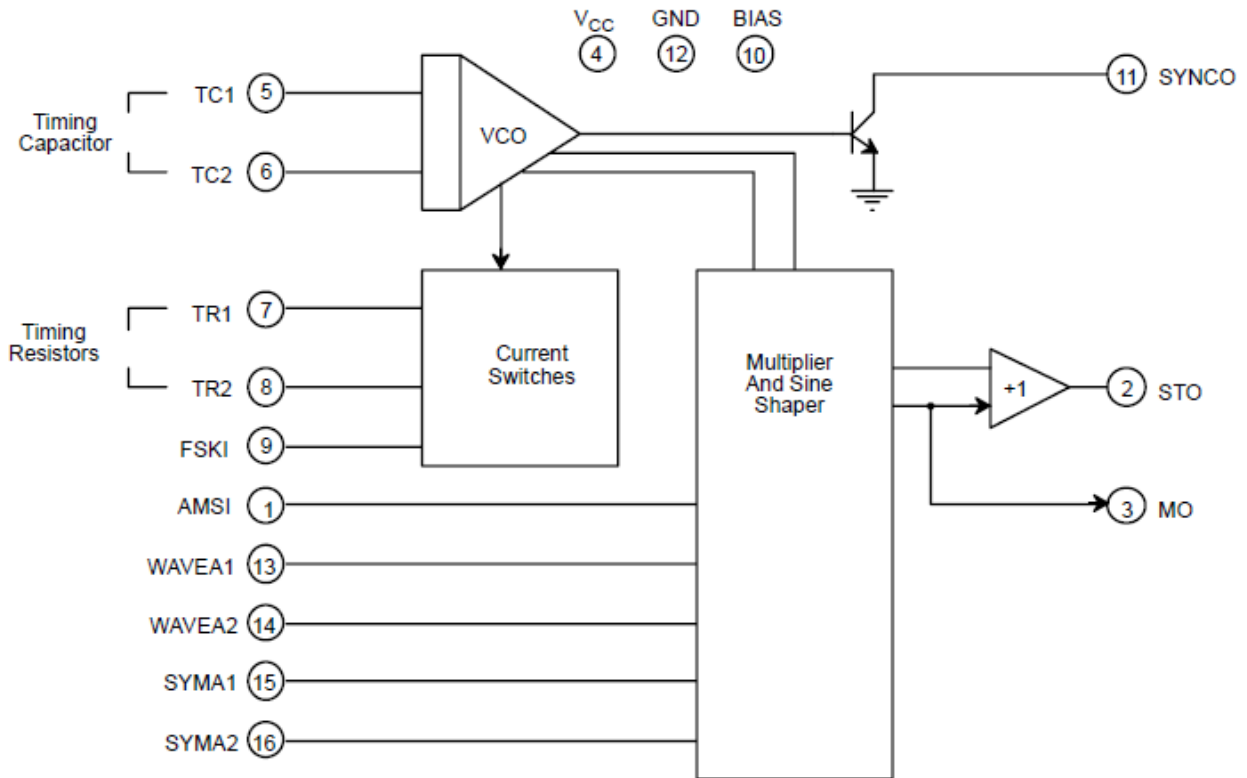
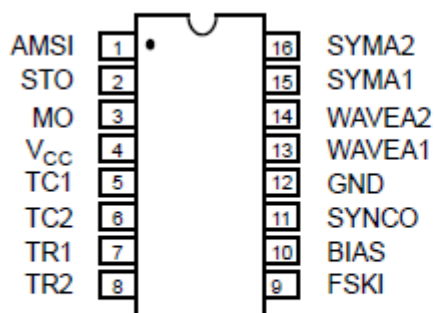
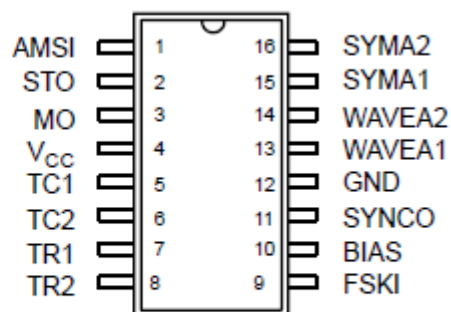


Figure 1. XR-2206 Block Diagram

XR-2206**16 Lead PDIP, CDIP (0.300")****16 Lead SOIC (Jedec, 0.300")****PIN DESCRIPTION**

Pin #	Symbol	Type	Description
1	AMSI	I	Amplitude Modulating Signal Input.
2	STO	O	Sine or Triangle Wave Output.
3	MO	O	Multiplier Output.
4	V _{CC}		Positive Power Supply.
5	TC1	I	Timing Capacitor Input.
6	TC2	I	Timing Capacitor Input.
7	TR1	O	Timing Resistor 1 Output.
8	TR2	O	Timing Resistor 2 Output.
9	FSKI	I	Frequency Shift Keying Input.
10	BIAS	O	Internal Voltage Reference.
11	SYNCO	O	Sync Output. This output is a open collector and needs a pull up resistor to V _{CC} .
12	GND		Ground pin.
13	WAVEA1	I	Wave Form Adjust Input 1.
14	WAVEA2	I	Wave Form Adjust Input 2.
15	SYMA1	I	Wave Symetry Adjust 1.
16	SYMA2	I	Wave Symetry Adjust 2.

XR-2206



DC ELECTRICAL CHARACTERISTICS

Test Conditions: Test Circuit of Figure 2 $V_{CC} = 12V$, $T_A = 25^\circ C$, $C = 0.01\mu F$, $R_1 = 100k\Omega$, $R_2 = 10k\Omega$, $R_3 = 25k\Omega$
 Unless Otherwise Specified. S_1 open for triangle, closed for sine wave.

Parameters	XR-2206M/P			XR-2206CP/D			Units	Conditions
	Min.	Typ.	Max.	Min.	Typ.	Max.		
General Characteristics								
Single Supply Voltage	10		26	10		26	V	$R_1 \geq 10k\Omega$
Split-Supply Voltage	± 5		± 13	± 5		± 13	V	
Supply Current		12	17		14	20	mA	
Oscillator Section								
Max. Operating Frequency	0.5	1		0.5	1		MHz	$C = 1000pF$, $R_1 = 1k\Omega$
Lowest Practical Frequency		0.01			0.01		Hz	$C = 50\mu F$, $R_1 = 2M\Omega$
Frequency Accuracy		± 1	± 4		± 2		% of f_0	$f_0 = 1/R_1C$
Temperature Stability		± 10	± 50		± 20		ppm/ $^{\circ}C$	$0^{\circ}C \leq T_A \leq 70^{\circ}C$
Frequency								$R_1 = R_2 = 20k\Omega$
Sine Wave Amplitude Stability ²		4800			4800		ppm/ $^{\circ}C$	
Supply Sensitivity		0.01	0.1		0.01		%/V	$V_{LOW} = 10V$, $V_{HIGH} = 20V$, $R_1 = R_2 = 20k\Omega$
Sweep Range	1000:1	2000:1			2000:1		$f_H = f_L$	$f_H @ R_1 = 1k\Omega$ $f_L @ R_1 = 2M\Omega$
Sweep Linearity								
10:1 Sweep		2			2		%	$f_L = 1kHz$, $f_H = 10kHz$
1000:1 Sweep		8			8		%	$f_L = 100Hz$, $f_H = 100kHz$
FM Distortion		0.1			0.1		%	$\pm 10\%$ Deviation
Recommended Timing Components								
Timing Capacitor: C	0.001		100	0.001		100	μF	Figure 5
Timing Resistors: R_1 & R_2	1		2000	1		2000	k Ω	
Triangle Sine Wave Output ¹								Figure 3
Triangle Amplitude		160			160		mV/k Ω	Figure 2, S_1 Open
Sine Wave Amplitude	40	60	80		60		mV/k Ω	Figure 2, S_1 Closed
Max. Output Swing		6			6		V _{p-p}	
Output Impedance		600			600		Ω	
Triangle Linearity		1			1		%	
Amplitude Stability		0.5			0.5		dB	For 1000:1 Sweep
Sine Wave Distortion								
Without Adjustment		2.5			2.5		%	$R_1 = 30k\Omega$
With Adjustment		0.4	1.0		0.5	1.5	%	See Figure 7 and Figure 8

Notes

¹ Output amplitude is directly proportional to the resistance, R_3 , on Pin 3. See Figure 3.

² For maximum amplitude stability, R_3 should be a positive temperature coefficient resistor.

Bold face parameters are covered by production test and guaranteed over operating temperature range.

XR-2206**DC ELECTRICAL CHARACTERISTICS (CONT'D)**

Parameters	XR-2206M/P			XR-2206CP/D			Units	Conditions
	Min.	Typ.	Max.	Min.	Typ.	Max.		
Amplitude Modulation								
Input Impedance	50	100		50	100		kΩ	For 95% modulation
Modulation Range		100			100		%	
Carrier Suppression		55			55		dB	
Linearity		2			2		%	
Square-Wave Output								
Amplitude		12			12		Vp-p	Measured at Pin 11.
Rise Time		250			250		ns	C _L = 10pF
Fall Time		50			50		ns	C _L = 10pF
Saturation Voltage		0.2	0.4		0.2	0.6	V	I _L = 2mA
Leakage Current		0.1	20		0.1	100	μA	V _{CC} = 26V
FSK Keying Level (Pin 9)	0.8	1.4	2.4	0.8	1.4	2.4	V	See section on circuit controls
Reference Bypass Voltage	2.9	3.1	3.3	2.5	3	3.5	V	Measured at Pin 10.

Notes

¹ Output amplitude is directly proportional to the resistance, R_3 , on Pin 3. See Figure 3.

² For maximum amplitude stability, R_3 should be a positive temperature coefficient resistor.

Bold face parameters are covered by production test and guaranteed over operating temperature range.

Specifications are subject to change without notice

ABSOLUTE MAXIMUM RATINGS

Power Supply 26V
 Power Dissipation 750mW
 Derate Above 25°C 5mW/°C

Total Timing Current 6mA
 Storage Temperature -65°C to +150°C

SYSTEM DESCRIPTION

The XR-2206 is comprised of four functional blocks; a voltage-controlled oscillator (VCO), an analog multiplier and sine-shaper; a unity gain buffer amplifier; and a set of current switches.

The VCO produces an output frequency proportional to an input current, which is set by a resistor from the timing

terminals to ground. With two timing pins, two discrete output frequencies can be independently produced for FSK generation applications by using the FSK input control pin. This input controls the current switches which select one of the timing resistor currents, and routes it to the VCO.

XR-2206

EXAR

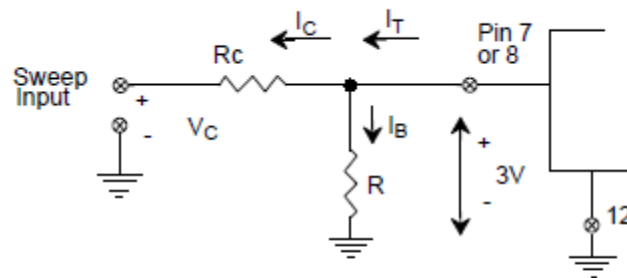


Figure 10. Circuit Connection for Frequency Sweep.

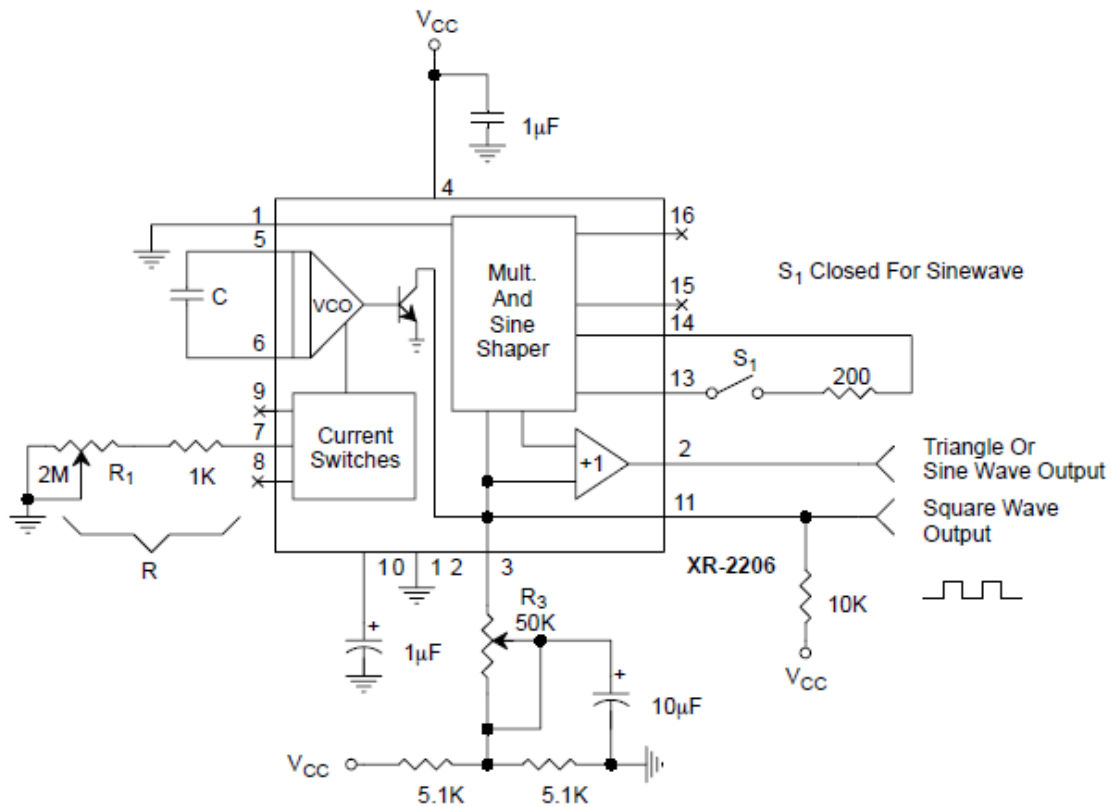


Figure 11. Circuit for Sine Wave Generation without External Adjustment.
(See Figure 3 for Choice of R_3)

With External Adjustment:

The harmonic content of sinusoidal output can be reduced to -0.5% by additional adjustments as shown in *Figure 12*. The potentiometer, R_A , adjusts the sine-shaping resistor, and R_B provides the fine adjustment for the waveform symmetry. The adjustment procedure is as follows:

1. Set R_B at midpoint and adjust R_A for minimum distortion.
2. With R_A set as above, adjust R_B to further reduce distortion.

Triangle Wave Generation

The circuits of *Figure 11* and *Figure 12* can be converted to triangle wave generation, by simply open-circuiting Pin 13 and 14 (i.e., S_1 open). Amplitude of the triangle is approximately twice the sine wave output.

FSK Generation

Figure 13 shows the circuit connection for sinusoidal FSK signal operation. Mark and space frequencies can be independently adjusted by the choice of timing resistors, R_1 and R_2 ; the output is phase-continuous during transitions. The keying signal is applied to Pin 9. The circuit can be converted to split-supply operation by simply replacing ground with V^- .

Pulse and Ramp Generation

Figure 14 shows the circuit for pulse and ramp waveform generation. In this mode of operation, the FSK keying terminal (Pin 9) is shorted to the square-wave output (Pin 11), and the circuit automatically frequency-shift keys itself between two separate frequencies during the positive-going and negative-going output waveforms. The pulse width and duty cycle can be adjusted from 1% to 99% by the choice of R_1 and R_2 . The values of R_1 and R_2 should be in the range of $1k\Omega$ to $2M\Omega$.

PRINCIPLES OF OPERATION**Description of Controls****Frequency of Operation:**

The frequency of oscillation, f_0 , is determined by the external timing capacitor, C , across Pin 5 and 6, and by the timing resistor, R , connected to either Pin 7 or 8. The frequency is given as:

$$f_0 = \frac{1}{RC} \text{ Hz}$$

and can be adjusted by varying either R or C . The recommended values of R , for a given frequency range, as shown in *Figure 5*. Temperature stability is optimum for $4k\Omega < R < 200k\Omega$. Recommended values of C are from $1000pF$ to $100\mu F$.

Frequency Sweep and Modulation:

Frequency of oscillation is proportional to the total timing current, I_T , drawn from Pin 7 or 8:

$$f = \frac{320I_T(mA)}{C(\mu F)} \text{ Hz}$$

Timing terminals (Pin 7 or 8) are low-impedance points, and are internally biased at +3V, with respect to Pin 12. Frequency varies linearly with I_T , over a wide range of current values, from $1\mu A$ to $3mA$. The frequency can be controlled by applying a control voltage, V_C , to the activated timing pin as shown in *Figure 10*. The frequency of oscillation is related to V_C as:

$$f = \frac{1}{RC} \left(1 + \frac{R}{R_c} \left(1 - \frac{V_C}{3} \right) \right) \text{ Hz}$$

where V_C is in volts. The voltage-to-frequency conversion gain, K , is given as:

$$K = \partial f / \partial V_C = - \frac{0.32}{R_c C} \text{ Hz/V}$$

CAUTION: For safety operation of the circuit, I_T should be limited to $\leq 3mA$.

EXPERIMENT # 5

Introduction to Phase Locked Loop

Objective

- Learn the basic principles of Phase-locked loop (PLL)
- Learn to set up practical circuit of PLL
- Measure the characteristics of the PLL (CD4046)

Apparatus

- IC CD4046
- Resistor
- Capacitor
- Oscilloscope
- Function generator

Theory

PLL stands for '*Phase-Locked Loop*' and is basically a closed loop frequency control system, whose functioning is based on the phase sensitive detection of phase difference between the input and output signals of the controlled oscillator. The PLL is a useful building block in communication circuits. There are a wide variety of uses for a PLL including FM and AM detection, phase tracking, frequency synthesis, and frequency multiplication. The purpose of this lab is to look at the PLL characteristics.

Figure 1 shows the classic configuration of a PLL. It consists of three main components: a voltage-controlled oscillator (VCO), a phase comparator, and a loop filter.

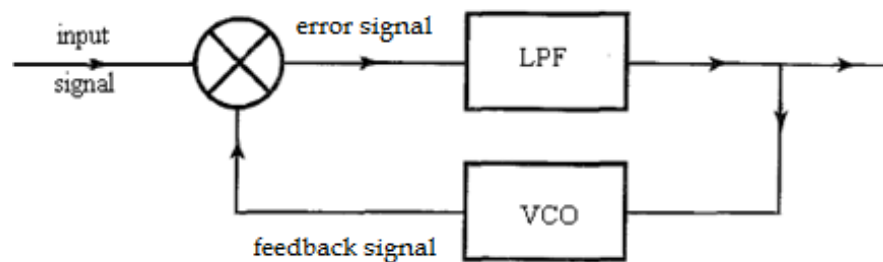


Figure 5.1: PLL block diagram

1. Voltage-controlled oscillator (VCO)

A VCO is an oscillator (a free-running multi-vibrator) with a stable frequency of oscillation that depends on external voltage. When a dc or slow changing ac voltage is applied to the VCO input, the output frequency changes or deviates proportionally.

2. Phase comparator/Phase detector

The phase detector is a device that compares two input frequencies, generating an output that is a measure of their phase difference.

3. Loop filter/Low pass filter

The operation of the PLL is similar to that of a feedback system. When an input signal of frequency f_{IN} is initially applied to the PLL, the phase comparator compares the frequency of the input signal to the frequency of the VCO signal. The phase comparator produces the phase-error signal that is proportional to the difference in frequency between the two signals. The phase-error signal, after being filtered and amplified, causes the VCO frequency to deviate in the direction of f_{IN} . If conditions are right, the VCO will quickly "lock" to f_{IN} maintaining a fixed relationship with the input signal.

The PLL has three operating states.

a) Free Running

When there is no signal or when the feedback loop is open, the VCO operates at a preset frequency called its natural or free-running frequency (f_0).

b) Capture

To be in the capture state there must be an input signal and the feedback loop must be closed. In the capture state the PLL is in the process of acquiring a lock. The frequency range over which the input will cause the loop to lock is called the **acquisition range** or **capture range**. **Pull-in range** is the capture range expressed as a peak value. The lowest frequency the PLL can lock onto is called the lower capture limit (f_{cl}), and the highest frequency the PLL can lock onto is called the upper capture limit (f_{cu}).

$$\text{Capture range of PLL} \quad 2f_c = f_{cu} - f_{cl}$$

c) Lock

In the lock state, the VCO output frequency is locked onto the (equal to) the frequency of the input signal. In the lock state, the VCO output frequency tracks (follows) changes in the frequency of the external input signal. A PLL can track the incoming frequency only over a finite range of frequency shift and it is called **tracking range** or **lock range**. **Hold-in range** is the lock range expressed as a peak value. The lowest frequency a PLL will track is called the lower lock limit (f_{ll}), and the highest frequency that a PLL will track is called the upper lock limit (f_{lu}).

$$\text{Lock range of PLL} \quad 2f_L = f_{lu} - f_{ll}$$

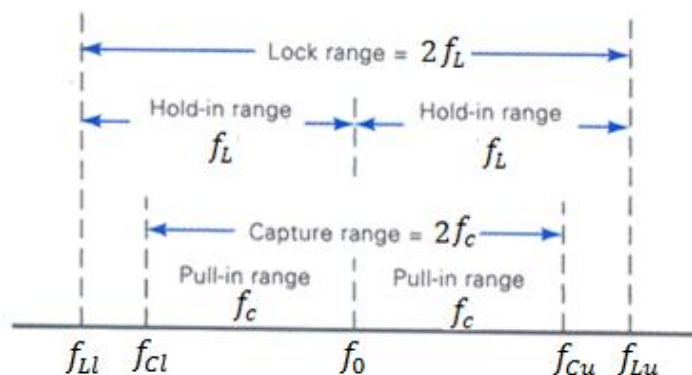


Figure 5.2: PLL Capture and Lock ranges

The first part of the experiment focuses on the Voltage Controlled Oscillator and measurements will be made of frequency vs. voltage characteristics of the VCO. The second part entails building a PLL with the VCO

from part one. Here, lock and capture ranges will be measured. We will be using CD4046 chip for VCO and PLL implementation. Data sheet is provided along with the manual.

Voltage controlled Oscillator (VCO)

In order to design a VCO with specific characteristics, Figure 5-7 of the data sheet provide the necessary component information. For example figure 5 shows how the center frequency varies with the capacitance C1, resistance R1, and supply voltage VDD. Figure 6 & 7 gives information about the minimum and maximum frequency respectively.

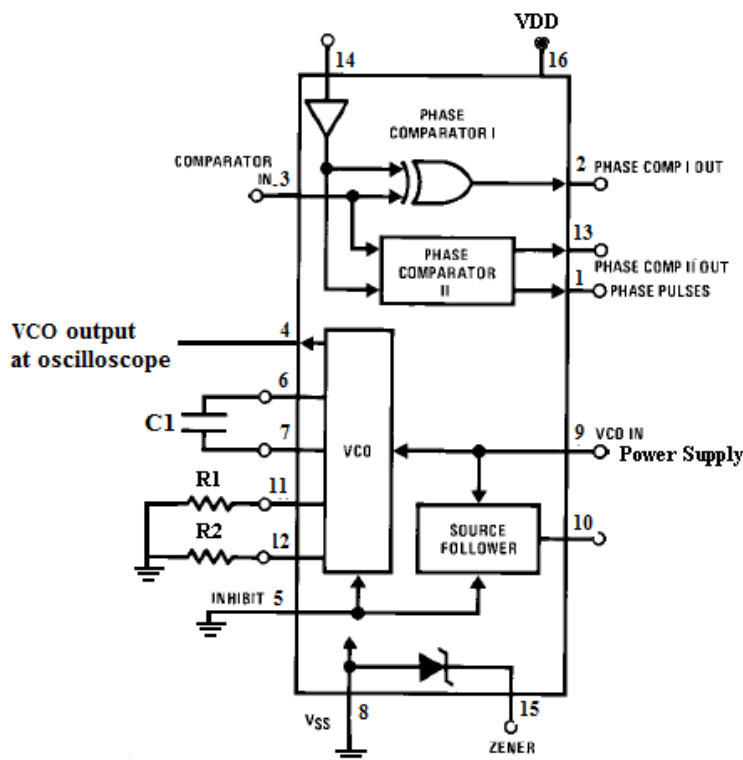


Figure 5.3: Circuit for VCO using CD4046

Procedure

1. Using the following component values and Figure 3 assemble the VCO on bread board.

$$VDD = 15V$$

$$R1 = 10\ k\Omega$$

$$R2 = 100\ k\Omega$$

$$C1 = 0.01\ \mu F$$

2. The INHIBIT pin (Pin 5) must be grounded, and it is suitable for VSS (Pin 8) to be ground as well.
3. Make sure that the supply voltages are turned low and the power supply is initially off.
4. When this setup is complete, turn the power supply on.
5. Observe the output at Pin 4 on oscilloscope.
6. Increase the voltage on Pin 9, VCO in, up to 15V in steps. Record the frequency of the output waveform at each step in Table 1.

Observations**Table 5.1:** VCO Data (Frequency vs. voltage)

VCO_IN (V) Pin 9	VCO_OUT Frequency (kHz) Pin 4
0.0	
1.5	
3	
4.5	
6	
7.5	
9	
10.5	
12	
13.5	
15	

Frequency at $V_{DD}/2$ = center frequency f_0 = _____

Phase Locked Loop (PLL)

To configure the CD4046 chip as a PLL we will use the VCO from the previous part and add the external circuitry for loop filter. In this part we will observe the functioning of PLL and will measure the lock and capture range of PLL. The lock range of a PLL is set by the VCO where as the capture range is set by the loop filter. The capture range can be equal to the lock range or smaller, but never larger. In this experiment the loop components are chosen to make the capture range equal to the lock range.

Procedure

1. Set up the circuit as shown in figure 4 with following loop filter components.

$$C2 = 0.01\mu F$$

$$R3 = 10\text{ k}\Omega$$

2. Select square wave from function generator and set the frequency close to center frequency of VCO. Apply square wave signal at Pin 14 through the capacitor. Observe the applied signal on channel 1, and the VCO (or PLL) output on channel 2 of oscilloscope. PLL should show a stable waveform and have the same frequency as the input square wave.
3. To measure the lock range start from the center frequency and slowly decrease the input frequency until the signals are just unlocking, this is the lower lock limit f_{LL} .
4. Then increase the frequency until the signals start to lock again; this is the lower capture limit f_{CL} . Note that these frequencies will be pretty close to each other since the loop filter was designed to give a capture range equal to the lock range.
5. Continue increasing the frequency until the signals are unlocked again this is the upper lock limit f_{LU} .

6. Lastly, decrease the frequency from this point until the signals are locked again; this is the upper capture limit f_{Cu} .

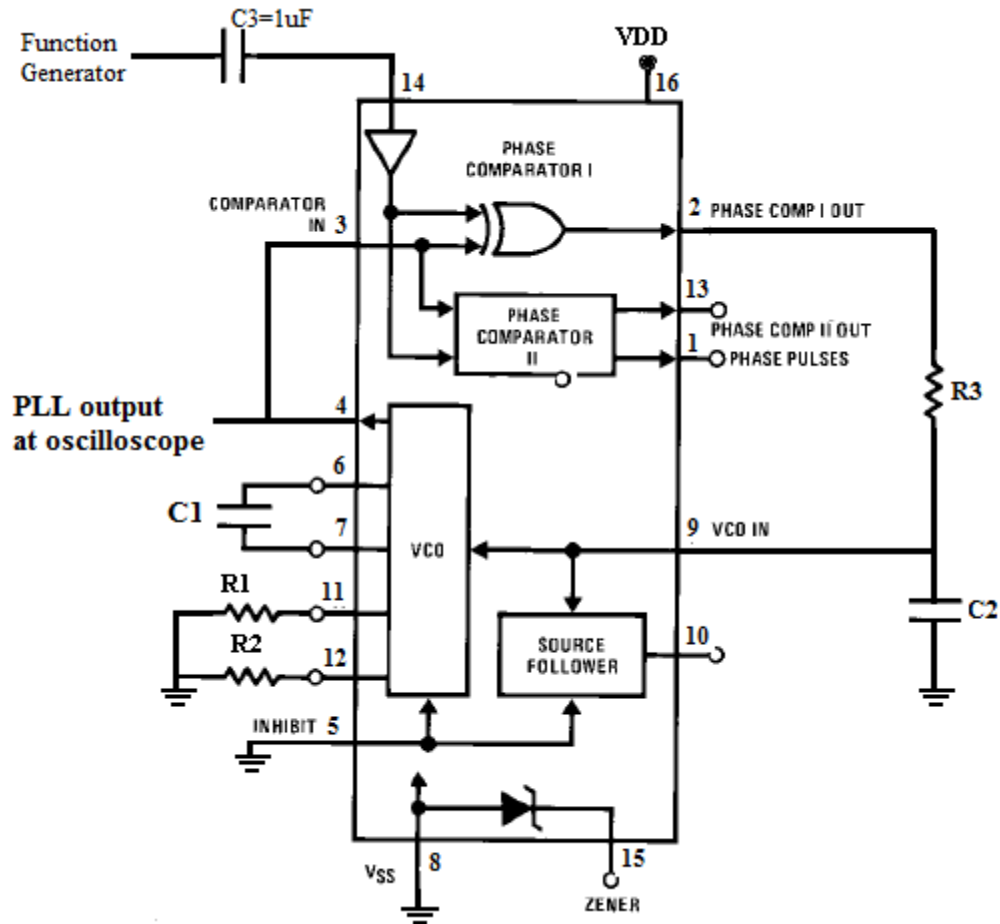


Figure 5.4: Circuit for PLL using CD4046

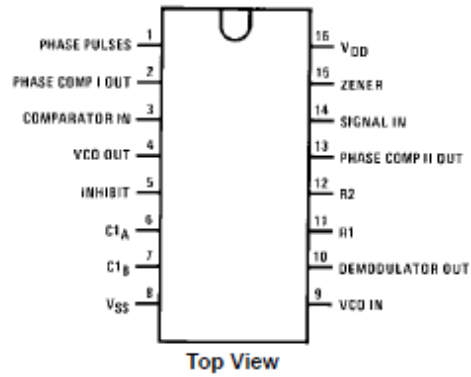
Observations

Table 5.2: PLL Characteristics

Upper lock limit f_{Lu}	
Lower lock limit f_{Ll}	
Upper capture limit f_{Cu}	
Lower capture limit f_{Cl}	
Lock range f_L	
Capture range f_c	

CD4046BC

Connection Diagram



Block Diagram

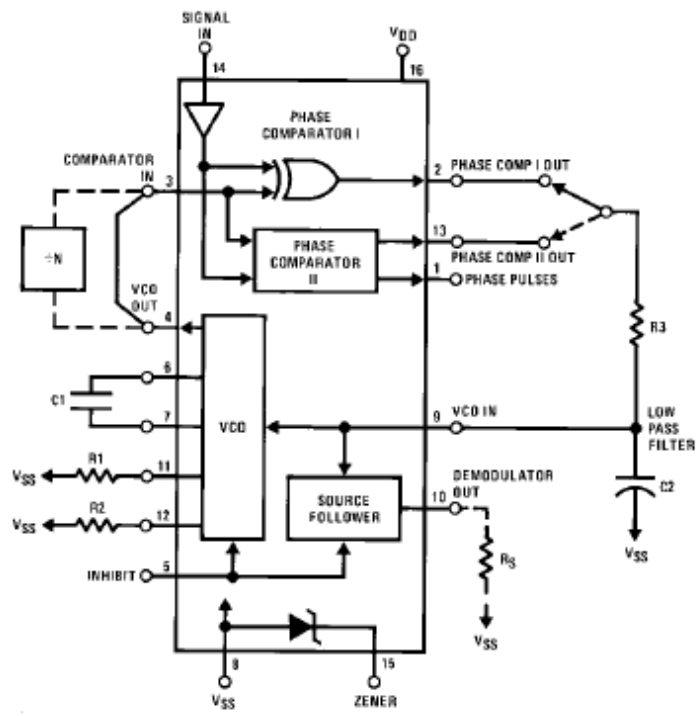


FIGURE 1.

Absolute Maximum Ratings(Note 1)

(Note 2)

DC Supply Voltage (V_{DD})	-0.5 to +18 V_{DC}
Input Voltage (V_{IN})	-0.5 to V_{DD} +0.5 V_{DC}
Storage Temperature Range (T_S)	-65°C to +150°C
Power Dissipation (P_D)	
Dual-In-Line	700 mW
Small Outline	500 mW
Lead Temperature (T_L)	
(Soldering, 10 seconds)	260°C

Recommended Operating Conditions (Note 2)

DC Supply Voltage (V_{DD})	3 to 15 V_{DC}
Input Voltage (V_{IN})	0 to V_{DD} V_{DC}
Operating Temperature Range (T_A)	-40°C to +85°C

Note 1: "Absolute Maximum Ratings" are those values beyond which the safety of the device cannot be guaranteed. They are not meant to imply that the devices should be operated at these limits. The table of "Recommended Operating Conditions" and "Electrical Characteristics" provides conditions for actual device operation.

Note 2: $V_{SS} = 0V$ unless otherwise specified.

DC Electrical Characteristics (Note 2)

Symbol	Parameter	Conditions	-40°C		+25°C			+85°C		Units
			Min	Max	Min	Typ	Max	Min	Max	
I_{DD}	Quiescent Device Current	Pin 5 = V_{DD} , Pin 14 = V_{DD} , Pin 3, 9 = V_{SS} $V_{DD} = 5V$		20		0.005	20		150	μA
		$V_{DD} = 10V$		40		0.01	40		300	μA
		$V_{DD} = 15V$		80		0.015	80		600	μA
		Pin 5 = V_{DD} , Pin 14 = Open, Pin 3, 9 = V_{SS} $V_{DD} = 5V$		70		5	55		205	μA
		$V_{DD} = 10V$		530		20	410		710	μA
		$V_{DD} = 15V$		1500		50	1200		1800	μA
V_{OL}	LOW Level Output Voltage	$V_{DD} = 5V$		0.05		0	0.05		0.05	V
		$V_{DD} = 10V$		0.05		0	0.05		0.05	V
		$V_{DD} = 15V$		0.05		0	0.05		0.05	V
V_{OH}	HIGH Level Output Voltage	$V_{DD} = 5V$	4.95		4.95	5		4.95		V
		$V_{DD} = 10V$	9.95		9.95	10		9.95		V
		$V_{DD} = 15V$	14.95		14.95	15		14.95		V
V_{IL}	LOW Level Input Voltage Comparator and Signal In	$V_{DD} = 5V, V_O = 0.5V$ or $4.5V$		1.5		2.25	1.5		1.5	V
		$V_{DD} = 10V, V_O = 1V$ or $9V$		3.0		4.5	3.0		3.0	V
		$V_{DD} = 15V, V_O = 1.5V$ or $13.5V$		4.0		6.25	4.0		4.0	V
V_{IH}	HIGH Level Input Voltage Comparator and Signal In	$V_{DD} = 5V, V_O = 0.5V$ or $4.5V$	3.5		3.5	2.75		3.5		V
		$V_{DD} = 10V, V_O = 1V$ or $9V$	7.0		7.0	5.5		7.0		V
		$V_{DD} = 15V, V_O = 1.5V$ or $13.5V$	11.0		11.0	8.25		11.0		V
I_{OL}	LOW Level Output Current (Note 4)	$V_{DD} = 5V, V_O = 0.4V$	0.52		0.44	0.88		0.36		mA
		$V_{DD} = 10V, V_O = 0.5V$	1.3		1.1	2.25		0.9		mA
		$V_{DD} = 15V, V_O = 1.5V$	3.6		3.0	8.8		2.4		mA
I_{OH}	HIGH Level Output Current (Note 4)	$V_{DD} = 5V, V_O = 4.6V$	-0.52		-0.44	-0.88		-0.36		mA
		$V_{DD} = 10V, V_O = 9.5V$	-1.3		-1.1	-2.25		-0.9		mA
		$V_{DD} = 15V, V_O = 13.5V$	-3.6		-3.0	-8.8		-2.4		mA
I_{IN}	Input Current	All Inputs Except Signal Input $V_{DD} = 15V, V_{IN} = 0V$		-0.3		-10^{-5}	-0.3		-1.0	μA
		$V_{DD} = 15V, V_{IN} = 15V$		0.3		10^{-5}	0.3		1.0	μA
C_{IN}	Input Capacitance	Any Input (Note 3)					7.5			pF
P_T	Total Power Dissipation	$f_o = 10$ kHz, $R1 = 1$ M Ω , $R2 = \infty$, $CXO_{IN} = 5$ nA/2								
		$V_{DD} = 5V$				0.07				mW
		$V_{DD} = 10V$				0.6				mW
		$V_{DD} = 15V$				2.4				mW

Note 3: Capacitance is guaranteed by periodic testing.

Note 4: I_{OH} and I_{OL} are tested one output at a time.

CD4046BC

AC Electrical Characteristics (Note 5) $T_A = 25^\circ\text{C}$, $C_L = 50\text{ pF}$

Symbol	Parameter	Conditions	Min	Typ	Max	Units	
VCO SECTION							
I _{DD}	Operating Current	f _o = 10 kHz, R1 = 1 MΩ, R2 = ∞, φXO _{IN} = ζ _{ΔΔ} /2 V _{DD} = 5V V _{DD} = 10V V _{DD} = 15V		20 90 200		μA μA μA	
f _{MAX}	Maximum Operating Frequency	C1 = 50 pF, R1 = 10 kΩ, R2 = ∞, φXO _{IN} = ζ _{ΔΔ} V _{DD} = 5V V _{DD} = 10V V _{DD} = 15V	0.4 0.6 1.0	0.8 1.2 1.6		MHz MHz MHz	
	Linearity	VCO _N = 2.5V ±0.3V, R1 ≥ 10 kΩ, V _{DD} = 5V VCO _N = 5V ±2.5V, R1 ≥ 400 kΩ, V _{DD} = 10V VCO _N = 7.5V ±5V, R1 ≥ 1 MΩ, V _{DD} = 15V		1 1 1		% % %	
		Temperature-Frequency Stability No Frequency Offset, f _{MIN} = 0	%/°C ∝ 1/φ, ζ _{ΔΔ} R2 = ∞ V _{DD} = 5V V _{DD} = 10V V _{DD} = 15V		0.12–0.24 0.04–0.08 0.015–0.03		%/°C %/°C %/°C
		Frequency Offset, f _{MIN} ≠ 0	V _{DD} = 5V V _{DD} = 10V V _{DD} = 15V		0.06–0.12 0.05–0.1 0.03–0.06		%/°C %/°C %/°C
	VCO _{IN}	Input Resistance	V _{DD} = 5V V _{DD} = 10V V _{DD} = 15V		10 ⁸ 10 ⁸ 10 ⁸		MΩ MΩ MΩ
VCO	Output Duty Cycle	V _{DD} = 5V V _{DD} = 10V V _{DD} = 15V		50 50 50		% % %	
t _{THL}	VCO Output Transition Time	V _{DD} = 5V		90	200	ns	
t _{THL}		V _{DD} = 10V		50	100	ns	
		V _{DD} = 15V		45	80	ns	
PHASE COMPARATORS SECTION							
R _{IN}	Input Resistance Signal Input	V _{DD} = 5V	1	3		MΩ	
		V _{DD} = 10V	0.2	0.7		MΩ	
		V _{DD} = 15V	0.1	0.3		MΩ	
	Comparator Input	V _{DD} = 5V		10 ⁸		MΩ	
		V _{DD} = 10V		10 ⁸		MΩ	
		V _{DD} = 15V		10 ⁸		MΩ	
	AC-Coupled Signal Input Voltage Sensitivity	C _{SERIES} = 1000 pF f = 50 kHz V _{DD} = 5V		200	400	mV	
		V _{DD} = 10V		400	800	mV	
		V _{DD} = 15V		700	1400	mV	

AC Electrical Characteristics (Continued)						
Symbol	Parameter	Conditions	Min	Typ	Max	Units
DEMODULATOR OUTPUT						
$V_{CO_{IN}}-V_{DEM}$	Offset Voltage	$R_S \geq 10\text{ k}\Omega$, $V_{DD} = 5V$		1.50	2.2	V
		$R_S \geq 10\text{ k}\Omega$, $V_{DD} = 10V$		1.50	2.2	V
		$R_S \geq 50\text{ k}\Omega$, $V_{DD} = 15V$		1.50	2.2	V
	Linearity	$R_S \geq 50\text{ k}\Omega$				
		$V_{CO_{IN}} = 2.5V \pm 0.3V$, $V_{DD} = 5V$		0.1		%
		$V_{CO_{IN}} = 5V \pm 2.5V$, $V_{DD} = 10V$		0.6		%
		$V_{CO_{IN}} = 7.5V \pm 5V$, $V_{DD} = 15V$		0.8		%
ZENER DIODE						
V_Z	Zener Diode Voltage	$I_Z = 50\text{ }\mu A$	6.3	7.0	7.7	V
R_Z	Zener Dynamic Resistance	$I_Z = 1\text{ mA}$		100		Ω

Note 5: AC Parameters are guaranteed by DC correlated testing.

Phase Comparator State Diagrams

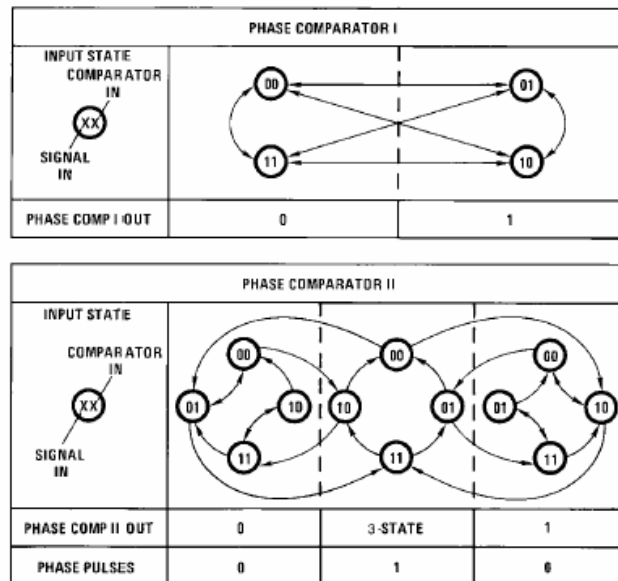


FIGURE 2.

Typical Performance Characteristics

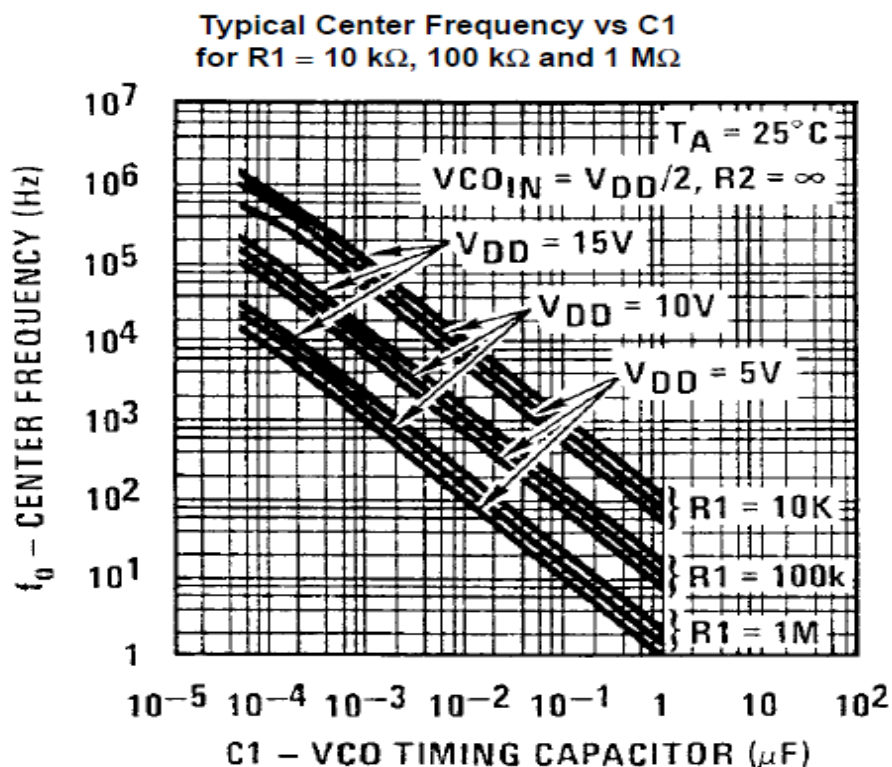


FIGURE 5.

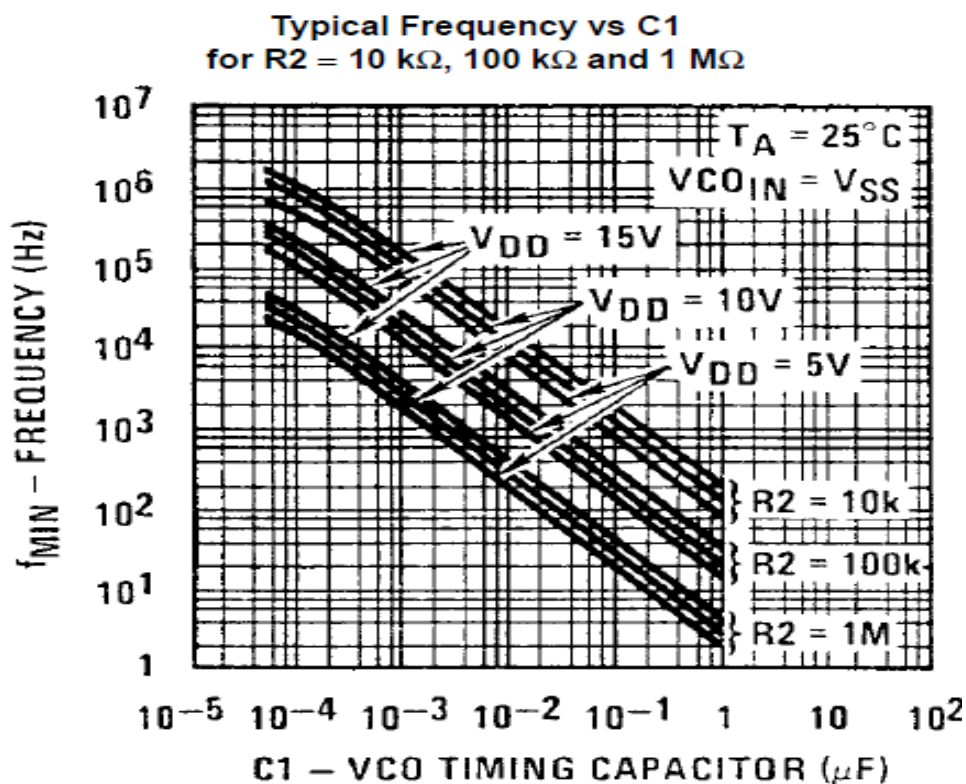


FIGURE 6.

Typical Performance Characteristics (Continued)

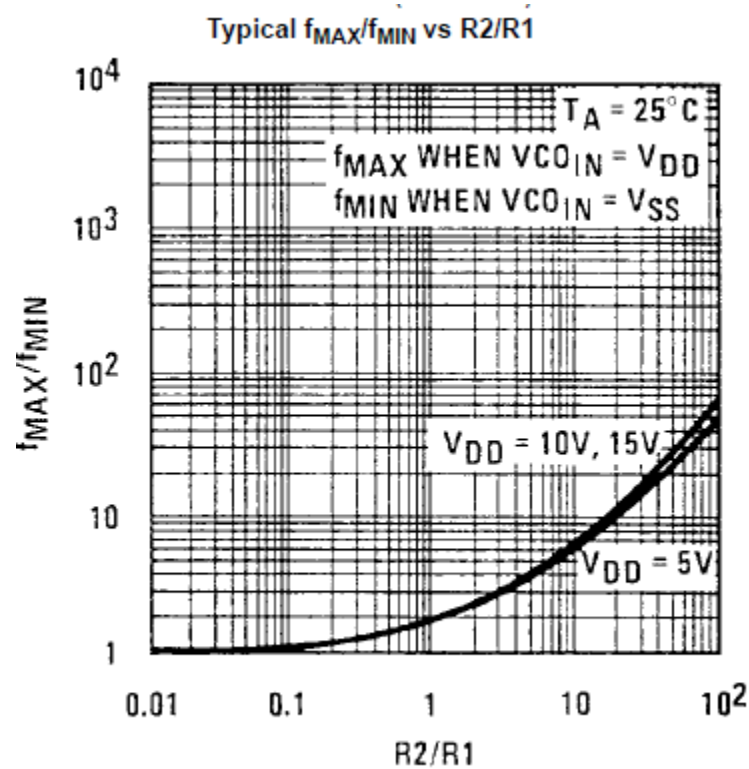


FIGURE 7.

POST LAB

A PLL has a VCO with a free running frequency of 12MHz. As the frequency of the reference input signal is gradually raised from zero, loop locks at 10MHz and comes out of lock again at 16MHz. Find the capture range and lock range.

1. What is the function of Pin 5 in CD4046?
2. Explain the role of the resistance marked R2 in CD4046 circuit with reference to the experiment performed.

EXPERIMENT # 6

FM Demodulator

Objective

- To understand the demodulation of an FM signal using PLL

Apparatus

- IC LM565
- Resistors
- Capacitors
- DC power supply
- Function Generator
- Oscilloscope
- Connecting probes and cables

Theory

Frequency demodulator, also called frequency discriminator, is a circuit, which converts instantaneous frequency variations to linear voltage changes. There are many types of circuit used in communication system as FM to AM conversion, balanced, and phase discriminators and phase-locked loop (PLL) frequency demodulators.

1) Slope Detection

An operational amplifier differentiator followed by an envelope detector can serve the purpose of FM demodulator.

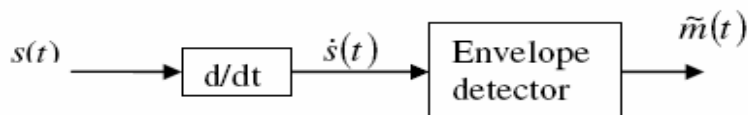


Figure 6.1. Slope Detection

A simple tuned circuit followed by an envelope detector can serve the purpose of demodulator because its frequency response below or above the tuned frequency is approximately linear this method of demodulation is known as slope detection. However the slope of $|H(\omega)|$ for such a demodulator is linear only for a small band this problem can be removed by using Balanced discriminator or ratio detectors.

2) Zero-crossing detector

These are the frequency counters designed to measure the instantaneous frequency by counting the number of zero crossings. The rate of zero crossings is equal to the instantaneous frequency of the signal.

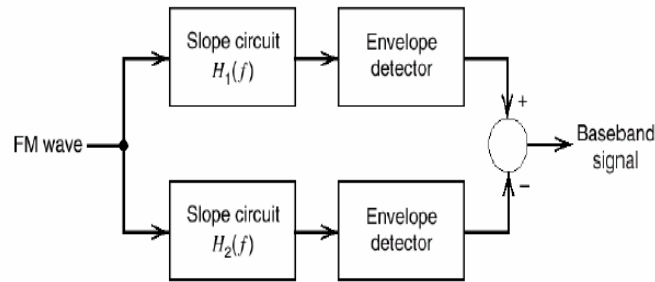


Figure 6.2. Zero Crossing Detector

3) Phase-locked loop

A basic phase locked loop is a simple control loop, which locks a VCO (voltage controlled oscillator) to some reference frequency. The VCO in an oscillator with output frequency proportional to input control voltage. We consider here only the process by which the loop stays “in lock” and the VCO tracks the phase/frequency of the reference input. Therefore, the VCO adjusts itself so that the error signal $e(t)$ tends to zero. When $e(t)$ is close to zero, $r(t) \simeq s(t)$ and $v(t) \simeq m(t)$ (the message signal).

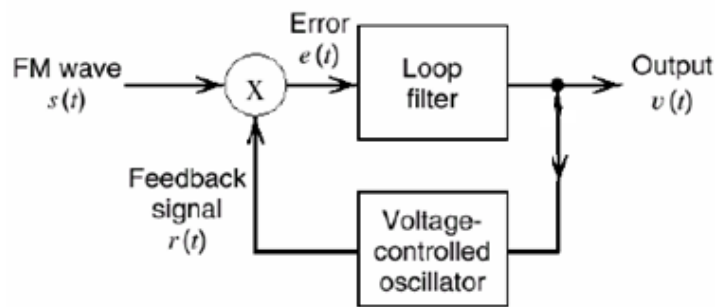


Figure 6.3. Phase Locked Loop

In this experiment we will introduce the operations of PLL frequency demodulator using LM565. The PLL circuit of Figure 1 can be used as a frequency demodulator.

Procedure

1. Implement the circuit in Figure 4 on bread board.
2. Calculate the free running frequency using following formula

$$f_0 = \frac{0.3}{R_1 C_1} = \underline{\hspace{2cm}}$$

3. Plug in the FM signal generated in previous experiment on Pin 2.
4. Observe the output at Pin 7.

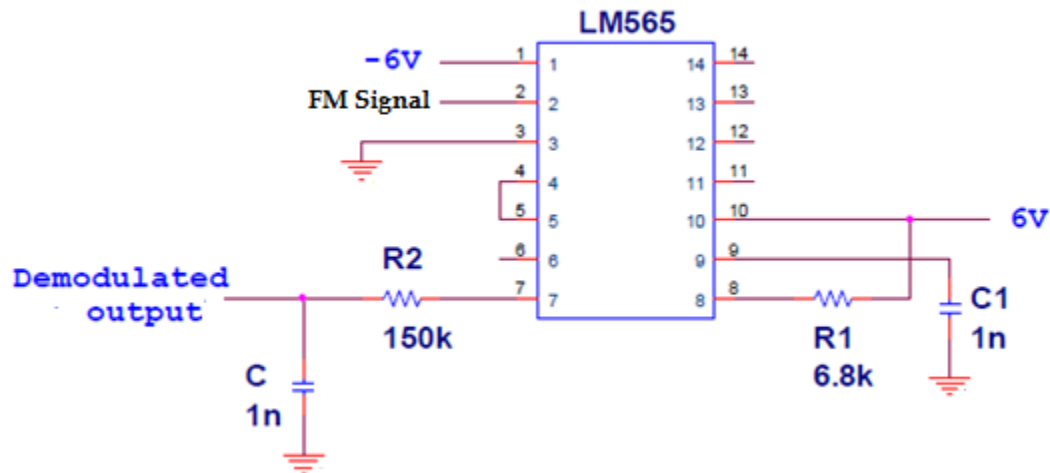


Figure 6.4: FM demodulator circuit

Graphical Analysis

Sketch the frequency modulated and frequency demodulated waveforms as seen on the oscilloscope. Also mention the time/div and volts/div for each channel.

POST LAB

1. What effect does changing the amplitude of the modulation signal have on the demodulated output?
2. Referring to Figure 4. How would you change the VCO free-running frequency from 20 to 50 KHz?
3. Why VCO control voltage used as the demodulated output?

Experiment # 7

Digital Modulation: FSK

Objective

- Learn the basic concept of frequency shift keying (FSK)
- Learn to implement FSK using XR2206

Apparatus

- XR2206
- Oscilloscope
- Function Generator
- Power supply
- Resistors
- Capacitors

Theory

FSK modulation requires the swapping of frequency from one level to another. A “0” is transmitted by a pulse of frequency f_1 and “1” is transmitted by using a pulse of frequency f_2 as show in figure 1. Hence the binary information is contained in the frequency of the carrier wave.

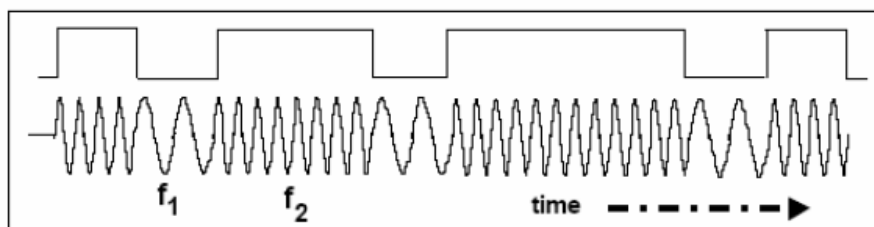


Figure 7.1: FSK waveform

Generation \ Modulation of FSK Signal

FSK signal can be generated by using two oscillators tuned at two different frequencies but connected to single output terminal. ‘0’ can be transmitted by selecting the output of one of the oscillators while ‘1’ can be transmitted by selecting the other oscillator. The selection between the two oscillators will be determined by the message signal.

We will be using XR-2206 for Frequency Shift Keying. Data sheet is provided along with the manual. The XR-2206 can be operated with two separate timing resistors R_1 and R_2 , connected to timing Pins 7 and 8 respectively, as shown in Figure 2. Depending on the polarity of the logic signal at Pin 9, either one or the other of these timing resistors is activated. If Pin 9 is open-circuited or connected to a bias voltage $\geq 2V$, only R_1 is activated. Similarly, if the voltage level at Pin 9 is $\leq 1V$, only R_2 is activated. Thus, the output frequency can be keyed between two levels, f_1 and f_2 , as

$$f_1 = \frac{1}{R_1 C} \text{ and } f_2 = \frac{1}{R_2 C}$$

Procedure

1. Design an FSK modulator to have $f_1 = 50,000$ Hz and $f_2 = 10,000$ Hz. Use any values of resistors or capacitors. But remember to look at data sheet very carefully.

Table 7.1

Frequencies	Capacitor	Timing Resistors
$f_1 = 50kHz$	C =	$R_1 =$
$f_2 = 10kHz$		$R_2 =$

2. Set up the circuit as show in figure 2.
3. Generate a digital signal having frequency 2 KHz and amplitude 5Vp-p with dc bias from function generator and apply it on Pin 9.
4. Observe the output at Pin 2 on oscilloscope.

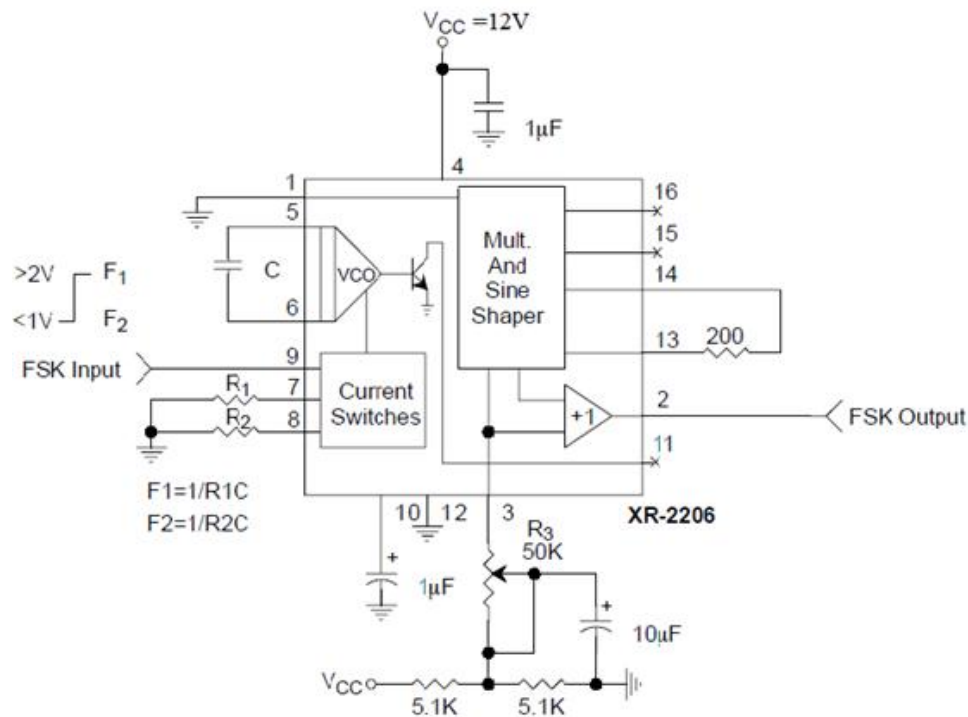


Figure 7.2: Sinusoidal FSK Generator

Graphical Analysis

Sketch the message signal and FSK modulated signal as seen on the oscilloscope. Also mention the time/div and volts/div for each channel.

POST LAB

1. What will happen if we remove the resistor between 13 and 14 pin of XR-2206 IC?
2. What does variable resistor do in the given figure 2.

Experiment # 8

Pre-Emphasis & De-Emphasis

Objective

- Learn how the characteristics of pre-emphasis and De-emphasis differ from each other

Apparatus

- Oscilloscope
- Function Generator
- Resistors
- Capacitors

Theory

The noise has an effect on the higher modulating frequencies than on the lower ones. Thus, if the higher frequencies were artificially boosted at the transmitter and correspondingly cut at the receiver, an improvement in noise immunity could be expected, thereby increasing the SNR ratio. This boosting of the higher modulating frequencies at the transmitter is known as pre-emphasis and the compensation at the receiver is called de-emphasis.

Procedure

Pre-emphasis

1. Connect the circuit as per the circuit diagram:

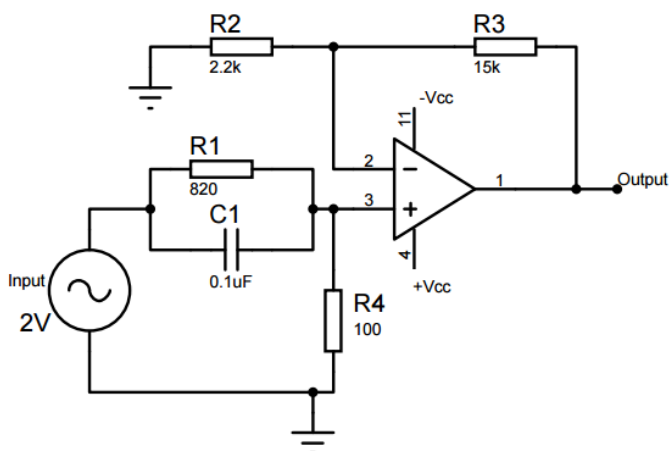


Figure 8.1: Pre-emphasis Circuit

2. Apply the sinusoidal signal of amplitude 2V as input signal to pre-emphasis circuit.
3. Vary the frequency in steps and note down the output voltage, V_o in Table 1.
4. Calculate the gain in dB.

Table 8.1: Pre-emphasis

Frequency (Hz)	Output voltage V_o	Gain in dB= $20\log(V_o/V_i)$
1 k		
3 k		
5 k		
7 k		
10 k		
13 k		
15 k		
17 k		
20 k		

Graphical Analysis

Plot the graph between gain in dB vs. frequency of pre-emphasis network.

De-emphasis

1. Connect the circuit as per the circuit diagram

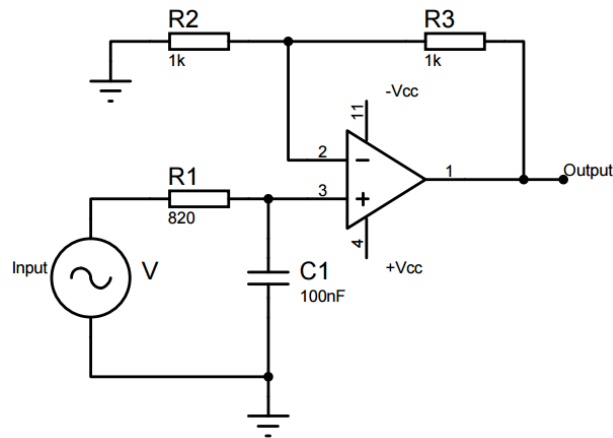


Figure 8.2: De-emphasis Circuit

2. Apply the sinusoidal signal of amplitude 2V as input signal to de-emphasis circuit.
3. Vary the frequency in steps and note down the output voltage, V_o .
4. Calculate the gain in dB.

Table 8.2: De-emphasis

Frequency (Hz)	Output voltage V_o	Gain in dB= $20\log(V_o/V_i)$
1 k		
3 k		
5 k		
7 k		
10 k		
13 k		
15 k		
17 k		
20 k		

Graphical Analysis

Plot the graph between gain (in dB) Vs. frequency of de-emphasis network.

POST LAB

1. What should be the time constant for the de-emphasis circuit?
2. Why pre-emphasis is done after modulation?
3. What is pre-emphasis? How is it used to improve the S/N of FM transmission?

Experiment # 9

Pulse Code Modulation

Objective

- Familiarize students with the contents of the experiment and give them hands on experience regarding the experiment
- To generate the pulse code modulated and demodulated signals

Apparatus

- Communication trainer DEV-2786
- Oscilloscope
- Probes
- Wires

Theory

Pulse code modulation is basically the conversion of an analog signal to digital signal. A signal in its Analog form can take on infinite number of values however it can attain only limited values in its digital form. An analog signal can be converted to digital signal by means of sampling and quantization, which is rounding off the sampled value to one of its closest number called quantization Levels.

The stream of pulses and non-pulse streams of 1's and 0's are not easily affected by interference and noise. Even in the presence of noise, the presence or absence of a pulse can be easily determined. Since PCM is digital, a more general reason would be that digital signals are easy to process by cheap standard techniques. This makes it easier to implement complicated communication systems such as telephone networks (covered later in this course).

The practical implementation of PCM makes use of other processes. The processes are carried out in the order in which they appear below:

- Filtering
- Sampling
- Quantizing
- Encoding

The filtering stage removes frequencies above the highest signal frequency. These frequencies if not removed, may cause problems when the signal is going through the stage of sampling. Generally, while modulating a signal for PCM a simple A/D converter can be used.

Procedure

1. Generate a sinusoidal signal from the trainer and connect it to the analogue input part of the "PAM/PCM" module on the trainer.
2. Observe the output of the "analogue Output" block.
3. Now Connect a square wave to the clock section of the module and move the selector switch to "ext".

4. Observe the changes in the output by varying
 - Amplitude of input signal
 - Amplitude of clock signal
 - Frequency of the input signal
 - Frequency of the Clock signal
 - The Offset potentiometer on the module

POST LAB

1. Draw the block diagram of PCM modulation system?
2. Differentiate between 6, 7 and 8-bit PCM system?
3. If we change the **interrupt** to the EXT position and increase the frequency of the carrier (which is acting as the sampling frequency), what will be the effect on the output of DAC?

EXPERIMENT # 10

MATLAB Basics for Communication System Design

Objective

- To understand the use of MATLAB for solving communication engineering problems.
- Learn the basics of MATLAB as used in Analogue Communication.
- To develop understanding of MATLAB environment, commands and syntax.

MATLAB

MATLAB is a powerful tool that is utilized by the engineers and others professionals in development and testing of various projects. It is versatile software, with the help of which you can solve and develop any sort of engineering problem. The name MATLAB stands for MATRIX LABORAORY. All the work done in MATLAB is basically in the form of matrices. Scalars are referred as 1-to-1 matrix and vectors are matrices having more than 1 row and column. MATLAB is programmable and have the same logical, relational, conditional and loop structures as in other programming languages, such as C, Java etc. It's very easy to use MATLAB, all we need is to practice it and become a friend of it.

Summary:

- **Scalars**
- **Vectors**
- **Matrices**
- **Plotting**
- **m-files**
- **functions**

Getting Started:

- a) Go to the start button, then programs, MATLAB and then start MATLAB. It is preferred that you have MATLAB7. You can then start MATLAB by double clicking on its icon on Desktop, if there is any.

- b) The Prompt:

>>

The operator shows above is the prompt in MATLAB. MATLAB is interactive language like C, Java etc. We can write the commands over here.

- c) In MATLAB we can see our previous commands and instructions by pressing the up key. Press the key once to see the previous entry, twice to see the entry before that and so on. We can also edit the text by using forward and back-word keys.

Help in MATLAB

In order to use the built-in help of the MATLAB we use the **help** keyword. Write it on the prompt and see the output.

```
>> help sin
```

Also try

```
>> lookfor sin
```

Scalars

A scalar is a single number. A scalar is stored in the MATLAB as a 1 x 1 matrix. Try these on the prompt.

```
>> A = 2;
```

```
>> B = 3;
```

```
>> C = A^B
```

```
>> C = A*B
```

Try these instructions as well

```
>> C = A+B
```

```
>> C = A-B
```

```
>> C = A/B
```

```
>> C = A\B
```

Note the difference between last two instructions.

Try to implement these two relations and show the result in the provided space

a) $25 (3^{1/3}) + 2 (2+9^2) =$ _____

b) $5x^3 + 3x^2 + 5x + 14$ for $x = 3$ is _____

c) Solve this quadratic equation using quadratic formula.

$$a = 2.5, b = 5, c = -6$$

$$x = \text{_____ and } \text{_____}$$

Vectors

Vectors are also called arrays in MATLAB. Vectors are declared in the following format.

```
>> X = [1 2 3 4]
```

```
>> Y = [2 5 8 9]
```

Try these two instructions in MATLAB and see the results.

>> length (X) = _____

>> size (X) = _____

What is the difference between these two?

Try these instructions and see the results.

>> X.*Y = _____

>> X.^Y = _____

>> X+Y = _____

>> X-Y = _____

>> X./Y = _____

>> X' = _____

Also try some new instructions for this like and notice the outputs in each case.

>> ones (1,4)

>> ones (2,4)

>> ones (4,1)

>> zeros (1,4)

>> zeros (2,4)

There is an important operator, the colon operator (:), it is very important operator and frequently used during these labs. Try this one.

>> X = [0:0.1:1]

Notice the result. And now type this

>> length (X)

>> size (X)

What did the first and second number represent in the output of last instruction?

Now try this one.

```
>> A = [ones(1,3), [2:2:10], zeros(1,3)]
```

What is the length and size of this?

Length = _____

Size = _____

Try '*help ones*' and '*help zeros*' as well, and note down its important features.

MATRICES

Try this and see the output.

```
>> A = [1 2 3;4 5 6;7 8 9]
```

```
>> B = [1,2,3;4,5,6;7,8,9]
```

Is there any difference between the two? Try to implement 2-to-3 matrix and 3-to-2 matrix.

Also take help on **mod**, **rem**, **det**, **inv** and **eye** and try to implement them. Try to use **length** and **size** commands with these matrices as well and see the results.

Try to solve these.

1. $6x + 12y + 4z = 70$

$$7x - 2y + 3z = 5$$

$$2x + 8y - 9z = 64$$

2. $A = \begin{bmatrix} 2 & 3 & 4 & 5 \\ 1 & 8 & 9 & 0 \\ 2 & 3 & 1 & 3 \\ 5 & 8 & 9 & 3 \end{bmatrix}$

$$\text{Solve } 6A - 2I + A^2 =$$

PLOTTING

Plotting is very important as we have to deal with various type of waves and we have to view them as well.

Try these and have a look on the results.

```
>> x = [0:0.1:10];
```

```
>> y = sin (x);
```

```
>> z = cos (x);
```

```
>> subplot (3,1,1);
```

```
>> plot (x,y);
```

```
>> grid on;
```

```
>> subplot (3,1,2);
```

```
>> plot (x,z);
```

```
>> grid on; hold on;
```

```
>> subplot (3,1,3);
```

```
>> stem (x,z);  
>> grid on;  
>> hold on;  
>> subplot (3,1,3);  
>> stem (x,y, 'r');
```

Take **help** on the functions and commands that you don't know. See the difference between the **stem** and **plot**.

See help on plot, figure, grid, hold, subplot, stem and other features of it.

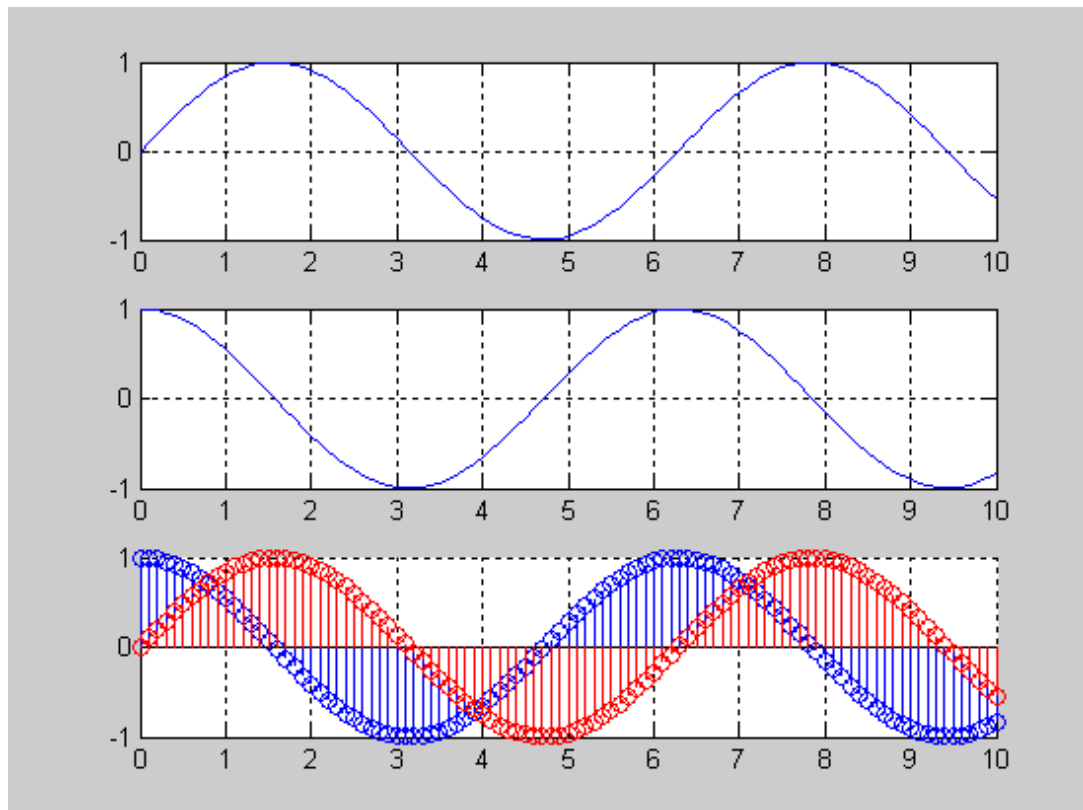


Figure 10.1

M-FILES

MATLAB can execute a sequence of statements stored in disk files. Such files are called M-files because they must have the file type `‘.m’`. Lot of our work will be done with creation of m-files.

There are two types of m-files: Script and function files.

Script Files

We can use script files in order to write long programs such as one on the previous page. A script file may contain any command that can be entered on the prompt. Script files can have any name but they should be saved with `“.m”` extension. In order to excuse an m-file from the prompt, just type its name on the prompt. You can make an m-file by typing **edit** on the prompt or by clicking on the file then new and m-file. See an example of m-file. Write it and see the results.

```
% This is comment
```

```
% A comment begins with a percent symbol
```

```
% The text written in the comments is ignored by the MATLAB
```

```
% comments in your m-files.
```

```
clear;
clc;
x = [0:0.1:10];
y = sin (x);
subplot (2,2,1);
plot (x,y, 'r');
grid on;
z = cos (x);
subplot (2,2,2);
plot (x,z);
grid on;
w = 90;
yy = 2*pi*sin (x+w)
subplot (2,2,3);
plot (x,yy);
grid on;
zz = sin (x+2*w);
subplot (2,2,4);
stem (x,zz, 'g');
hold on;
stem (x,y, 'r');
grid on;
```

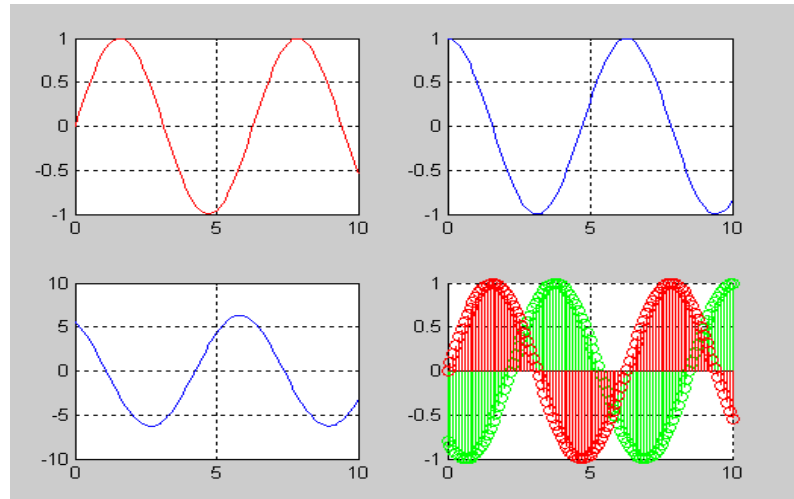


Figure 10.2

Function Files

MATLAB have many built-in functions including trigonometry, logarithm, calculus and hyperbolic functions etc. In addition we can define our own functions and we can use built-in functions in our functions files as well. The function files should be started with the function definition and should be saved with the name of function. The general format of the function file is

Function [output_variables] = function name (input_variables)

See the following example and implement it.

```
% this is a function file
```

```
% this function computes the factorial of a number
```

```
function [y] = my_func (x)
```

```
y = factorial (x);
```

POST LAB

Try to develop a function that will compute the maximum and minimum of two numbers.

Experiment # 11

Communication Signals: Generation and Interpretation

Objective

- To the use of MATLAB for generation of different signals important in communication theory.
- Learn the basics of signals and its operations as used in Analogue Communication.
- To develop understanding of communication signals and their properties.

Generation of Signals

Signals are represented mathematically as a function of one or more independent variables. We will generally refer to the independent variable as time. Therefore we can say a signal is a function of time. Write these instructions in m-file as execute to see the result.

Sinusoidal Sequence:

```
% Example 2.1
% Generation of sinusoidal signals
%  $2\sin(2\pi t - \pi/2)$ 
t = [-5:0.01:5];
x = 2*sin((2*pi*t)-(pi/2));
plot(t,x)
grid on;
axis([-6 6 -3 3])
ylabel('x(t)')
xlabel('Time(sec)')
title('Figure 2.1')
```

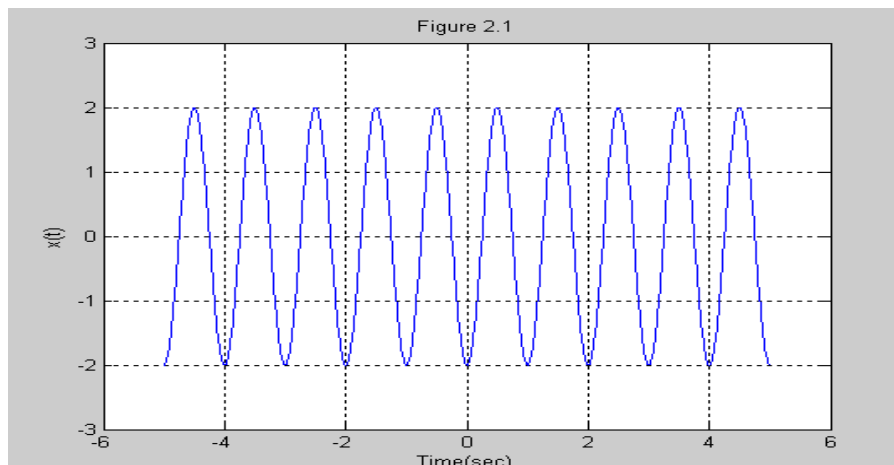


Figure 11.1

See the output, change the phase shift value and observe the differences.

Discrete Time Sequences:

See the example below:

```
% Example 2.2
% Generation of discrete time signals
n = [-5:5];
x = [0 0 1 1 -1 0 2 -2 3 0 -1];
stem (n,x);
axis ([-6 6 -3 3]);
xlabel ('n');
ylabel ('x[n]');
title ('Figure 2.2');
```

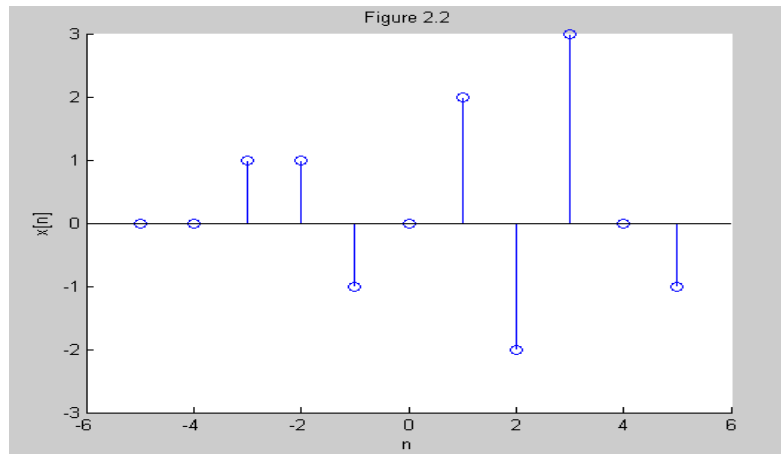


Figure 11.2

Unit Impulse Sequence:

A unit impulse sequence is defined as

$$\Delta(n) = \begin{cases} 1 & n = 0 \\ 0 & n \neq 0 \end{cases}$$

We are making a function named `impseq` and we further use this function in next experiments of this lab. The MATLAB code is given below:

```
function [x,n] = impseq(n0,n1,n2)

% Generates x(n) = delta (n-n0); n1<=n,n0 <= n2
% x[x,n] = impseq(n0,n1,n2)
% n0 = impulse position, n1 = starting index, n2 = ending index
If ((n0 < n1) | (n0 > n2) | (n1 > n2))
    Error('arguments must satisfy n1 <= n0 <= n2')
end
n = [n1:n2];
% x = [zeros(1,(n0-n1)),1,zeros(1,(n2-n0))];

x = [(n-n0) == 0];
```

Unit Step Sequence:

It is defined as

$$u(n) = \begin{cases} 1 & n \geq 0 \\ 0 & n < 0 \end{cases}$$

The MATLAB code for stem sequence function is given below:

```
function [x,n] = stepseq(n0,n1,n2)
% Generates x(n) = u(n-n0); n1 <= n, n0<=n2
% [x,n] = stepseq(n0,n1,n2)
if ((n0 < n1) | (n0 > n2) | (n1 > n2))
    error('arguments must satisfy n1 <= n0 <= n2')
end
n = [n1:n2];
% x = [zeros(1,(n0-n1)),ones(1,(n2-n0+1))];
x = [(n-n0) >= 0];
```

Real Valued Exponential Sequence:

It is define as:

$$x(n) = a^n, \text{ for all } n; a \in \text{Real numbers}$$

We require an array operator “.^” to implement a real exponential sequence. See the MATLAB code below

```
>> n = [0:10];
>> x = (0.9).^n;
```

Observe the result

Complex Valued Exponential Sequence:

It is define as:

$$x(n) = e^{(a + jb)n}, \text{ for all } n$$

Where **a** is called the attenuation and **b** is the frequency in radians. It can be implemented by following MATLAB script.

```
>> n = [0:10];
>> x = exp ((2+3j)*n);
```

Random Sequence:

Many practical sequences cannot be described by the mathematical expressions like above, these are called random sequences. They depend upon the parameters like probability density function or their statistical moments. In MATLAB two types of random sequences are available. See the code below:

```
>> rand (1,N)
```

And

```
>> randn (1,N)
```

The above instruction generates a length **N** random sequence whose elements are uniformly distributed between [0,1]. And the last instruction, **randn** generates a length **N** Gaussian random sequence with mean 0 and variance 1. Plot these sequences.

% example 2.3

%Generation of random sequence

```
n = [0:10];
x = rand (1, length (n));
y = randn (1, length (n));
plot (n,x) ;
grid on;
hold on;
plot(n,y,'r');
ylabel ('x & y')
xlabel ('n')
title ('Figure 2.3')
```

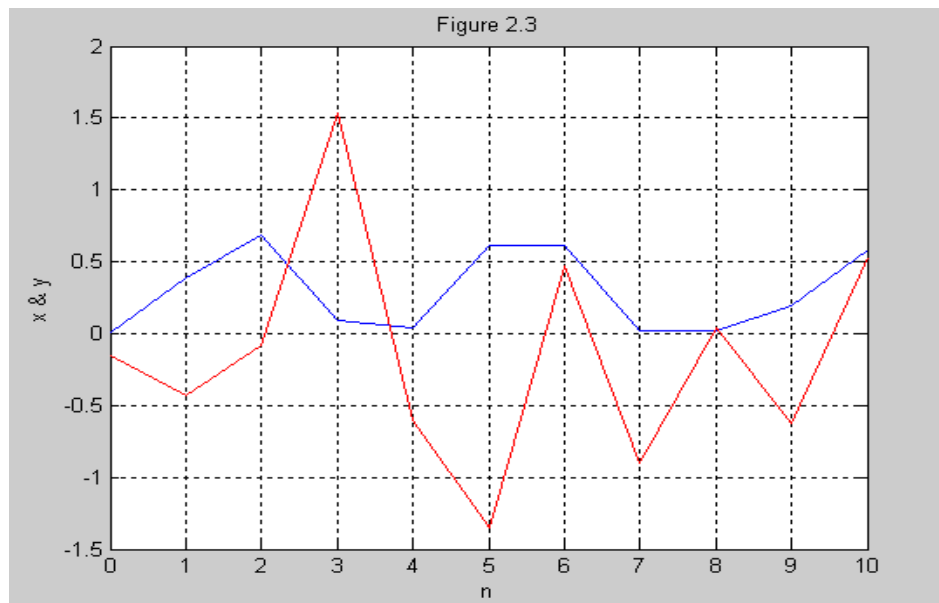


Figure 11.3

Periodic Sequences:

A sequence is periodic if it repeats itself after equal interval of time. The smallest interval is called the fundamental period. Implement code given below and see the periodicity.

% Example 2.4

% Generation of periodic sequences

```
n = [0:4];
x = [1 1 2 -1 0];
subplot (2,1,1);
stem (n,x);
grid on;
axis ([0 14 -1 2]);
xlabel ('n');
ylabel ('x(n)');
title ('Figure 2.4(a)');
xtilde = [x,x,x];
length_xtilde = length (xtilde);
n_new = [0:length_xtilde-1];
subplot (2,1,2);
stem (n_new,xtilde,'r');
grid on;
xlabel ('n');
ylabel ('periodic x(n)');
title ('Figure 2.4(b));
```

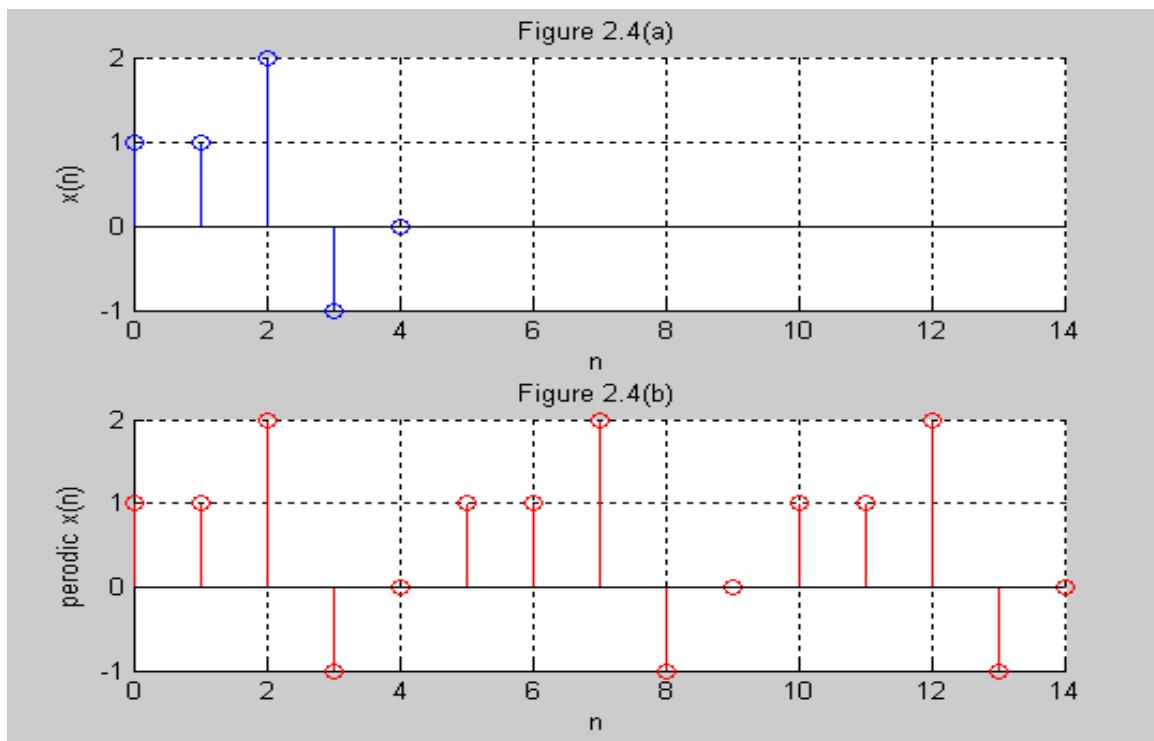


Figure 11.4

SIGNALS OPERATIONS:

Signal Addition

This is basically sample by sample addition. The definition is given below:

$$\{x_1(n)\} + \{x_2(n)\} = \{x_1(n) + x_2(n)\}$$

The length of x1 and x2 should be equal. See the MATLAB code below:

```
function [y,n] = sigadd(x1,n1,x2,n2)

% implement  $y(n) = x1(n) + x2(n)$ 

% [y,n] = sigadd (x1,n1,x2,n2)

% y = sum sequence over n, which include n1 and n2

% x1 = first sequence over n1

% x2 = second sequence over n2 (n2 can be different from n1)

n = min(min(n1),min(n2)): max(max(n1),max(n2));    %duration of y(n)

y1 = zeros(1,length(n));                          % initialization

y2 = y1;

y1(find((n>=min(n1))&(n<=max(n1))==1))=x1;          % x1 with duration of y

y2(find((n>=min(n2))&(n<=max(n2))==1))=x2;          % x2 with duration of y

y = y1 + y2;
```

See example of signal addition below

% Example 2.5

% signal addition using sigadd function

```
clear;
clc;
n1 = [0:10];
x1 = sin (n1);
n2 = [-5:7];
x2 = 4*sin(n2);
[y,n] = sigadd(x1,n1,x2,n2);
subplot (3,1,1);
stem (n1,x1);
grid on;
axis ([-5 10 -5 5]);
xlabel ('n1');      ylabel ('x1(n)');
title ('1st signal');
subplot (3,1,2);
stem (n2,x2);
grid on; hold on;
axis ([-5 10 -5 5]);
xlabel ('n2');      ylabel ('x2(n)');
title ('2nd signal');
subplot (3,1,3);   stem (n,y,'r');
grid on;
```

```
axis([-5 10 -5 5]);
xlabel('n');      ylabel('y(n)');
title('Added Signals');
```

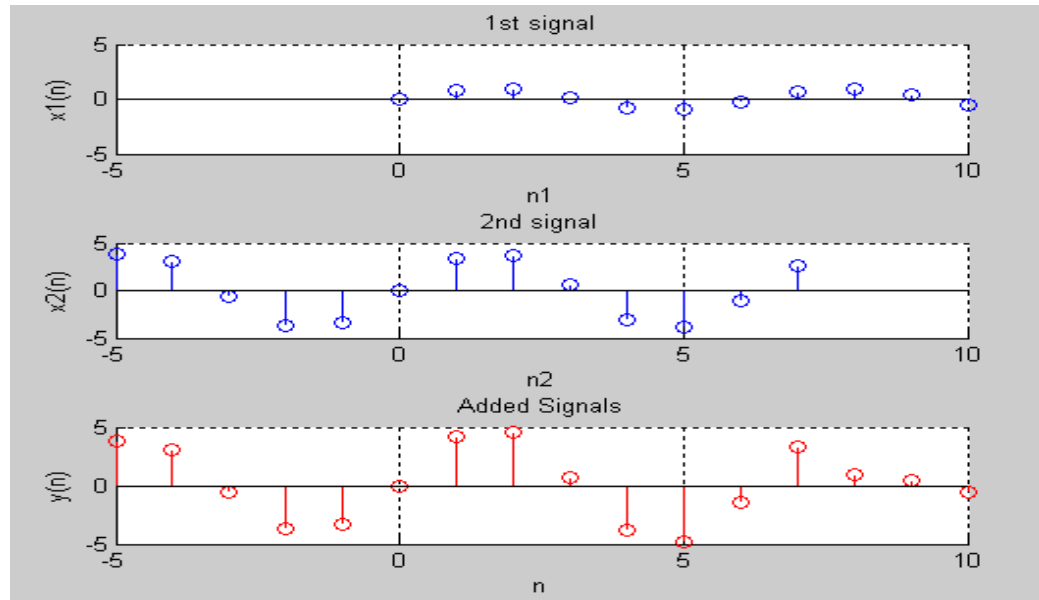


Figure 11.5

Signal Multiplication:

The multiplication of two signals is basically sample by sample multiplication or you can say dot multiplication. By definition it is

$$\{x1(n)\} \cdot \{x2(n)\} = \{x1(n)x2(n)\}$$

It is implemented by the array operator ‘.*’ that we studied in last lab. A signal multiplication function is developed that is similar to the sigadd function. See the code below:

```
function [y,n] = sigmult (x1,n1,x2,n2)
% implement y(n) = x1(n) * x2 (n)
% [y,n] = sigmult (x1,n1,x2,n2)
% y = product sequence over n, which include n1 and n2
% x1 = first sequence over n1
% x2 = second sequence over n2 (n2 can be different from n1)
n = min(min(n1),min(n2)):0.1 : max(max(n1),max(n2)); %duration of y(n)
```

```

y1 = zeros(1,length(n));           % initialization
y2 = y1;
y1(find((n>=min(n1))&(n<=max(n1))==1))=x1;    % x1 with duration of y
y2(find((n>=min(n2))&(n<=max(n2))==1))=x2;    % x2 with duration of y
y = y1 .* y2;

```

See the example below:

% Example 2.6

% signal multiplication using sigmult function

```

clear;
clc;
n1 = [0:0.1:10];
x1 = sin (n1);
n2 = [-5:0.1:7];
x2 = 4*sin (n2);
[y,n] = sigmult(x1,n1,x2,n2);
subplot (3,1,1);
stem (n1,x1);
grid on;
axis ([-5 10 -5 5]);
xlabel ('n1');
ylabel ('x1(n)');
title ('1st signal');
subplot (3,1,2);
stem (n2,x2);
grid on;
hold on;
axis ([-5 10 -5 5]);
xlabel ('n2');
ylabel ('x2(n)');
title ('2nd signal');
subplot (3,1,3);
stem (n,y,'r');
grid on;
axis ([-5 10 -5 5]);
xlabel ('n');
ylabel ('y(n)');
title ('Multiplied Signals');

```

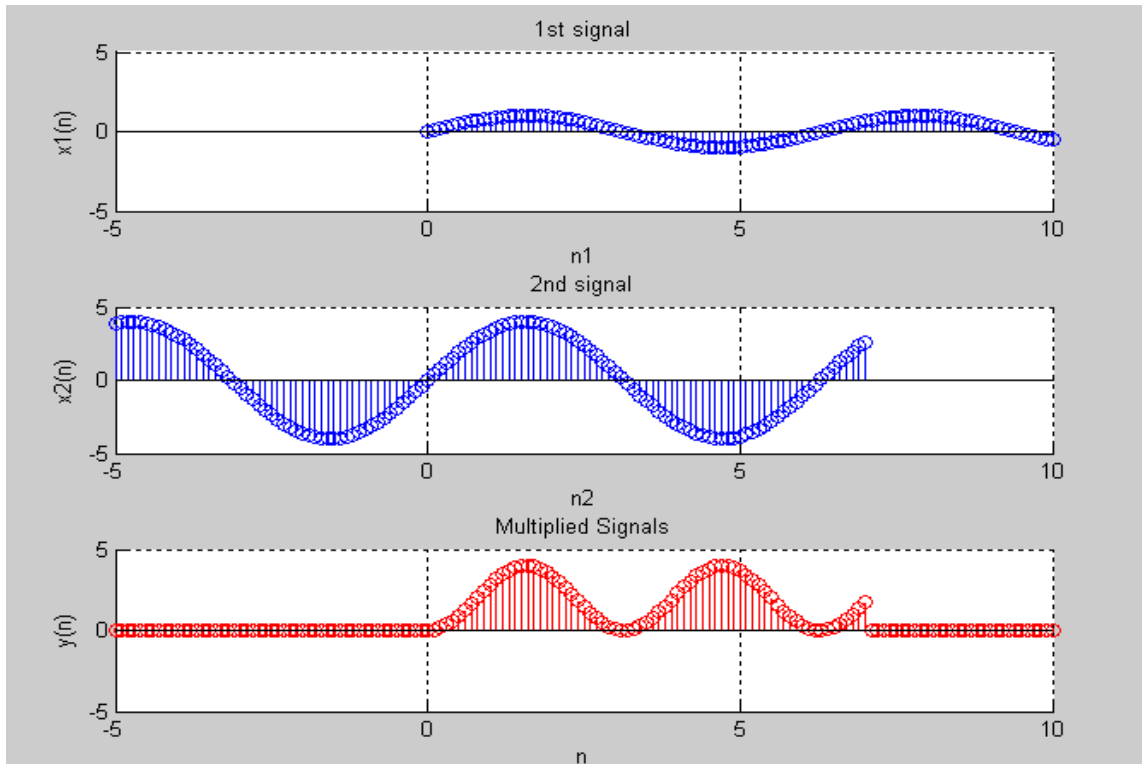


Figure 11.6

POST LAB

Write MATLAB code to plot these signals:

- $x[n] = 2\sin(3n) + 2\cos(3n)$
- $x[n] = u[n] + 4\cos(3n)$
- $x[n] = n[u(n) - u(n-10)] + 10e^{-0.3(n-10)}[u(n-10) - u(n-20)]$

You are not allowed to multiply impulse sequences with a number. Implement this by using **impseq**, **stepseq** and **sigadd** functions.

Experiment # 12

Communication Signals: Operations

Objective

- To learn the use of MATLAB for different operations on signals.
- To develop a thorough understanding of communication signals and their basic operations as used in Analogue Communication.

SUMMARY

- Signal operations (Scaling, Shifting, Folding, Sample Summation, Sample product, Energy, Even and Odd sequences, Convolution)

SIGNAL OPERATIONS:

1. Scaling:

In this operation the samples of the signal is multiplied by a scalar α . The mathematical operator $*$ is used for the implementation of the scaling property.

$$\alpha\{x(n)\} = \{ \alpha x(n) \}$$

```
>> [x,n] = stepseq (-1,-5,5);
```

```
>> a = 2;
```

```
>> y = a*x;
```

```
>> subplot (2,1,1);
```

```
>> stem (n,x);grid on;
```

```
>> subplot (2,1,2);
```

```
>> stem (n,y, 'r');
```

```
>> grid on;
```

2. Shifting

In this operation, each sample of the signal is shifted by k to get a shifted signal. By definition:

$$y(n) = \{x (n-k)\}$$

In this operation there is no change in the array or vector x , that contains the samples of the signal. Only n is changed by adding k to each element. The function is given below:

```
function [y,n] = sigshift (x,m,n0)
```

```
% implement  $y(n) = x(n-n0)$ 
```

```
% x = samples of original signal
% m = index values of the signal
% n0 = shift amount , may be positive or negative
% [y,n] = sigshift(x,m,n0)
n = m+n0;
y = x;
```

See the example of above function:

% Example 3.1

% This example demonstrate the use of stepseq, sigshift, sidadd & sigmult function

```
clc; clear;

%-----

[x,n] = stepseq (0,-10,10);
subplot (3,2,1);
stem (n,x);
axis ([-12 12 0 3]);
grid on;
xlabel ('n');
ylabel ('x(n)');
title ('Original Signals');

%-----

[y1,n1] = sigshift(x,n,2.5);
subplot (3,2,2);
stem (n1,y1);
axis ([-12 12 0 3]);
grid on;
xlabel ('n');
ylabel ('y1(n)');
title ('Shifted signal,x(n-2.5)');
```

```
%-----

[y2,n2] = sigshift (x,n,-2.5);

subplot (3,2,4);

stem (n2,y2);

axis ([-12 12 0 3]);

grid on;

xlabel ('n');

ylabel ('y2(n)');

title ('Shifted signal,x(n+2.5)');

%-----

[y_add,n_add] = sigadd(y1,n1,y2,n2);

subplot (3,2,6);

stem (n_add,y_add,'r');

axis ([-12 12 0 3]);

grid on;

xlabel ('n');

ylabel ('y1(n) + y2(n)');

title ('Added Signal');

%-----

[y_mul,n_mul] = sigmult(y1,n1,y2,n2);

subplot (3,2,5);

stem (n_mul,y_mul,'k');

axis ([-12 12 0 3]);

grid on;

xlabel ('n');

ylabel ('y1(n) * y2(n)');

title ('Multiplied Signal');

%-----
```

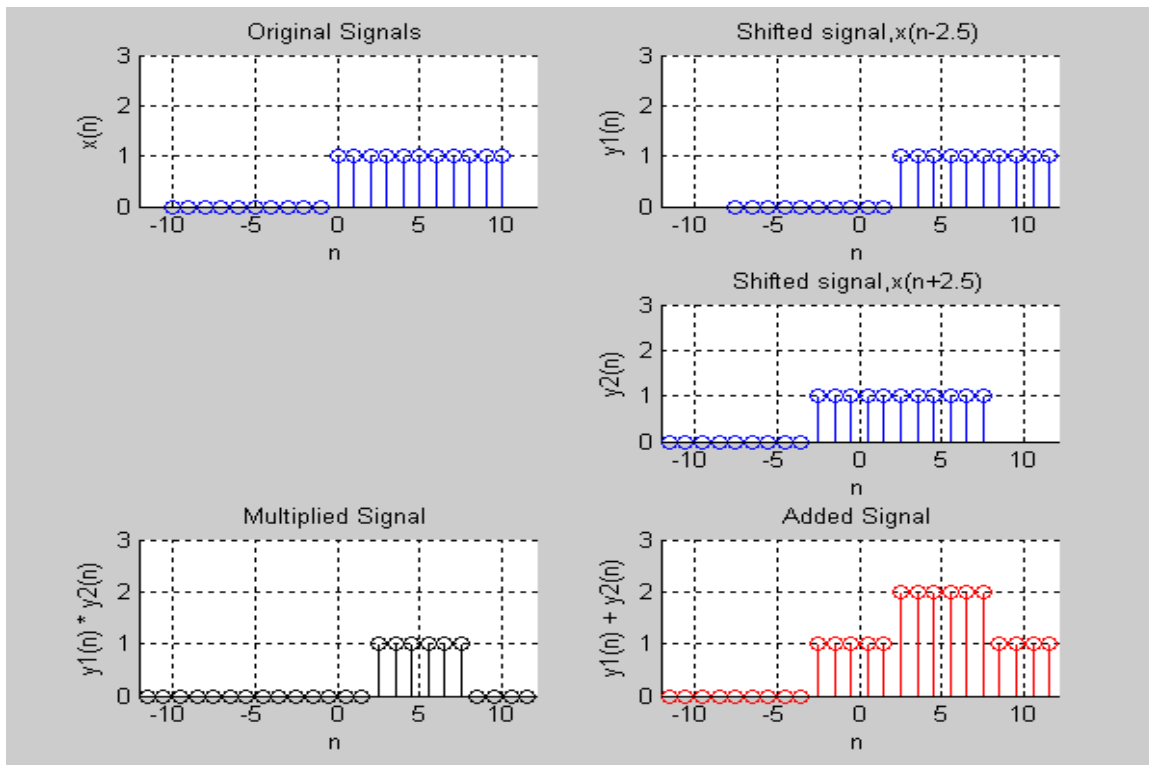


Figure 12.1

3. Folding:

In this operation each sample of $x(n)$ is flipped around $n=0$ to obtain a folded signal $y(n)$.

$$y(n) = \{x(-n)\}$$

In MATLAB, this function is implemented by using a built-in function **fliplr(x)** and **-fliplr(x)**. Its implementation is given below:

```
function [y,n] = sigfold(x,n)
```

```
% implements y(n) = x(-n)
```

```
% [y,n] = sigfold(x,n)
```

```
% x = samples of the original signal
```

```
% n = indexes of the original signal
```

```
y = fliplr(x);
```

```
n = -fliplr(n);
```

Do its example by yourself from any example signals.

4. Sample Summation:

This operation is different from **sigadd** function. In this operation we add all the sample values of any signal **x(n)** between any two of its index values. By definition

$$\sum x(n) = x(n1) + + x(n2)$$

In MATLAB it is implemented by the **sum(x(n1:n2))** command. See the code below for the demonstration of above function.

```
>> [x,n] = stepseq (5,0,10)
```

```
>> sum(x(2:7))
```

5. Sample Product:

This operation also differs from the **sigmult** function. It implies the sample values over the range **n1:n2**. It is implemented by the **prod(x(n1:n2))**. See the code below.

```
>> x = [0 1 2 3 4 5]
```

```
>> prod(x(2:5))
```

6. Energy:

The energy of any signal x is computed by the mathematical relation:

$$E_x = \sum x(n) x^*(n) = \sum |x(n)|^2$$

Where the subscript * is used for complex conjugate of the signal x. The energy of the finite duration signal is computed in MATLAB as.

```
>> Ex = sum (x.*conj(x));
```

Or

```
>> Ex = sum (abs(x).^2);
```

7. Even and Odd Sequence:

A real valued sequence $x_e(n)$ is called even if the following condition satisfies.

$$x_e(-n) = x_e(n)$$

Similarly a signal is said to be an odd signal if

$$x_o(-n) = -x_o(n)$$

See the example below:

```
% example 3.2
```

```
% Generation of even and odd signals
```

```
n1 = [0:0.01:1];
```

```
x1 = 2*n1;
n2 = [1:0.01:2];
x2 = -2*n2+4;
n = [n1,n2];
x = [x1,x2];
%Even Signal
[xe,ne] = sigfold(x,n);
%Plotting of original signal
subplot (3,1,1);
plot (n,x);
axis ([-4 4 0 2.5]);
grid on;
%Plotting of original signal + even signal
subplot (3,1,2);
plot (n,x/2,ne,x/2);
axis ([-4 4 0 2.5]);
grid on;
% Plotting of original signal + odd signal
xo = -xe;
no = ne;
subplot (3,1,3);
plot (n,x/2,no,xo/2);
axis ([-4 4 -2.5 2.5]);
grid on;
```

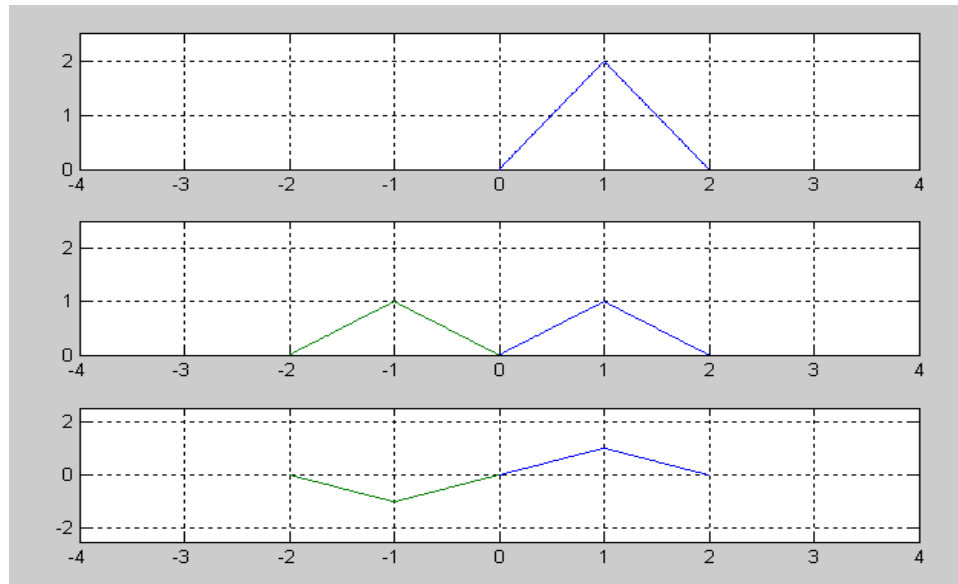


Figure 12.2

The above example shows to develop the even and odd signals from a given signal. Now we are going to develop a function to compute the even and odd signals for ourselves. See the code of function file below:

```
function [xe,xo,m] = evenodd (x,n)

% Decomposes a real function into its even and odd parts

% [xe,xo,m] = evenodd(x,n)

% xe = even signal

% xo = odd signal

% m = indexes

% x = original signal

% n = indexes for original signal

if any(imag(x)~=0)
    error('x is not a real sequence')
end

m = -fliplr(n);
m1 = min([m,n]);
m2 = max([m,n]);
m = m1:m2;
nm = n(1)-m(1);
```

```
n1 = 1:length(n);
x1 = zeros(1,length(m));
x1(n1+nm) = x;
x = x1;
xe = 0.5*(x+fliplr(x));
xo = 0.5*(x-fliplr(x));
```

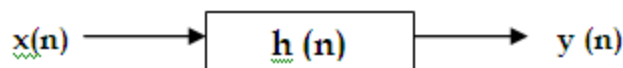
Now change the example 3.2 code to implement the same example with this function.

8. Convolution:

The convolution is very important operation as far the system as their impulse responses are concern. It is mathematically defines as:

$$y(n) = x(n) * h(n)$$

Where $h(n)$ is the impulse response of the system. The above definition is best depicted by the following diagram.



In MATLAB convolution is implemented by the following instructions.

```
>> x = [1 5 3 9 1 2 3 8 5 -3 0 4];
>> h = [1 0 2 3];
>> y = conv(x,h);
```

A function is developed which will evaluate convolution in a more precise form and also calculate the indexes to help us plot the sequences.

```
function [y,ny] = conv_m(x,nx,h,nh)

% Modified convolution routine for signal processing

% [y,ny] = conv_m(x,nx,h,nh)
% [y,ny] = convolution result
% x = original signal
% nx = index values
% h = impulse response signal
% nh = index values for impulse response

nyb = nx(1) + nh(1);
```

```
nye = nx(length(x)) + nh(length(h));
```

```
ny = [nyb:nye];
```

```
y = conv(x,h);
```

POST LAB

- $x(n) = u(n) - u(n-5)$. Decompose into even and odd components and plot them.
- The impulse response of LTI system is $h(n) = \delta(n-2)$, if the input to this system is a arbitrary sequence $x(n)$ of length 10, then plot the original and the convolved outputs of the system. What is the change if the $h(n) = x(n)$ and input signal is now the previous impulse response of the system.

c. $n = [-2:2]$

```
x1 = [3,2,1,-2,-3];
```

```
x2 = [1,1,1,1,1]
```

Implement $y = x1 * x2$

Experiment # 13

Introduction to Amplitude Modulation (Simulink Implementation)

Objective

- To identify the spectrum analyzer as used in frequency domain analysis
- To identify various types of linear modulated waveforms in time and frequency domain representation
- To implement theoretically functional circuits using the Communication Module Design System (CMDS)

Spectrum Analyzer and Function Generator

This section deals with looking at the spectrum of simple waves. We first look at the spectrum of a simple sine wave

To start Simulink: Start MATLAB then type simulink on the command line. A Simulink Library Window opens up as shown in figure 13.1.

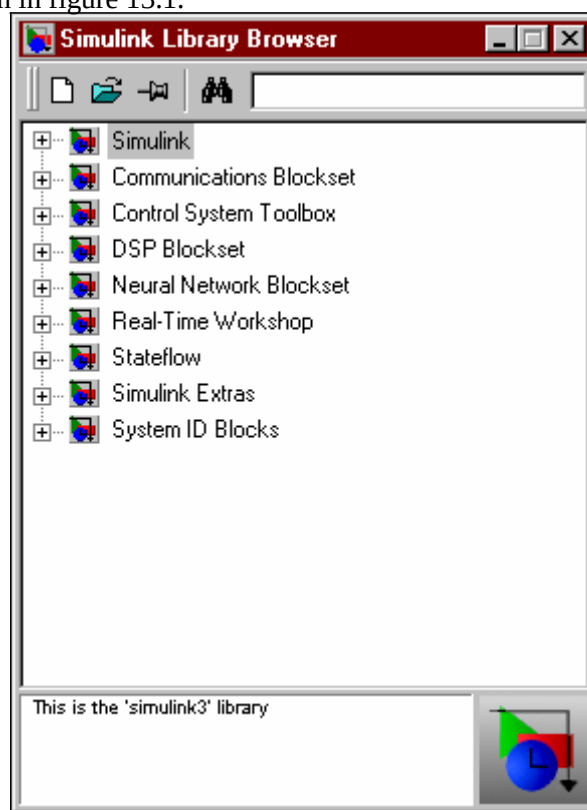


Figure 13.1

Spectrum of a simple sine wave: - Figure 13.2 shows the design for viewing the spectrum of a simple sine wave.

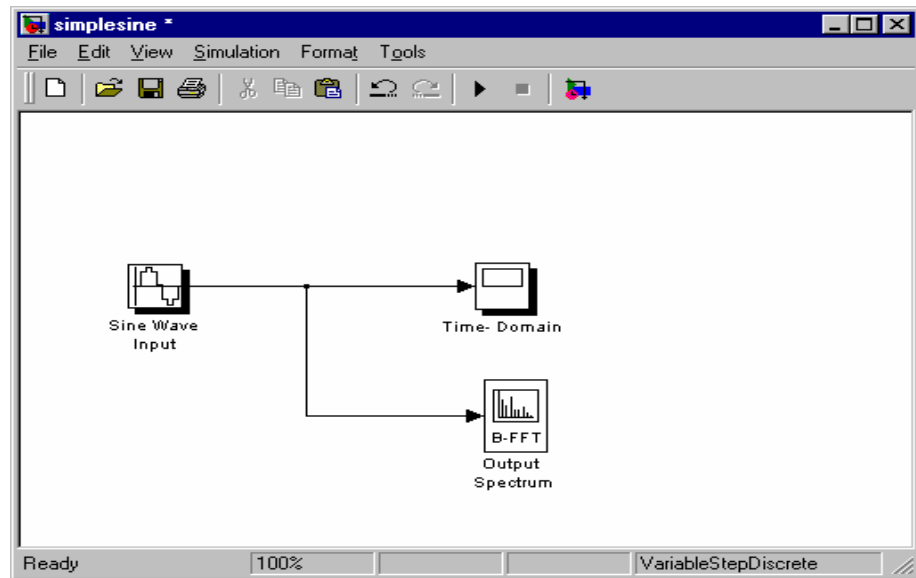


Figure 13.2

Figure 13.3 shows the time-domain sine wave and the corresponding frequency domain is shown in figure 13.4. The frequency domain spectrum is obtained through a buffered-FFT scope, which comprises of a Fast Fourier Transform of 128 samples which also has a buffering of 64 of them in one frame. The property block of the B-FFT is also displayed in figure 13.5.

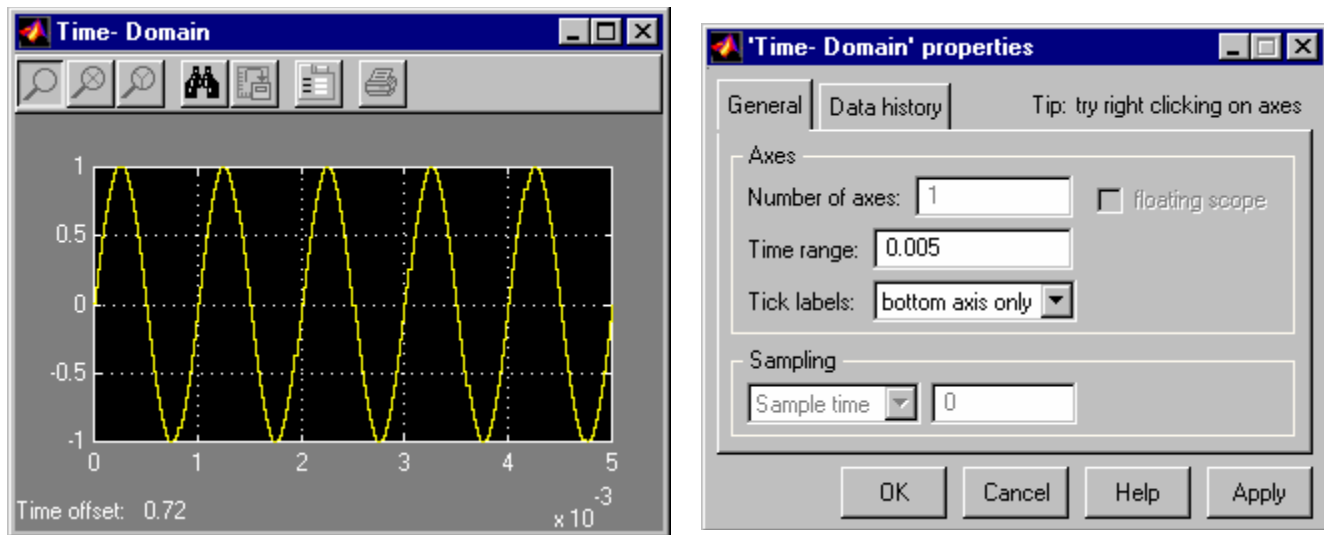


Figure 13.3

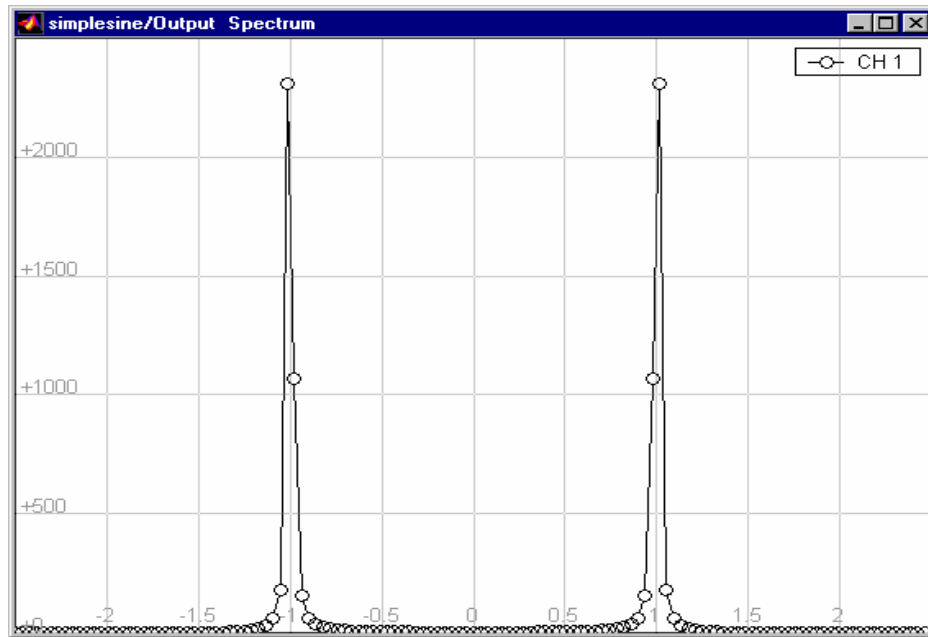


Figure 13.4

This is the property box of the Spectrum Analyzer

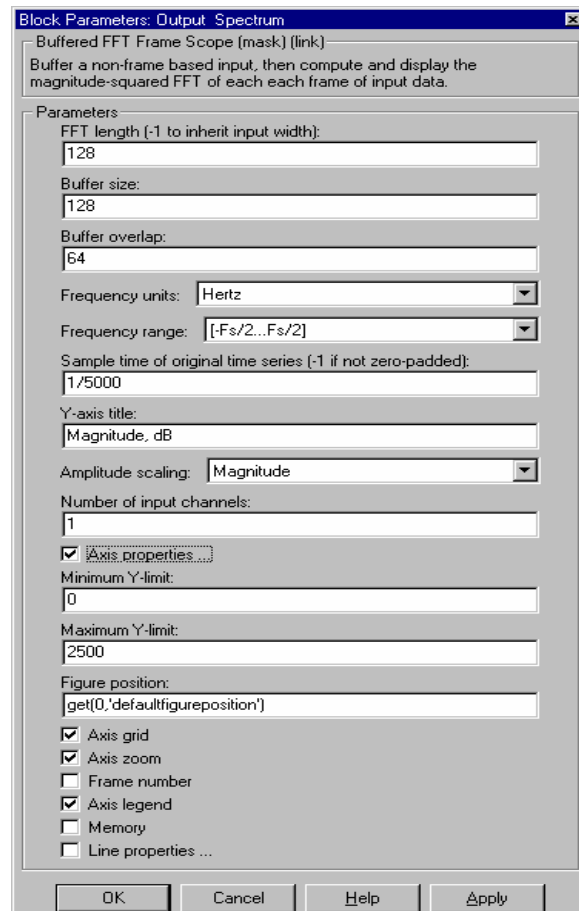


Figure 13.5

From the property box of the B-FFT scope the axis properties can be changed and the Line properties can be changed. The line properties are not shown in the above. The Frequency range can be changed by using the frequency range pop down menu and so can be the y-axis the amplitude scaling be changed to either real magnitude or the dB (log of magnitude) scale. The upper limit can be specified as shown by the Min and Max Y-limits edit box. The sampling time in this case has been set to 1/5000.

Note: The sampling frequency of the B-FFT scope should match with the sampling time of the input time signal.

Also as indicated above the FFT is taken for 128 points and buffered with half of them for an overlap.

Calculating the Power:

The power can be calculated by squaring the value of the voltage of the spectrum analyzer.

Note: The signal analyzer if chosen with half the scale, the spectrum is the single-sided analyzer, so the power in the spectrum is the total power.

Similar operations can be done for other waveforms – like the square wave, triangular. These signals can be generated from the signal generator block.

II. Waveform Multiplication (Modulation)

The equation $y = k_m \cos^2(2\pi(1,000)t)$ was implemented as in fig. 1B peak to peak voltage of the input and output signal of the multiplier was measured. Then k_m can be computed as

$$k_m = \frac{V_{pp}(2\text{ kHz})}{V_{pp}(1\text{ kHz})} * 2 = \frac{0.5}{2} * 2 = 0.5$$

The spectrum of the output when $k_m=1$ was shown below:

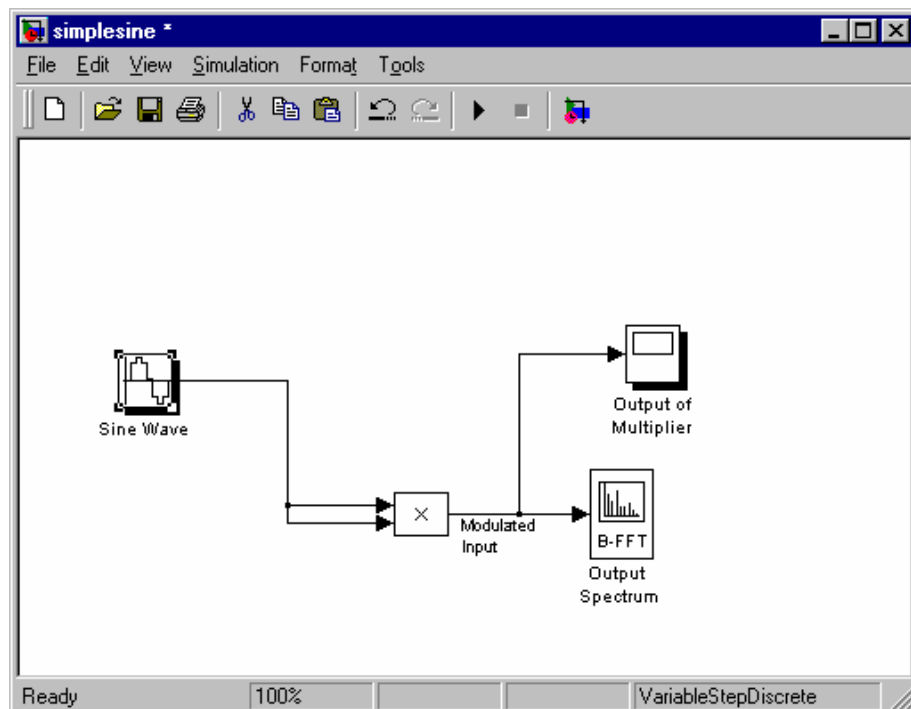


Figure 13.6

The following figure demonstrates the waveform multiplication. A sine wave of 1 kHz is generated using a sine wave generator and multiplied with a replica signal. The input signal and the output are shown in figures.

The input signal as generated by the sine wave is shown in figure.

The output of the multiplier is shown in figure and the spectral output is shown in figure.

It can be seen that the output of the multiplier in time domain is basically a sine wave but doesn't have the negative sides since they get cancelled out in the multiplication.

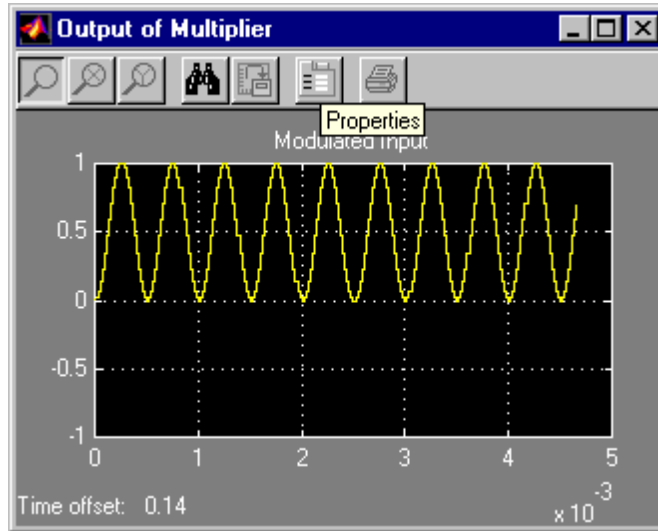


Figure 13.7

The spectral output of the spectrum is shown below. It can be seen that there are two side components in spectrum. The components at $f_c + f_m$ and $-(f_c + f_m)$ can be seen along with a central impulse.

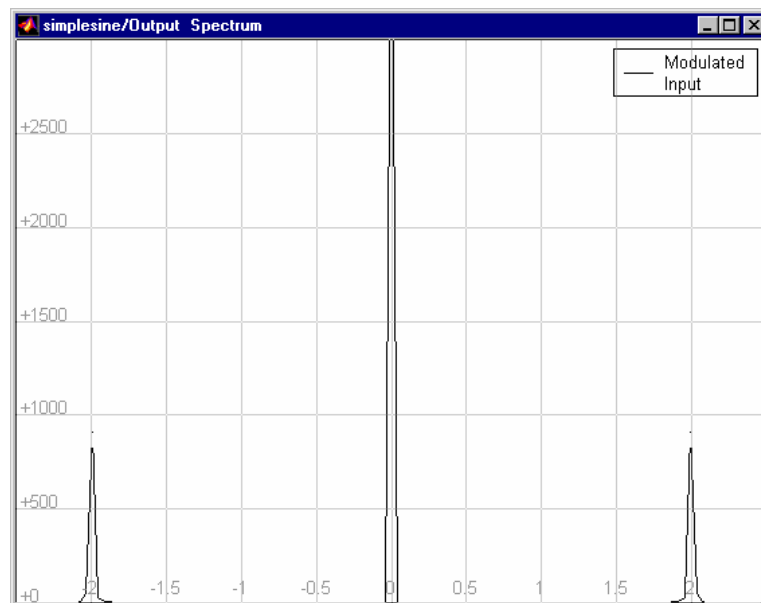


Figure 13.8

If a DC component was present in the input waveform, then

$$y = k_m * (\cos(2\pi(1,000)t) + V_{dc})^2$$

The effect of adding a dc component to the input has the overall effect of raising the amplitude of the 2 KHz component and decreases the 2 KHz component. However, for a value of $V_{dc} = 0.1V$, the 1KHz component reduces and for any other increase in the V_{dc} value, the 1KHz component increases.

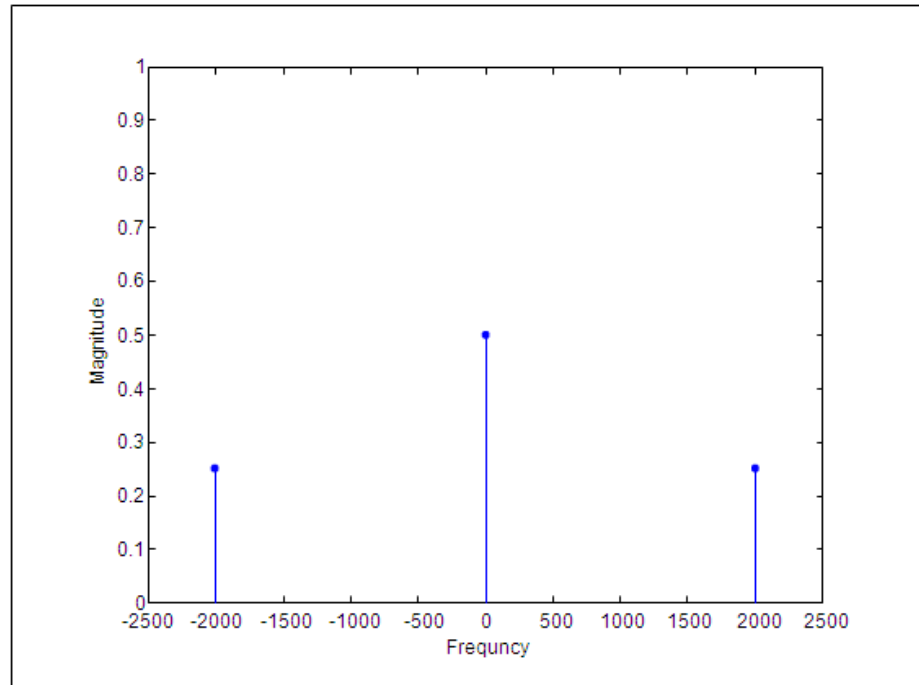


Figure 13.9

I. Double Side-Band Suppressed Carrier Modulation

Figure shows the implementation of a DSB-SC signal. The Signals are at 1 kHz and 10 kHz.

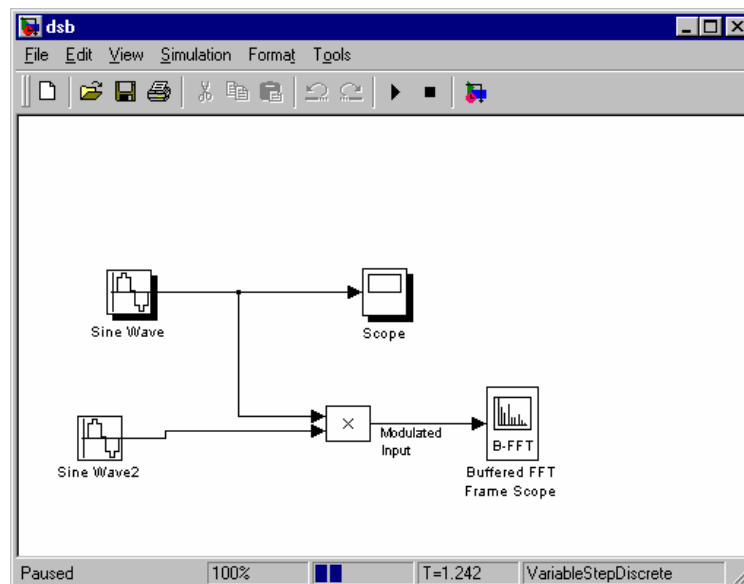


Figure 13.10

The output is shown below. It can be seen that the output consists of just two side bands at $+(f_c + f_m)$ and the other at $-(f_c + f_m)$, i.e. at 9kHz and 11kHz.

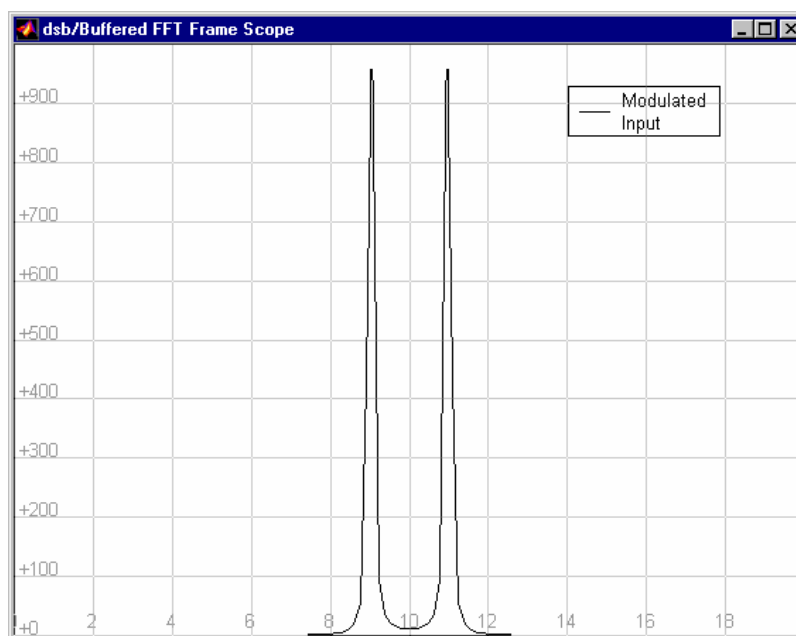


Figure 13.11

By multiplying the carrier signal and the message signal, we achieve modulation.

$$Y^*m(t) = [k_m \cos(2\pi 1000t) * \cos(2\pi 10000t)]$$

We observe the output to have no 10 KHz component i.e., the carrier is not present. The output contains a band at 9 KHz ($f_c - f_m$) and a band at 11 KHz ($f_c + f_m$). Thus we observe a double side band suppressed carrier. All the transmitted power is in the 2 sidebands.

Effect of Variations in Modulating and Carrier frequencies on DSB – SC signal.

By varying the carrier and message signal frequencies, we observe that the 2 sidebands move according to equation $f_c \pm f_m$.

Now, using a square wave as modulating signal, we see that DSBSC is still achieved.

The output from spectrum analyzer was slightly different from the theoretical output. In the result from the spectrum analyzer, there is a small peak at frequency = 10kHz (the carrier frequency) and other 2 peak at 0 and 1000 Hz. This may be caused by the incorrectly calibrated multiplier.

Next, the changes to the waveform parameters have been made and then the outputs have been observed. And here are the changes that have been made

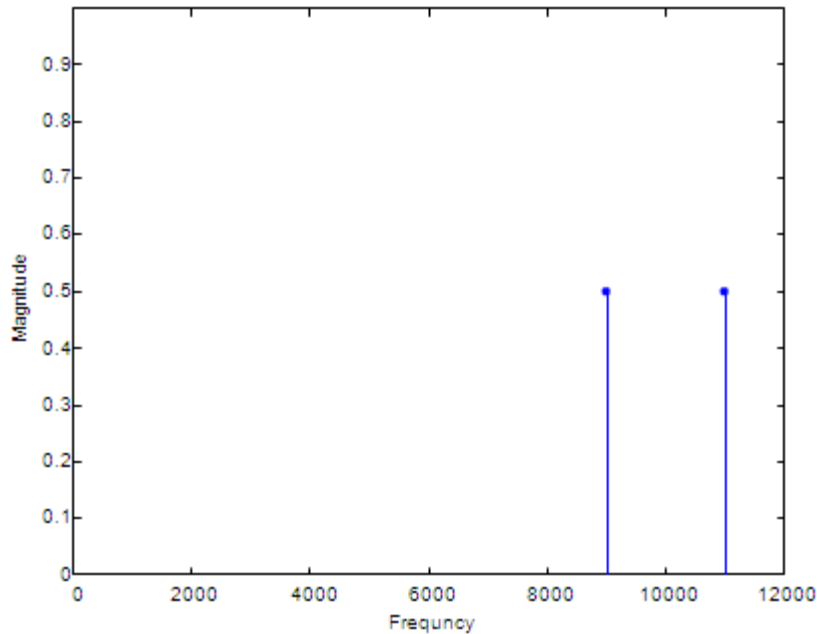


Figure 13.12

1 Vary the 10 kHz carrier frequency

Expected result: Both sidebands are expected to be centered on the new carrier frequency.

The real result is as expected.

2 Vary the modulating frequency and amplitude

Expected result: The position of the sidebands would have been changed when the modulating frequency is changed. The sidebands would move further from the carrier frequency if the modulating frequency is increased. The peak of the sidebands would be higher if the amplitude of the modulating signal increases.

The result of the experiment is as expected.

3 Change the carrier signal to a square wave.

Expected result: There would be the high peaks of the modulating signal around the carrier frequency. Expect for a small peak of the carrier because the time average of the square wave does not equal to zero. The waveform of the signal is expected to be square wave which the amplitude is the sine wave at 1 KHz. The result of the experiment is as expected

4 Change the modulating signal to a square wave

Expected result: It is likely to see the spectrum of the square wave in the both sidebands around the carrier frequency. The output waveform would be the sine wave, which the amplitude equals to the amplitude of the square wave.

The result of the experiment is as expected.

Amplitude Modulation

This experiment is the amplitude modulation for modulation index $a = 1$ and 0.5 .

From the equation of the AM

$$y = k_m (1 + a \cdot \cos(2\pi (1000)t) \cdot \cos(2\pi (10000)t)$$

The representation of the signal in both time-domain and frequency domain when $k_m=1$ for $a=1$ and $a=0.5$ were found to be as shown in figures.

The experimental set up for generating an AM signal looks like this: -

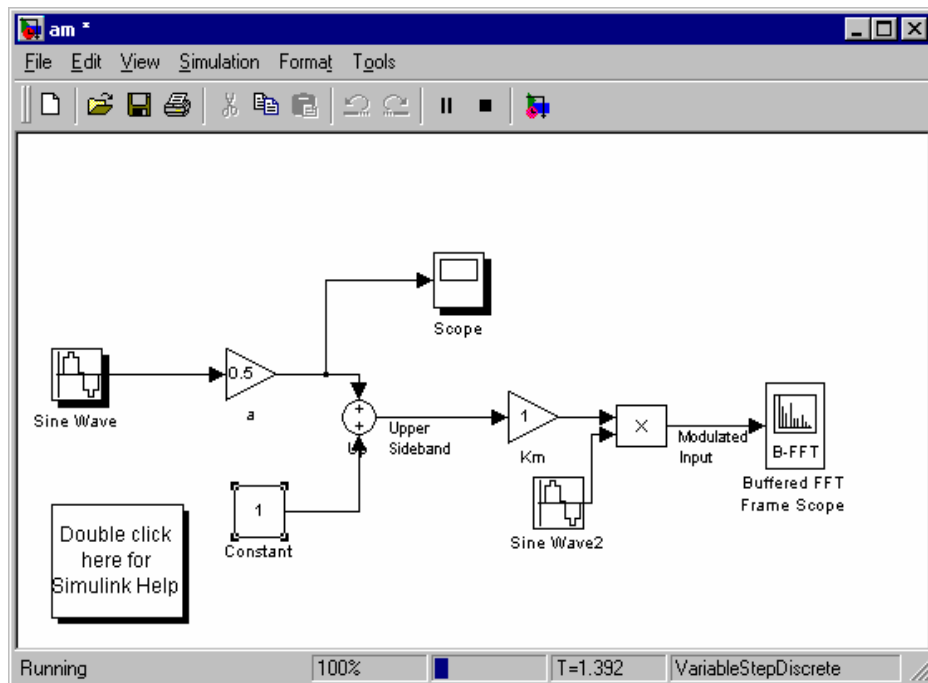


Figure 13.12

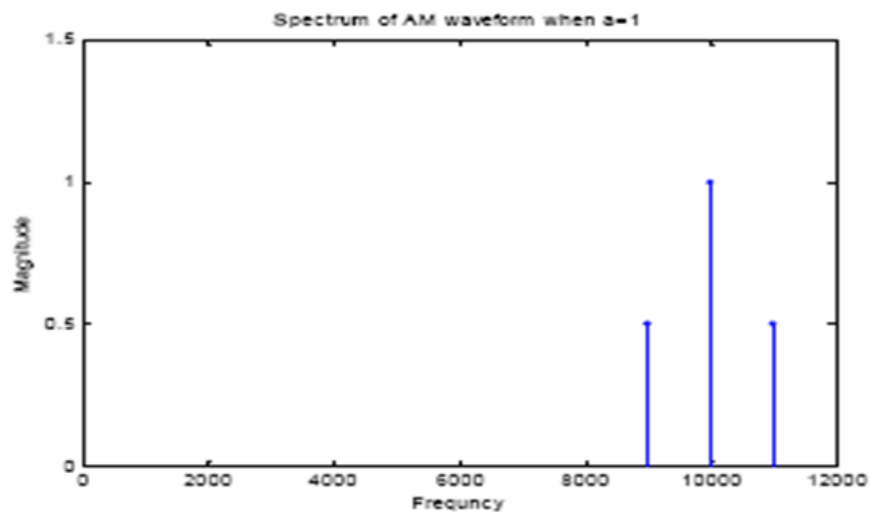


Figure 13.13

The input waveform 50% modulated is shown in figure:

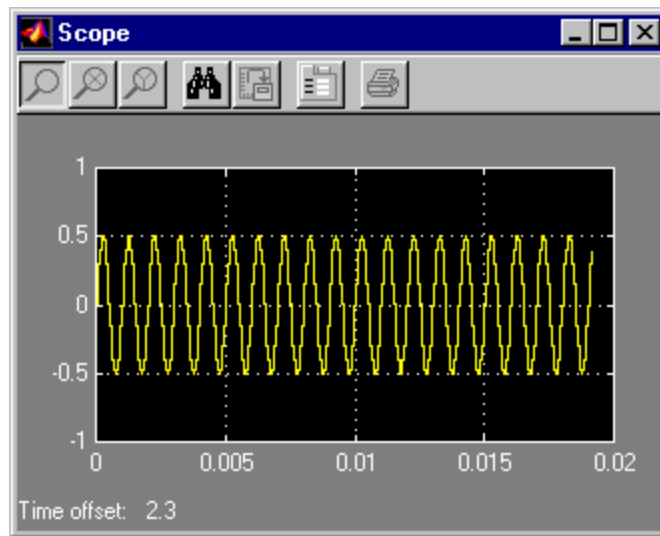


Figure 13.14

The output spectrum is shown below

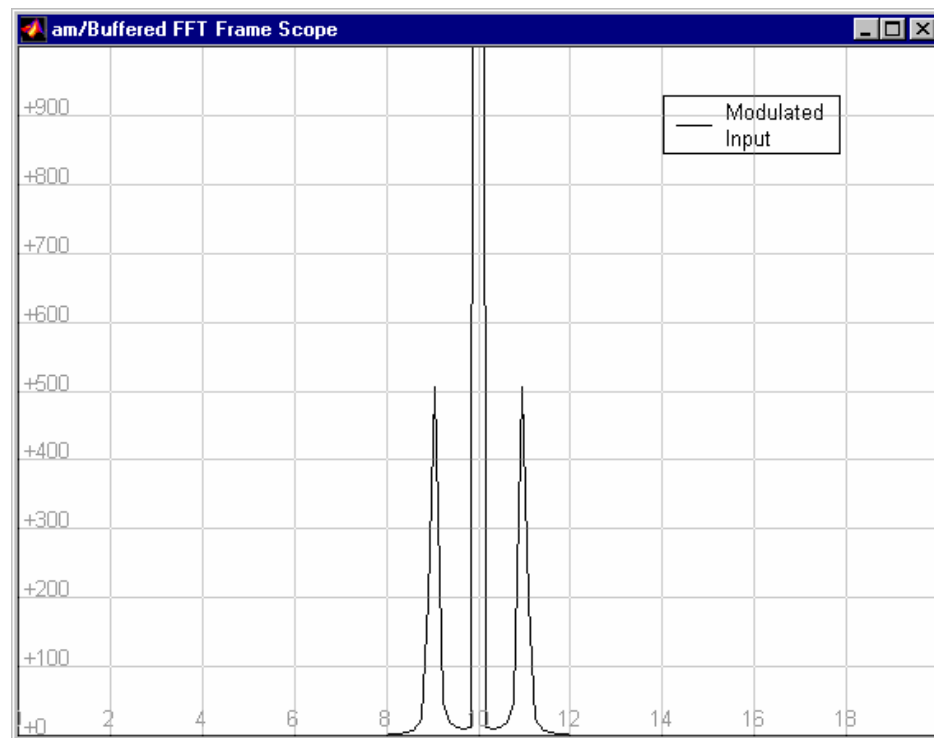


Figure 13.15

It must be noted here that the A.M signal can be converted into a DSB-SC signal by making the constant = 0.

The waveforms at various levels of modulation are shown in the following figures.

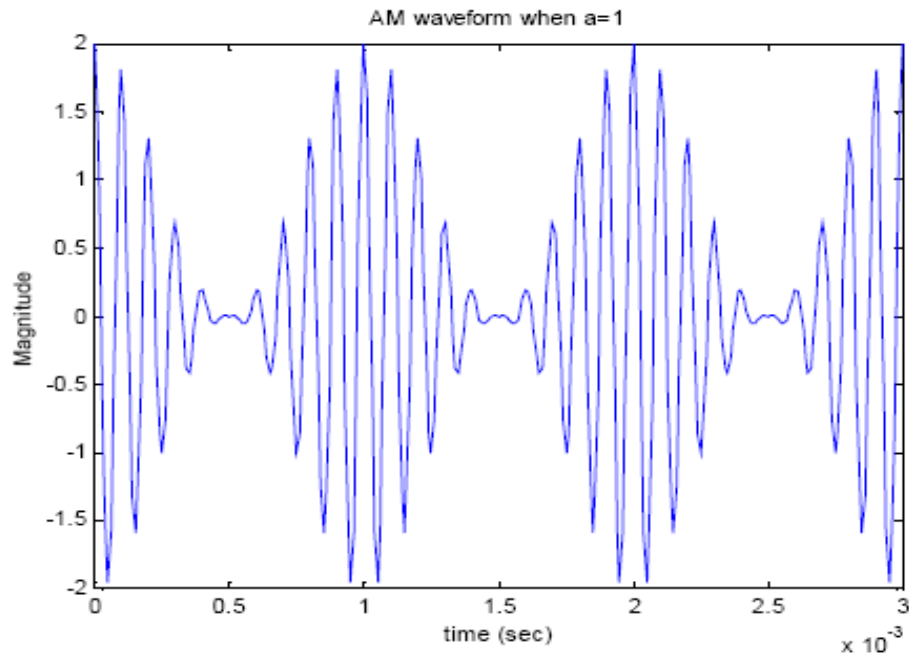


Figure 13.16

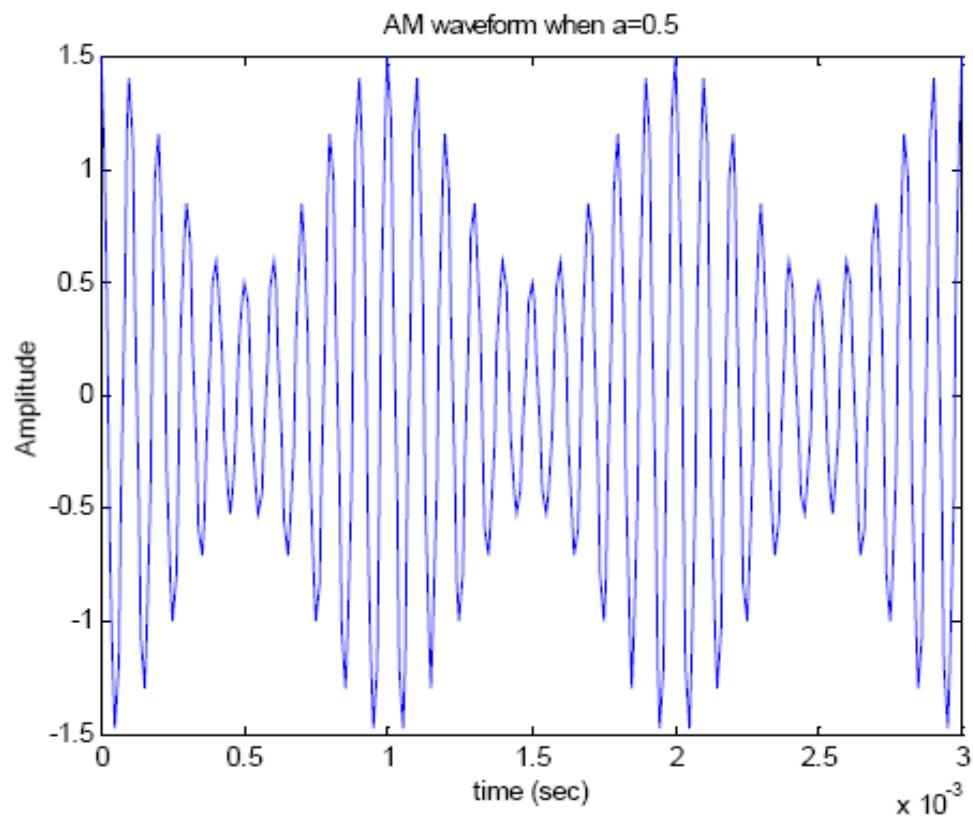


Figure 13.17

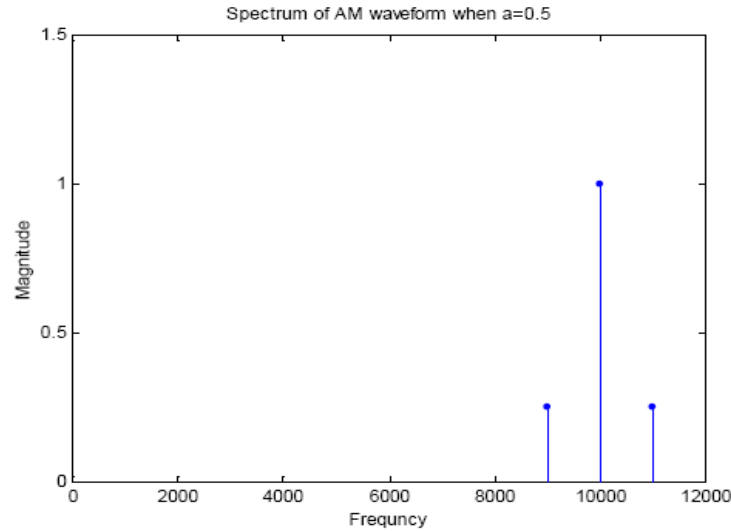


Figure 13.18

The results from the experiment were shown. The results from the experiment are pretty much the same as in the theoretical ones except there are 2 other peaks at 0 and 1000 kHz. This is the same as earlier experiment. The cause of this problem is probably the multiplier.

II. Two Tone Modulation

The last experiment in this section is the two tone modulation. In this experiment, the 2 kHz signal had been added to the modulating signal in the above experiment. Theoretically, the representation of the modulated signal in time-domain and frequency domain would have been as in the figure below. In the figure, 1 kHz and 2 kHz signals were modulated with 10 kHz carrier.

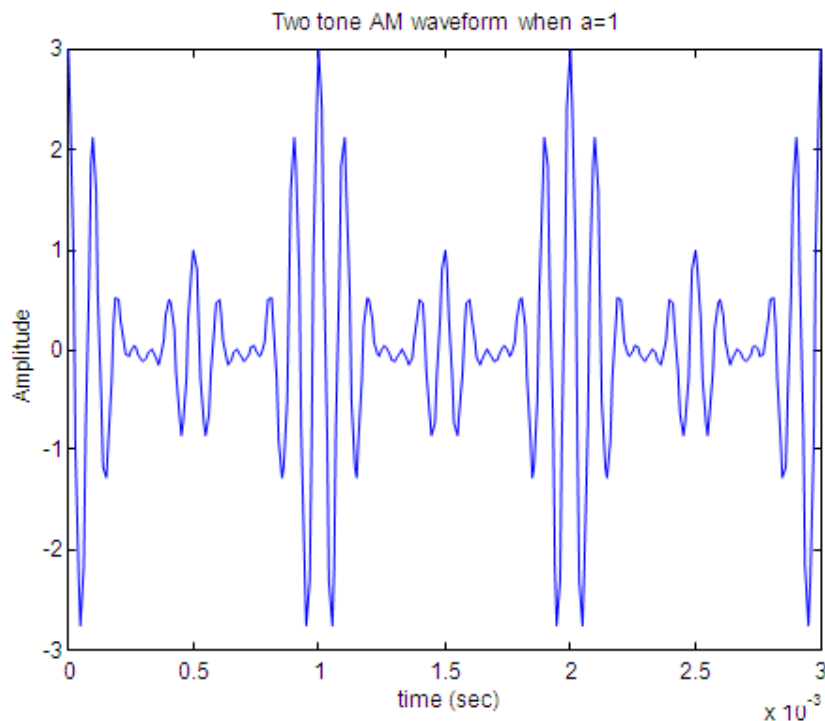


Figure 13.19

The experimental setup is shown below.

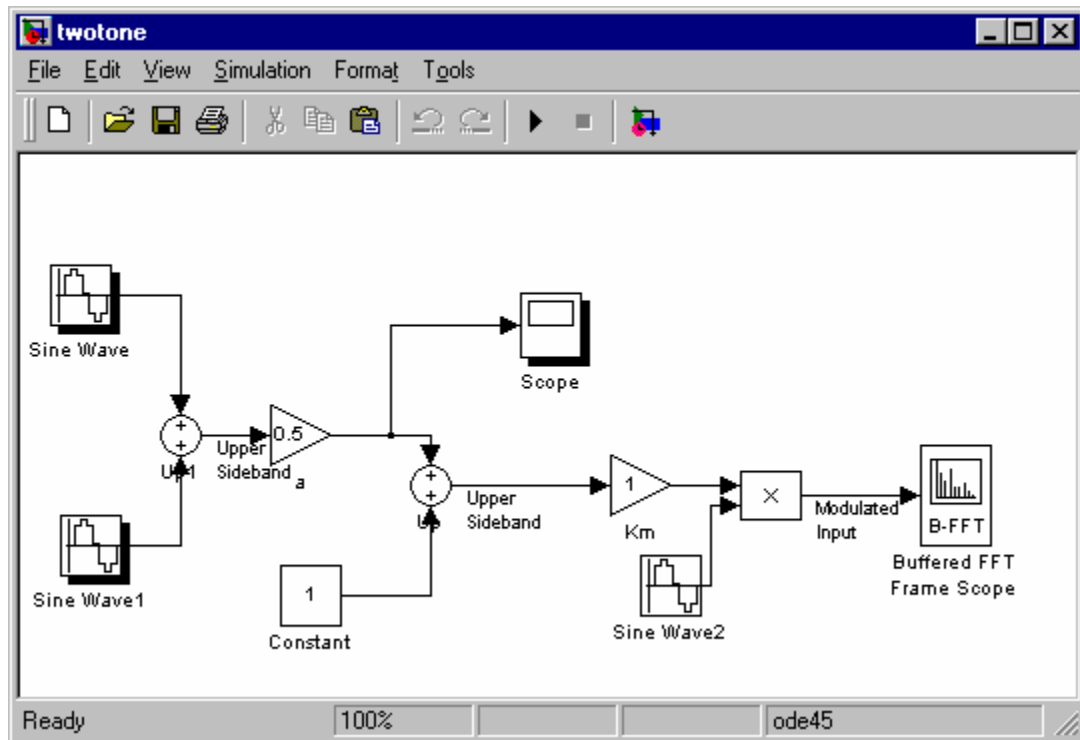


Figure 13.20

The two-tone waveform before being amplitude modulated.

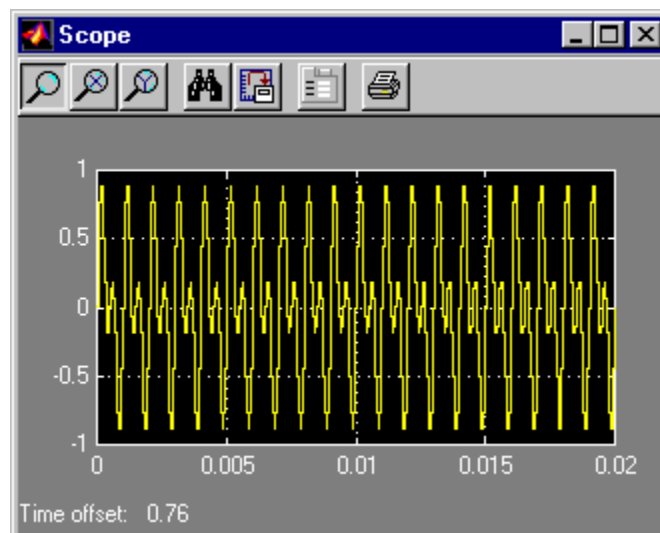


Figure 13.21

The two-tone signal is amplitude modulated using the same block model discussed in the previous section. The output spectrum is shown in figure. In this case the signals of 1 kHz and 2 kHz are modulated by a 10kHz carrier. The output spectrum is shown in figure

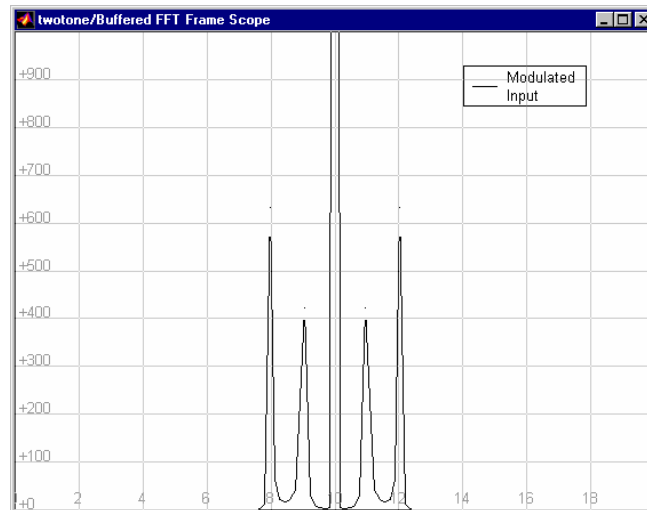


Figure 13.21

The result from the experiment was shown. The highest peak is at the carrier frequency as in the theoretical result. But there were differences on the sidebands. In the figure from MATLAB, both frequencies in the sidebands have the same magnitude, but from the experiment, the components at 9000Hz and 11000Hz have higher magnitude than the components at 8000Hz and 12000 Hz. There're also many small peaks of about 1000Hz apart in the experiment result. This might come from the incorrectly calibrated multiplier.

The final experiment in this section is to change the carrier frequency and the modulating frequency. When the carrier frequency increases, the spectrum of the modulated signal is expected to have the two sidebands centered at the new carrier frequency. And when one of the two modulating signals changes in frequency, the spectrum of the output signal should have two components move away from their original positions according to the change in frequency. The result from the experiment was shown. Both change in carrier frequency and modulating frequency is shown.

III. Single Sideband Modulation

The DSB-SC signal occupies twice the space necessary than required for holding the information. Therefore, by chopping off one part of the DSBSC, more signal transmission can be achieved. Filtering the DSBSC gives the output as either a LSB (Lower side band) or a USB (Upper side band). The simulation set up for the SSB signal is shown in figure below

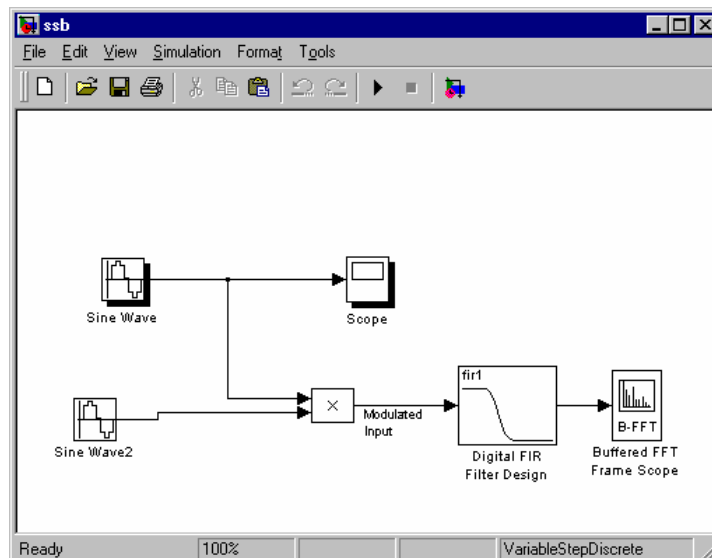


Figure 13.22

The output is going to be a side band. The output of this setup before and after the Filtering is shown in figures and figure. It can be noted that the output of the SSB signal before filtering has the higher order frequency components which are eliminated by the filter.

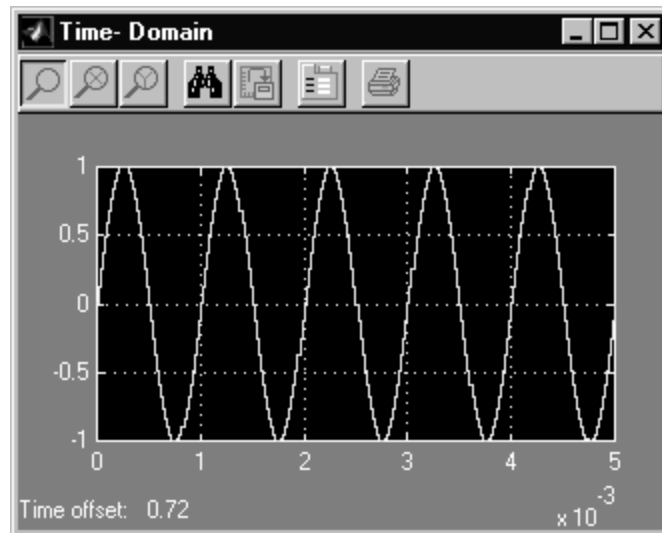


Figure 13.23

Instead of using a filter, the same task can be achieved by using a phase shifter and summer in conjunction with the existing circuit. Operating the summer as an adder causes the USB to be produced. If the summer is operated as an inverter, then, the LSB will be retained.

Without filtering

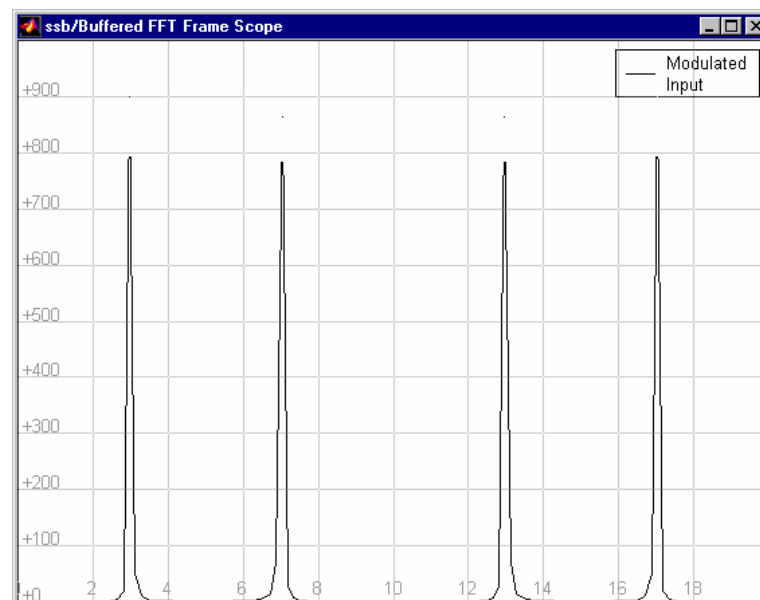


Figure 13.24

After filtering the higher order components are removed and we get a wave form of the form shown in figure

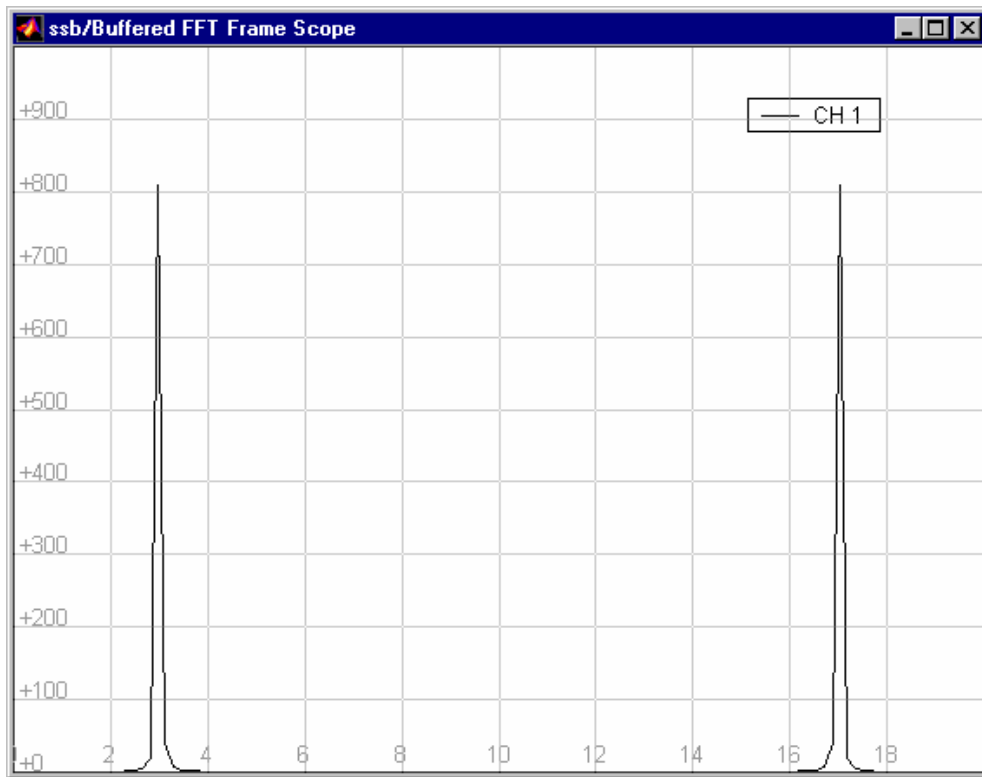


Figure 13.25

IV. Phase Shift SSB Modulation

Figure shows the experimental setup for the Phase Shift SSB Modulation. The signal consists of four input sine waves.

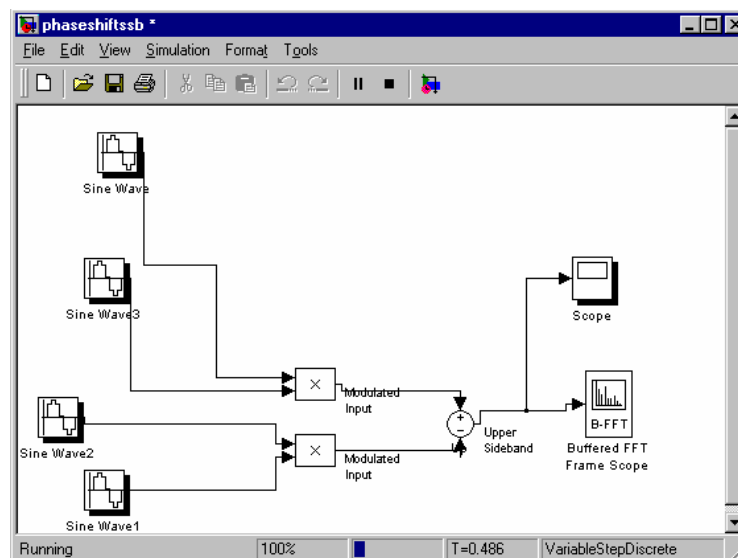


Figure 13.26

The output of the difference block in both the time domain and the frequency domain is of importance to us.

When the sign is $+$, it represents the lower side band and the wave form for $++$ represents the upper-side bands respectively. The output spectrum is shown in figure.

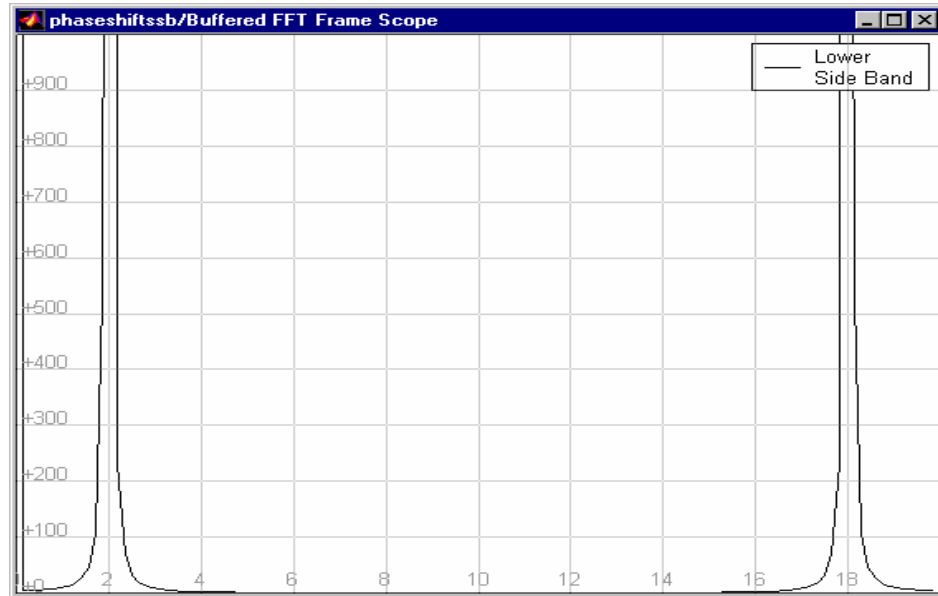


Figure 13.27

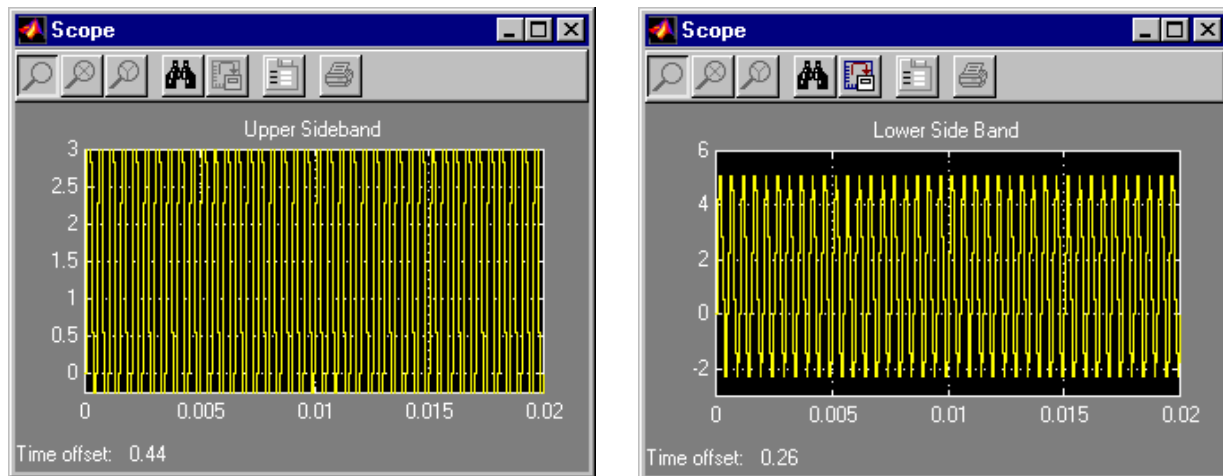


Figure 13.28

Conclusion

We learnt how to operate the spectrum analyzer, oscilloscope and the function generator to generate and view different waveforms. We also performed the different modulation schemes – DSBSC, AM and SSB. We conclude that the DSBSC modulating system is better as no power is lost in the carrier. SSB permits more of the information to be transmitted over the same channel by chopping off the duplicate sideband.

POST LAB

1. If message and carrier signal is a square and sine wave having frequency 1KHz and 10KHz respectively. Then Sketch the spectrum of modulated signal?
2. Draw simulink block diagram of given spectrum?

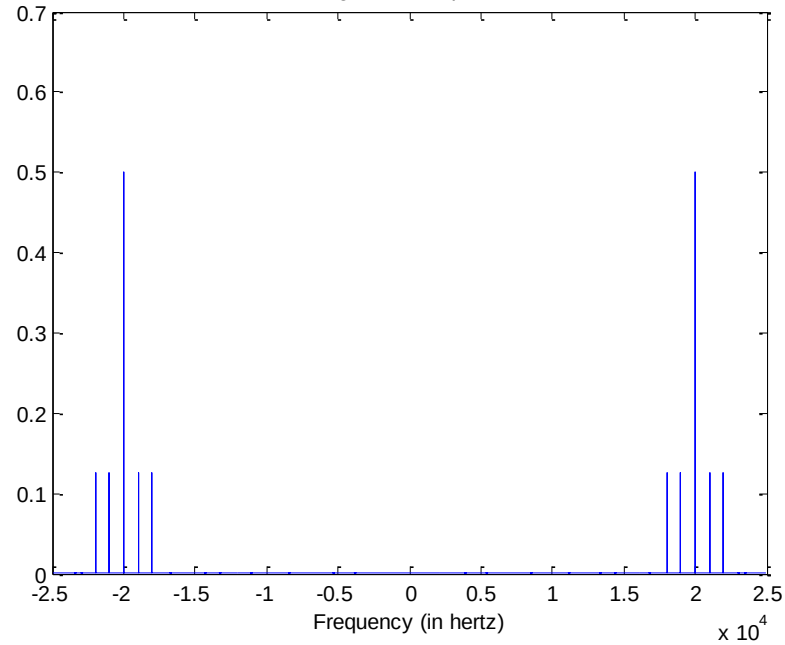


Figure 13.29

Experiment # 14

Introduction to Amplitude Modulation (MATLAB Implementation)

Objective

- To analyze the spectrum, in time and frequency domain, of Amplitude Modulation.

In this first part of the lab we will focus on a couple of simple examples and plot their spectrum, in time and in frequency domain. In second part of this lab we will write the code for Amplitude modulation with carrier and suppress carrier and then focus on two tone modulation and at the end of this lab we will write a code for single side band.

Sketch the time and frequency domain representations (magnitude only) of the following

A. $\cos 2\pi ft$ $f = 1\text{kHz}$

The time and frequency domain of the input signal is shown as below.

CODE:

```
% Time specifications:
Fs = 10000;
dt = 1/Fs;
StopTime = 0.5;
t = (0:dt:StopTime-dt)';
N = size(t,1);

Fc = 1000;
x = cos(2*pi*Fc*t);

subplot(2,1,1)
plot(t,x);
axis([0 1/100 -1 1]);
xlabel('Time');
ylabel('Magnitude')
% Fourier Transform:
X = fftshift(fft(x));
% Frequency specifications:
dF = Fs/N;
f = -Fs/2:dF:Fs/2-dF;
% Plot the spectrum:
subplot(2,1,2)
plot(f,abs(X)/N);
xlabel('Frequency (in hertz)');
ylabel('Magnitude')
```

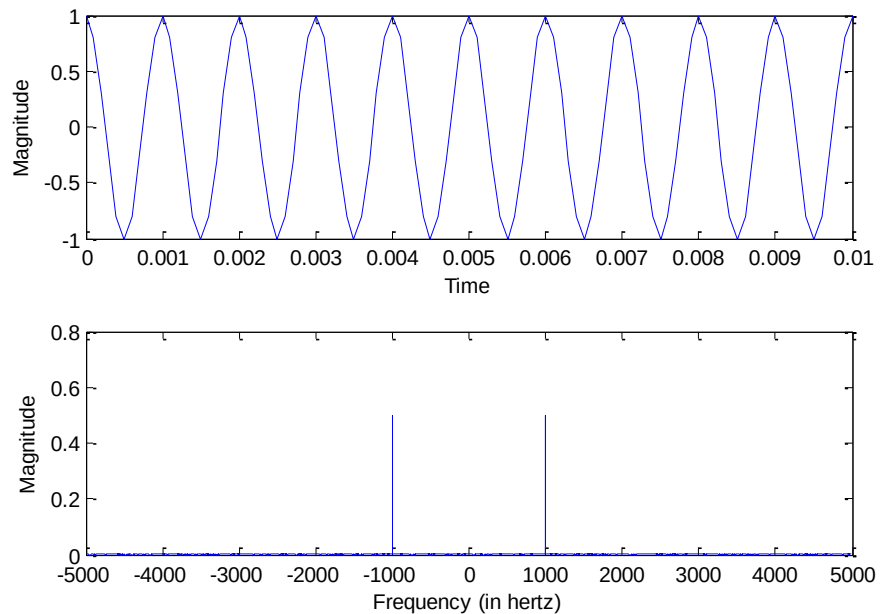


Figure 14.1 Spectrum of $\cos 2000\pi t$

B. Square wave period = 1msec, amplitude = 1v

```
Fs = 1000000;
dt = 1/Fs;
StopTime = 0.5;
t = (0:dt:StopTime-dt)';
N = size(t,1);

Fc = 1000;
x = SQUARE(2*3.14*Fc*t);

subplot(2,1,1)
plot(t,x);
axis([0 1/200 -2 2]);
xlabel('Time');
ylabel('Magnitude');
%% Fourier Transform:
X = fftshift(fft(x));
%% Frequency specifications:
dF = Fs/N;
f = -Fs/2:dF:Fs/2-dF;
%% Plot the spectrum:
subplot(2,1,2)
plot(f,abs(X)/N);
axis([-100000 100000 0 0.5]);
xlabel('Frequency (in hertz)');
ylabel('Magnitude');
```

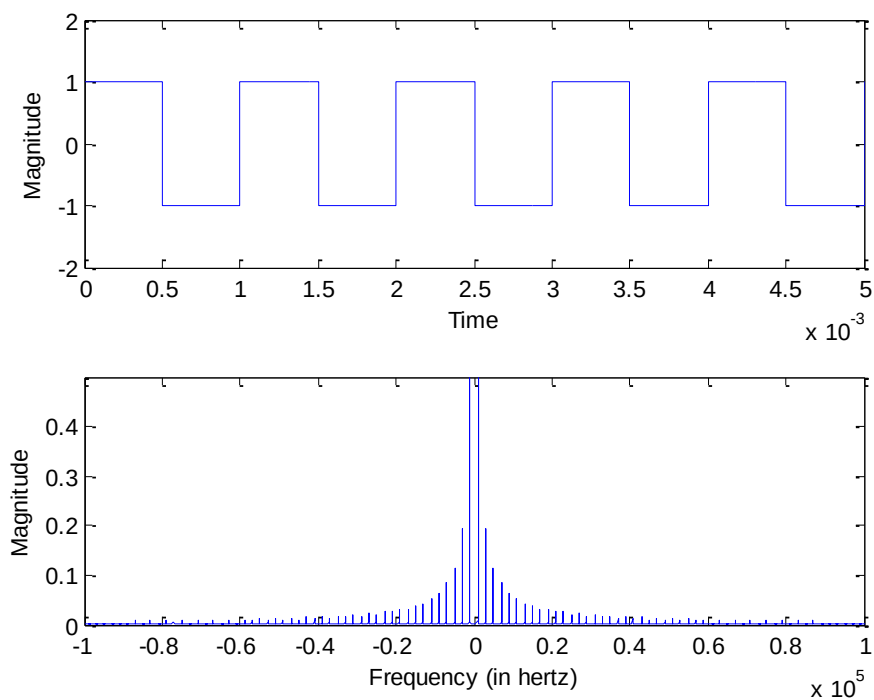


Figure 14.2 Spectrum of Square Wave (1000 Hz)

C. $\cos^2(2\pi ft)$ $f = 1\text{kHz}$

```
Fs = 30000;
dt = 1/Fs;
StopTime = 0.5;
t = (0:dt:StopTime-dt)';
N = size(t,1);

Fc = 1000;
x = cos(2*pi*Fc*t);
x=x.*x;

subplot(2,1,1)
plot(t,x);
xlabel('Time');
ylabel('Magnitude');
axis([0 1/100 -1 1]);
X = fftshift(fft(x));
dF = Fs/N;
f = -Fs/2:dF:Fs/2-dF;
subplot(2,1,2)
plot(f,abs(X)/N);
axis([-5000 5000 0 0.75]);
zoom on
xlabel('Frequency (in hertz)');
ylabel('Magnitude');
```

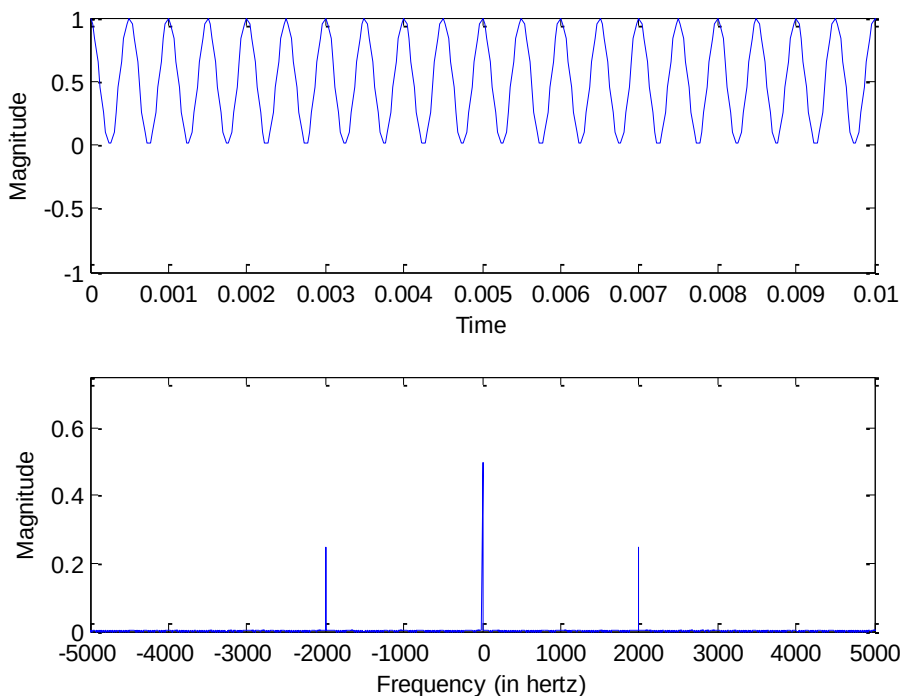


Figure 14.3 Spectrum of $\cos^2(2000\pi t)$

A carrier $\cos 2\pi (5000)t$ is modulated by a single tone $\cos 2\pi (300)t$

A. Double side-band – suppressed carrier modulation

```
Fs = 30000;
dt = 1/Fs;
StopTime = 0.5;
t = (0:dt:StopTime-dt)';
N = size(t,1);
Fc1 = 300;
x1 = cos(2*pi*Fc1*t);
Fc2 = 5000;
x2 = cos(2*pi*Fc2*t);
x=x1.*x2;
subplot(2,1,1)
plot(t,x);
axis([0 1/100 -1 1]);
xlabel('Time');
ylabel('Magnitude');

X = fftshift(fft(x));
dF = Fs/N;
f = -Fs/2:dF:Fs/2-dF;
subplot(2,1,2)
plot(f,abs(X)/N);
axis([-6000 6000 0 0.5]);
xlabel('Frequency (in hertz)');
ylabel('Magnitude');
```

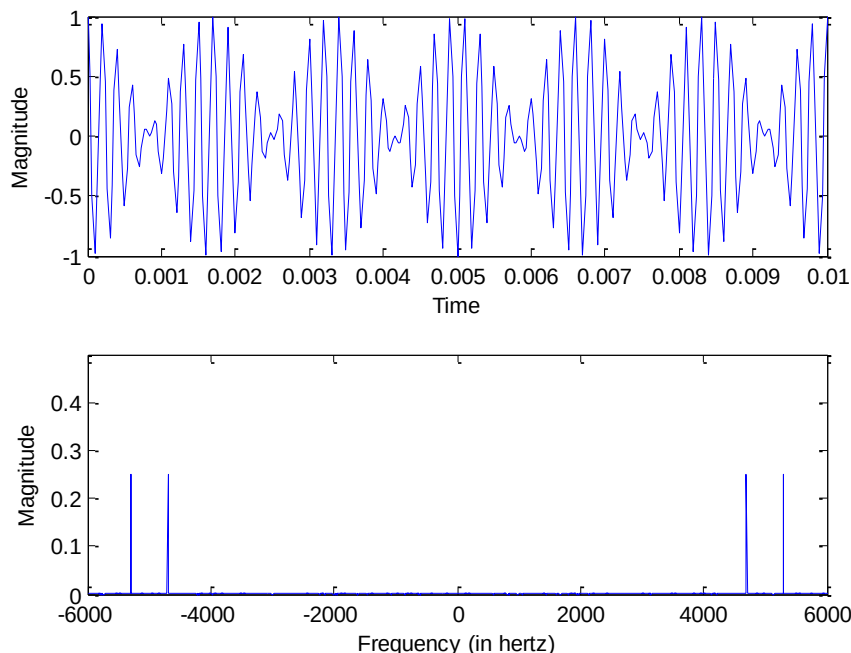


Figure 14.4 Spectrum of AM-SC (DSB)

B. Double side-band – with carrier modulation

```
Fs = 30000;
dt = 1/Fs;
StopTime = 0.5;
t = (0:dt:StopTime-dt)';
N = size(t,1);
Fc1 = 300;
x1 = cos(2*pi*Fc1*t);
Fc2 = 5000;
x2 = cos(2*pi*Fc2*t);
x=(1+x1).*x2;
subplot(2,1,1)
plot(t,x);
axis([0 1/100 -2 2]);
xlabel('Time');
ylabel('Magnitude');

X = fftshift(fft(x));
dF = Fs/N;
f = -Fs/2:dF:Fs/2-dF;
% Plot the spectrum:
subplot(2,1,2)
plot(f,abs(X)/N);
xlabel('Frequency (in hertz)');
ylabel('Magnitude');
```

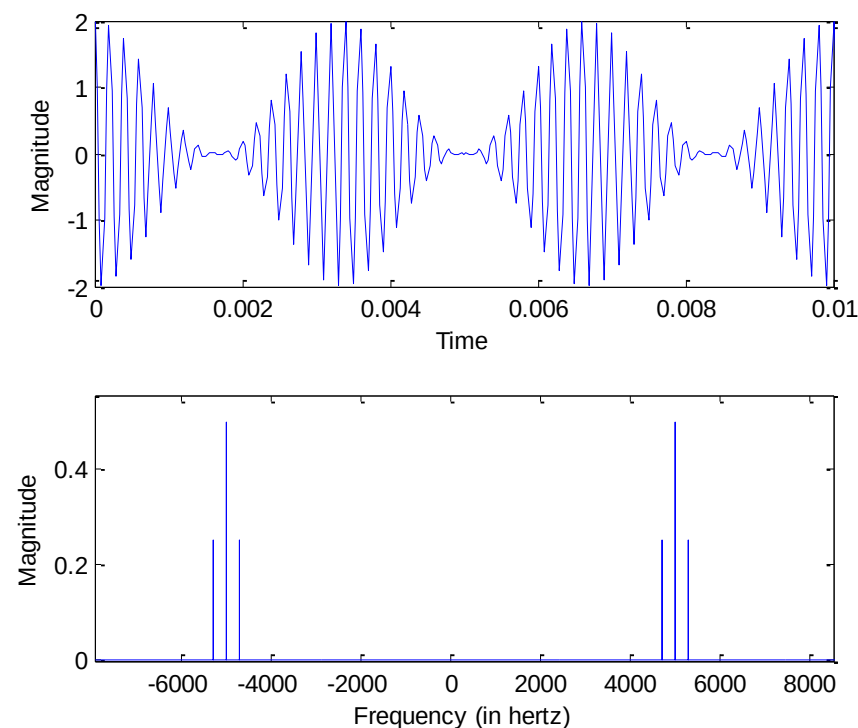


Figure 14.5 Spectrum of AM-WC ($\mu = 1$)

C. 50% AM modulation (modulation index = 0.5)

```

Fs = 30000;
dt = 1/Fs;
StopTime = 0.5;
t = (0:dt:StopTime-dt)';
N = size(t,1);
Fc1 = 300;
x1 = cos(2*pi*Fc1*t);
Fc2 = 5000;
x2 = cos(2*pi*Fc2*t);

x=(1+0.5*x1).*x2;
subplot(2,1,1)
plot(t,x);
axis([0 1/100 -2 2]);
xlabel('Time');
ylabel('Magnitude');

X = fftshift(fft(x));

dF = Fs/N;
f = -Fs/2:dF:Ff/2-dF;
subplot(2,1,2)
plot(f,abs(X)/N);
zoom on
xlabel('Frequency (in hertz)');
ylabel('Magnitude');

```

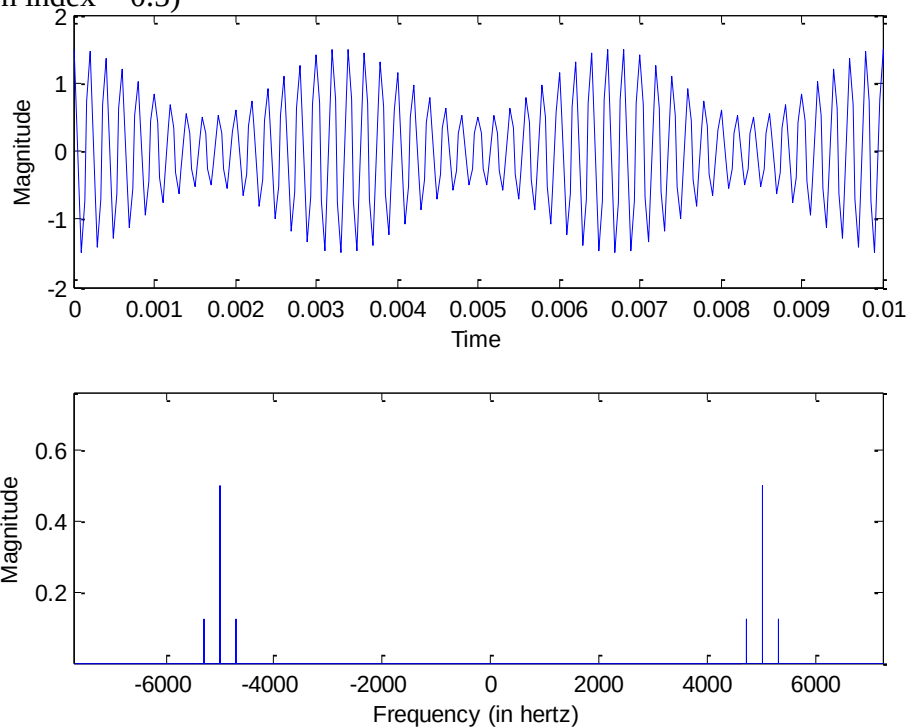


Figure 14.6 Spectrum of AM-WC ($\mu = 0.5$)

Two Tone (1 kHz and 2 kHz) modulating a carrier of 10 kHz.

A. Double side band suppressed carrier

```

Fs = 43000;
dt = 1/Fs;
StopTime = 0.5;
t = (0:dt:StopTime-dt)';
N = size(t,1);
Fc1 = 1000;
x1 = cos(2*pi*Fc1*t);
Fc2 = 2000;
x2 = cos(2*pi*Fc2*t);
Fc3 = 10000;
x3 = cos(2*pi*Fc3*t);

x=(x1+x2).*x3;
subplot(2,1,1)
plot(t,x);
xlabel('Time');
ylabel('Magnitude');
axis([0 1/100 -2 2]);

X = fftshift(fft(x));
dF = Fs/N;
f = -Fs/2:dF:Ff/2-dF;
subplot(2,1,2)
plot(f,abs(X)/N);

xlabel('Frequency (in hertz)');
ylabel('Magnitude');

```

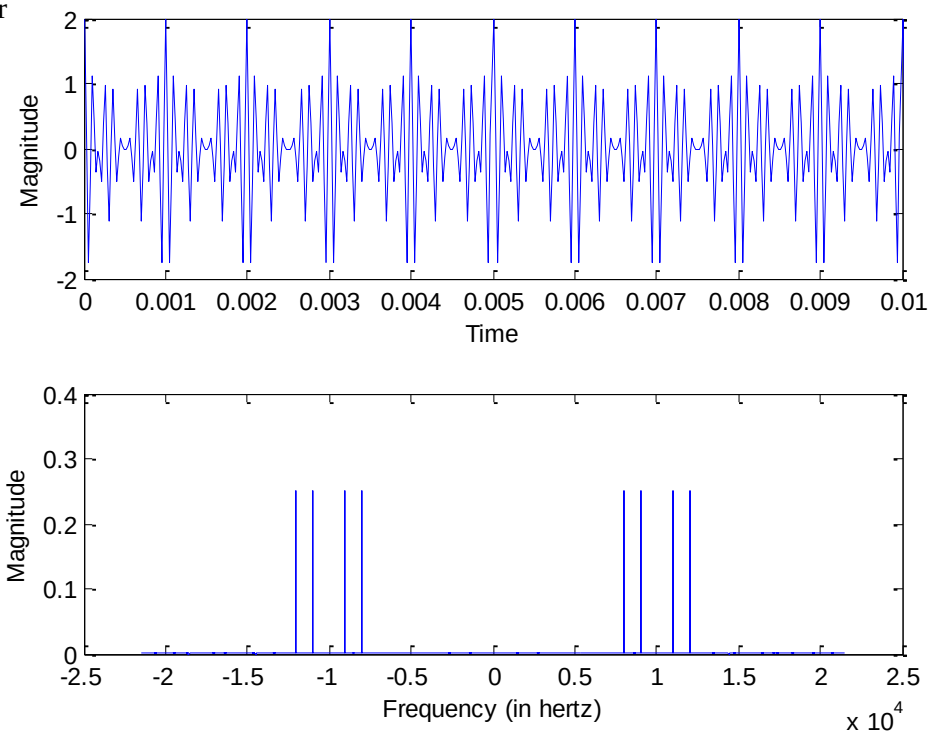


Figure 14.7 Spectrum of Two Tone Modulation (SC)

B. Double side band with carrier - 100% AM modulation (modulation index = 1)

```
Fs = 40000;
dt = 1/Fs;
StopTime = 1;
t = (0:dt:StopTime-dt)';
N = size(t,1);
Fc1 = 1000;
x1 = cos(2*pi*Fc1*t);
Fc2 = 2000;
x2 = cos(2*pi*Fc2*t);
Fc3 = 10000;
x3 = cos(2*pi*Fc3*t);
```

```
x=(1+(x1+x2)).*x3;
subplot(2,1,1)
plot(t,x);
xlabel('Time');
ylabel('Magnititude');
axis([0 1/100 -2 2]);

X = fftshift(fft(x));
dF = Fs/N;
f = -Fs/2:dF:Fs/2-dF;
subplot(2,1,2)
plot(f,abs(X)/N);
xlabel('Frequency (in hertz)');
ylabel('Magnititude');
```

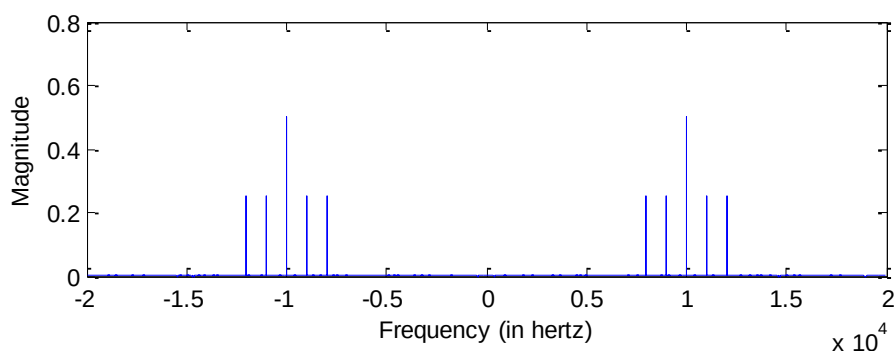
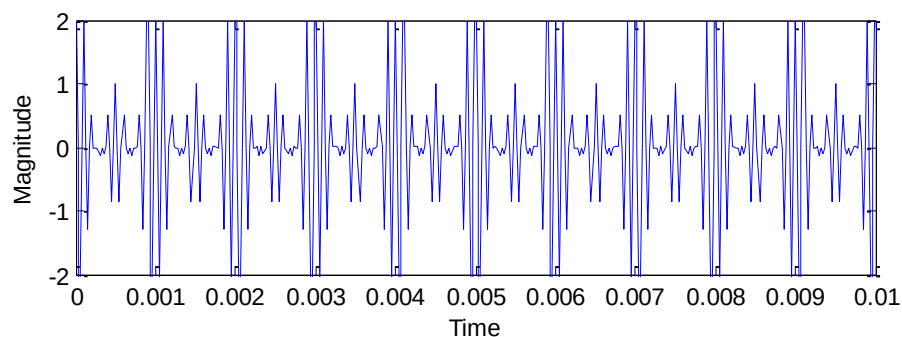


Figure 14.8 Spectrum of Two Tune Modulation ($\mu = 1$)

50% AM modulation (modulation index = 0.5)

Two tune Modulation, Modulating index 0.5

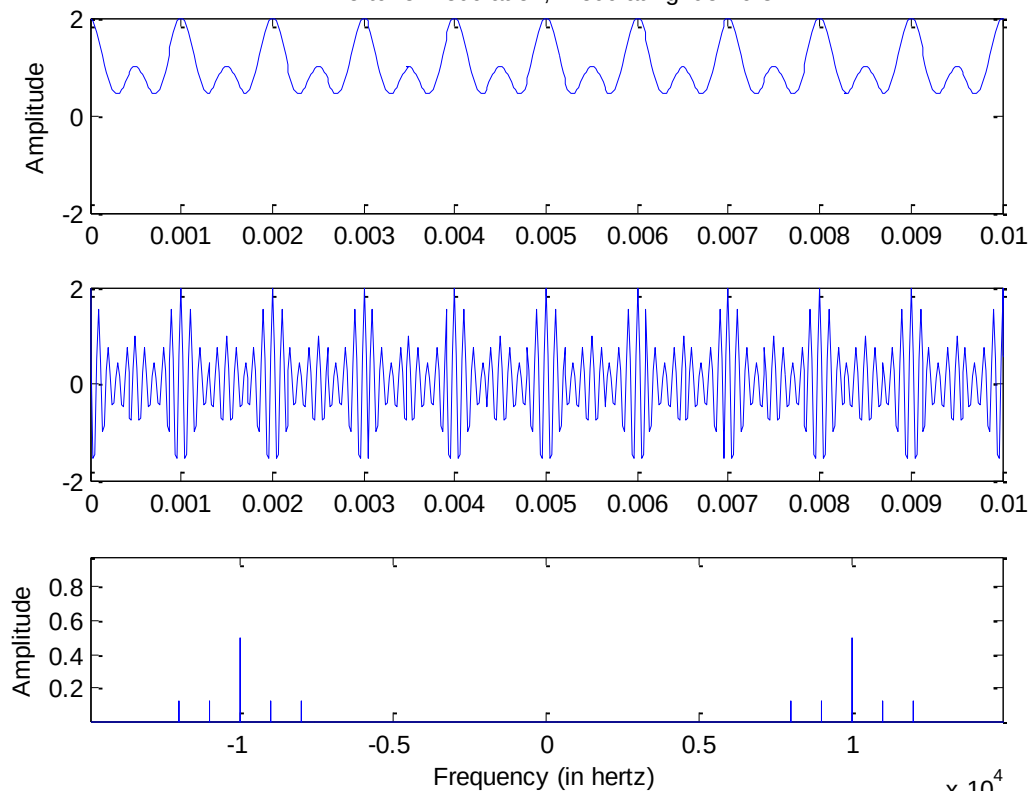


Figure 14.9 Spectrum of Two Tune Modulation ($\mu = 0.5$)

Single Side Band Modulation (lower side band)

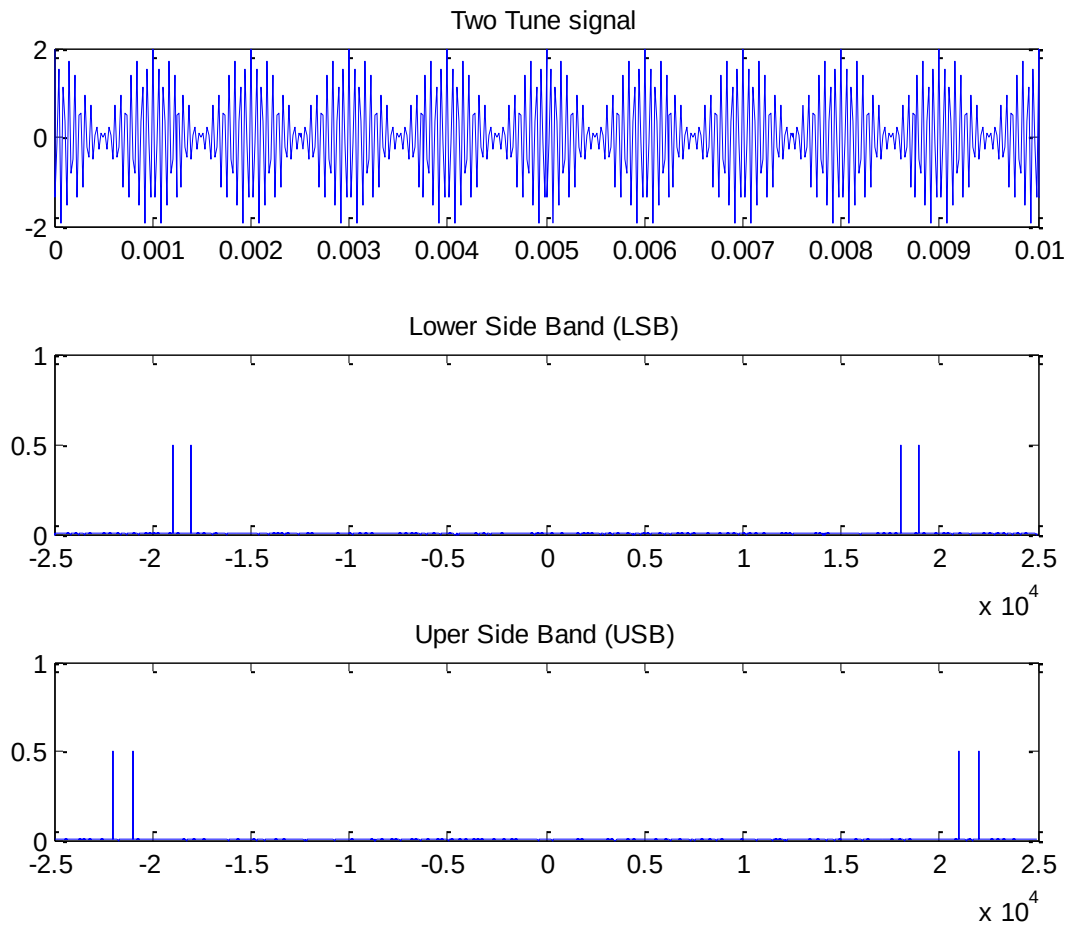


Figure 14.8 Spectrum of SSB

POST LAB

1. Write a matlab code to sketch the spectrum of modulated signal, if
Message signal = $\cos(2000\pi t) + 3\cos(3000\pi t) + 2\cos(4000\pi t)$
Carrier signal = Square wave (frequency = 20KHz)
2. Why we prefer SSB over DSB?

Appendix A: Lab Evaluation Criteria

1. Experiments and their reports	50%
a. Experiment	60%
b. Lab report	40%
2. Quizzes (3-4)	15%
3. Final evaluation	35%
a. Project Implementation	60%
b. Project report and quiz	40%

Notice:

Copying and plagiarism of lab reports is a serious academic misconduct. First instance of copying may entail ZERO in that experiment. Second instance of copying may be reported to DC. This may result in awarding FAIL in the lab course.

Appendix B: Safety around Electricity

In all the Electrical Engineering (EE) labs, with an aim to prevent any unforeseen accidents during conduct of lab experiments, following preventive measures and safe practices shall be adopted:

- Remember that the voltage of the electricity and the available electrical current in EE labs has enough power to cause death/injury by electrocution. It is around 50V/10 mA that the “cannot let go” level is reached. “The key to survival is to decrease our exposure to energized circuits.”
- If a person touches an energized bare wire or faulty equipment while grounded, electricity will instantly pass through the body to the ground, causing a harmful, potentially fatal, shock.
- Each circuit must be protected by a fuse or circuit breaker that will blow or “trip” when its safe carrying capacity is surpassed. If a fuse blows or circuit breaker trips repeatedly while in normal use (not overloaded), check for shorts and other faults in the line or devices. Do not resume use until the trouble is fixed.
- It is hazardous to overload electrical circuits by using extension cords and multi-plug outlets. Use extension cords only when necessary and make sure they are heavy enough for the job. Avoid creating an “octopus” by inserting several plugs into a multi-plug outlet connected to a single wall outlet. Extension cords should ONLY be used on a temporary basis in situations where fixed wiring is not feasible.
- Dimmed lights, reduced output from heaters and poor monitor pictures are all symptoms of an overloaded circuit. Keep the total load at any one time safely below maximum capacity.
- If wires are exposed, they may cause a shock to a person who comes into contact with them. Cords should not be hung on nails, run over or wrapped around objects, knotted or twisted. This may break the wire or insulation. Short circuits are usually caused by bare wires touching due to breakdown of insulation. Electrical tape or any other kind of tape is not adequate for insulation!
- Electrical cords should be examined visually before use for external defects such as: Fraying (worn out) and exposed wiring, loose parts, deformed or missing parts, damage to outer jacket or insulation, evidence of internal damage such as pinched or crushed outer jacket. If any defects are found the electric cords should be removed from service immediately.
- Pull the plug not the cord. Pulling the cord could break a wire, causing a short circuit.
- Plug your heavy current consuming or any other large appliances into an outlet that is not shared with other appliances. Do not tamper with fuses as this is a potential fire hazard. Do not overload circuits as this may cause the wires to heat and ignite insulation or other combustibles.
- Keep lab equipment properly cleaned and maintained.
- Ensure lamps are free from contact with flammable material. Always use lights bulbs with the recommended wattage for your lamp and equipment.
- Be aware of the odor of burning plastic or wire.
- ALWAYS follow the manufacturer recommendations when using or installing new lab equipment. Wiring installations should always be made by a licensed electrician or other qualified person. All electrical lab equipment should have the label of a testing laboratory.
- Be aware of missing ground prong and outlet cover, pinched wires, damaged casings on electrical outlets.

Lab Manual of Analog & Digital Communication

- Inform Lab engineer / Lab assistant of any failure of safety preventive measures and safe practices as soon you notice it. Be alert and proceed with caution at all times in the laboratory.
- Conduct yourself in a responsible manner at all times in the EE Labs.
- Follow all written and verbal instructions carefully. If you do not understand a direction or part of a procedure, ASK YOUR LAB ENGINEER / LAB ASSISTANT BEFORE PROCEEDING WITH THE ACTIVITY.
- Never work alone in the laboratory. No student may work in EE Labs without the presence of the Lab engineer / Lab assistant.
- Perform only those experiments authorized by your teacher. Carefully follow all instructions, both written and oral. Unauthorized experiments are not allowed.
- Be prepared for your work in the EE Labs. Read all procedures thoroughly before entering the laboratory. Never fool around in the laboratory. Horseplay, practical jokes, and pranks are dangerous and prohibited.
- Always work in a well-ventilated area.
- Observe good housekeeping practices. Work areas should be kept clean and tidy at all times.
- Experiments must be personally monitored at all times. Do not wander around the room, distract other students, startle other students or interfere with the laboratory experiments of others.
- Dress properly during a laboratory activity. Long hair, dangling jewelry, and loose or baggy clothing are a hazard in the laboratory. Long hair must be tied back, and dangling jewelry and baggy clothing must be secured. Shoes must completely cover the foot.
- Know the locations and operating procedures of all safety equipment including fire extinguisher. Know what to do if there is a fire during a lab period; “Turn off equipment, if possible and exit EE lab immediately.”

Appendix C: Guidelines on Preparing Lab Reports

Each student will maintain a lab notebook for each lab course. He will write a report for each experiment he performs in his notebook. A format has been developed for writing these lab reports.

Lab Report Format

1. **Introduction:** Introduce area explored in the experiment.
2. **Objective:** What are the learning goals of the experiment?
3. **Design/Measurements:** In your own words write how the experiment is performed. Include the circuit diagram with explanation.
4. **Issues:** Technical issues which were faced during the performance of the experiment and how they were resolved?
5. **Conclusions:** What conclusions can be drawn from experiment?
6. **Application:** Suggest a real world application where this exercise may apply.
7. Answers to post lab questions (if any).

Sample Lab Report:

Introduction

An RC circuit is a first order circuit that utilizes a capacitor as an energy storage element whereas a resistor as an energy wastage element. RC circuits are building blocks of electronic devices and their thorough understanding is important in comprehending advance engineering systems such as transistors and transmission lines.

An RC circuit can be operated with both DC and AC sources. In this lab we study transient response of RC circuits with a square wave as a DC source. During the DC operation of an RC circuit the voltage across the capacitor or the resistor show energy storing (capacitor charging) and dissipating (capacitor discharging via resistor) mechanisms of the circuit. The capacitor charging or discharging curves then lead to determine time constant of the circuit where the time constant signifies time required by the RC circuit to store or waste energy.

Objective:

To study transient response of a series RC circuit

Measurements:

The circuit used for the experiment is shown in Fig. 1.

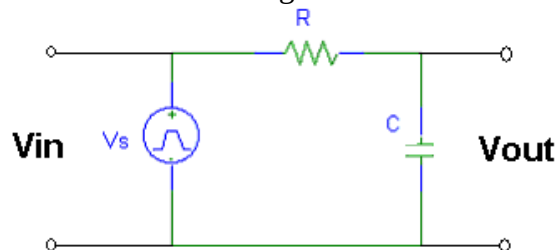


Fig.1. Circuit used in the experiment

Both input (a square wave) and output (voltage across capacitor) waveforms are monitored on an oscilloscope. The capacitor charging is observed during "on" part of the square waveform whereas the capacitor discharging is observed during "off" part of the square waveform (Fig. 2). We measure the time constant from the capacitor charging or discharging curve. While keeping the capacitor value constant, we also measure time constants with various resistor values (Table I).

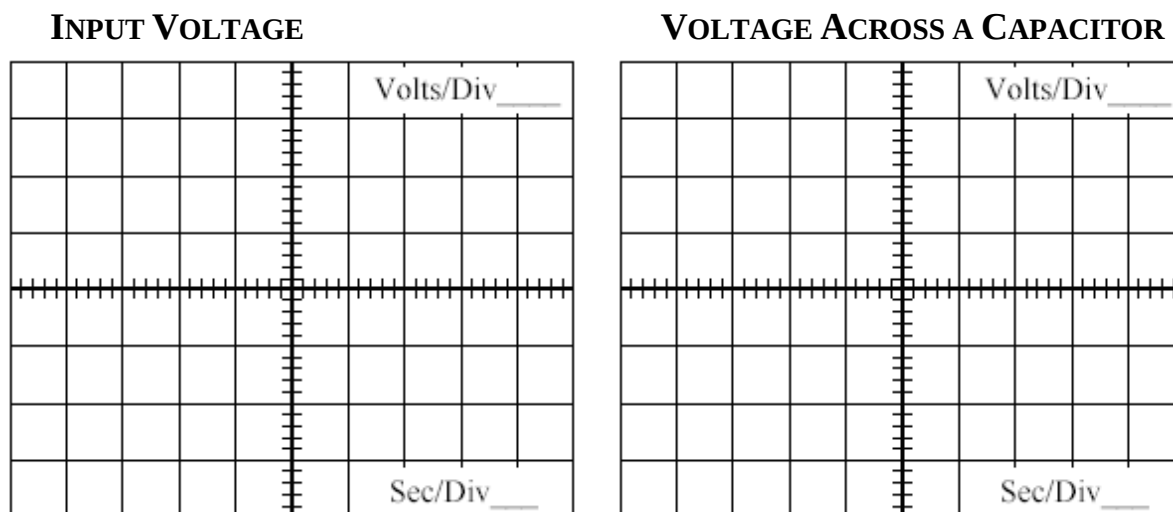


Fig. 2. Input and Output waveforms

TABLE I. Time constant as a function of the resistor values

Resistance (Nominal)	270 Ω	330 Ω	470 Ω	1 k Ω	2.2 k Ω	3.3 k Ω
Time constant (Calculated)						
Time constant (Measured)						

Conclusions:

From the measurements following conclusions can be drawn:

- The capacitor charging and discharging curves are exponential.
- The time constant is directly proportional to the resistor value.

Both of the above conclusions are also easily verifiable by solving differential equation for the RC circuit.

Applications:

An RC circuit can be employed for a camera flash. The capacitor discharges through the flash light during a picture taking event.