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Experiment No. 1

Aim:

To simulate Double side band suppressed carrier (DSBSC) modulation scheme using MATLAB

Software Used:

MATLAB 2016b

Theory:

DSB-SC is an amplitude modulated wave transmission scheme in which only sidebands are transmitted and the carrier isn't transmitted as it gets suppressed.

The carrier doesn't contain any information & its transmission results in loss of power. Hence this system employs simple multiplication process. The output here maybe written as

$$x(t) = m(t)\,c(t)$$
$$= A_c m(t) \cos \omega_c t$$

Assume m(t) to be a sinusoidal signal. At any given instant the amplitude of modulated signal can be seen to have the value $A_c m(t)$. But as $A_c m(t)$ is a varying function of time, the high frequency carrier is thus scaled in amplitude within an envelope created by the function $A_c m(t)$. This method of modulation is used in P2P two-way communication, Analog TV & FM Broadcasting.

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Discussion:

Double side band - Suppressed Carrier is a linear amplitude modulation scheme the idea behind which is to transmit the side bands and not the carrier wave as it gets suppressed.

For the sake of discussion we'll follow the conventions given below -

$m(t)$ is message signal

$c(t)$ is the carrier signal

$x_i(t)$ is modulated wave, i the type of modulation

$r(t)$ is the received signal at the receiving filter

$y(t)$ is final output or demodulated wave or received msg.

The idea behind the modulation is the ease of radiation. Suppose we want to transmit a signal of 4kHz. This would require an antenna of about 75 km which is practically impossible.

Due to modulation transmission of such signals take place at higher frequencies thereby making height of the antenna considerably smaller.

Moreover noise is more prelevant in certain regions of the spectrum. So we do take care that the transmission takes place at the frequency less susceptible to noise.

With multiplexing multiple baseband signals centred on different carrier frequencies can be transmitted over the same line simultaneously.
Sometimes we have equipment on which it is more convenient to process signals in a specific frequency band in comparison to others.

We begin by taking the Amplitude of message and carrier signal A_m & A_c respectively, f_m & f_c being the frequencies of the signals.
Our message signal is a cosine signal of amplitude A_m & frequency f_m while carrier signal is a ~~cos~~ sine signal of frequency f_c and amplitude A_c.
hence this step accounts for generation of carrier and message signal.

Next we multiply both of them, message and carrier signal. It gives us modulated signal $x(t)$. The type of multiplication performed is element wise.

At the receiving end we multiply the signal with a sinusoidal of amplitude 1 & frequency f_c.
For filtering process we used a butter filter

A butterworth filter is a type of signal processing filter that is designed to have a frequency response as flat as possible.

Syntax of butterworth filter function is given by

$$[b,a] = \text{butter}(n, Wn)$$

It returns transfer function coefficients of n^{th} order lowpass digital Butterworth filter with normalized cutoff frequency Wn

$$y = \text{filter}(b, a, x)$$

The filter inputs data x using a rational transfer function defined by numerator & denominator coefficients b and a. This returns the filtered data which in our case is demodulated signal.

We try to get the response for four different cases:

(1) Single sinusoidal Input, Ideal detection
(2) Single sinusoidal Input, Phase shift distortion
(3) Single Sinusoidal Input, Frequency mismatch
(4) Multiple Sinusoidal Input, Ideal recovery

Case I : Single Sinusoid Ideal detection

In this case, the carrier frequency & phase is known clearly at the receiving end. This is the ideal case where

$$y(t) = A\,m(t)$$

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Case II: Single Sinusoid, Phase Shift Distortion

For the sine signal $\sin(2\pi f_c t + \phi)$ when ϕ is not equal to zero ie. there's a Phase mismatch. received signal $r(t)$,

$$r(t) = A_c \sin w_c t \, m(t) * \sin(w_c t + \phi)$$
$$= \frac{A_c}{2} m(t) \left[\cos\phi - \cos(2w_c t + \phi) \right]$$

After low pass filtering

$$y(t) = \frac{A_c}{2} m(t) \cos\theta$$

At certain condition where $\theta = \frac{\pi}{2}$, $y(t) = 0$. Hence $\cos\theta$ becomes an attenuation factor. This is known as phase shift distortion.

Case III: Single Sinusoid, Frequency Mismatch

In case of frequency mismatch of δw

$$r(t) = A_c m(t) \sin w_c t \sin(w_c t + \delta w t)$$
$$= \frac{A_c}{2} m(t) \left[\cos(\delta w t) - \cos(2w_c t + \delta w t) \right]$$

After low pass filtering

$$y(t) = \frac{A_c}{2} m(t) \cos(\delta w t)$$

Instead of receiving only $m(t)$ we'll receive a waveform product of two functions of time.

closer to $\pi/2$

As $\delta\omega$ typically attains a very large value, m(t) will vary in an envelope defined by sine function. Such a varying distortion is known as Warmbling Effect.

Case IV: Multiple Sinusoidal Input, Ideal Recovery

Multiple Sinusoid Input in case of Ideal Recovery works in pretty much same manner as that of single sinusoid. The only difference is our msg signal is sum of sinusoids of different frequencies. Given there is no phase or frequency mismatch, the demodulated message signal is pretty much same as original message signal.

Result

We simulated Double Side Band - Suppressed Carrier (DSB-SC) using MATLAB and saw the results for various cases.

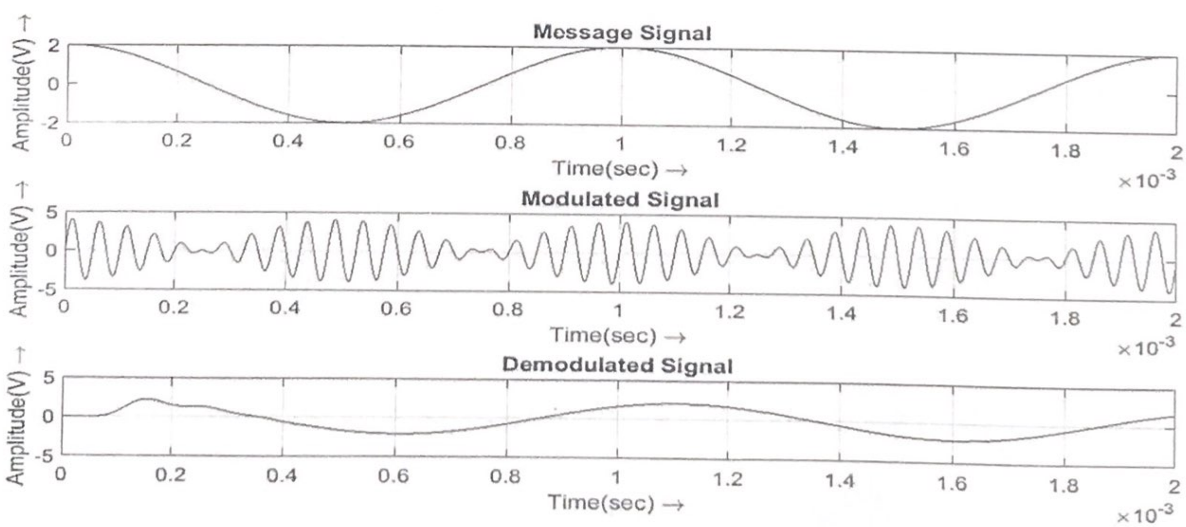
Experiment No.1

Aim: To simulate Double Side Band Suppressed Carrier (DSBSC) modulation scheme using MATLAB.

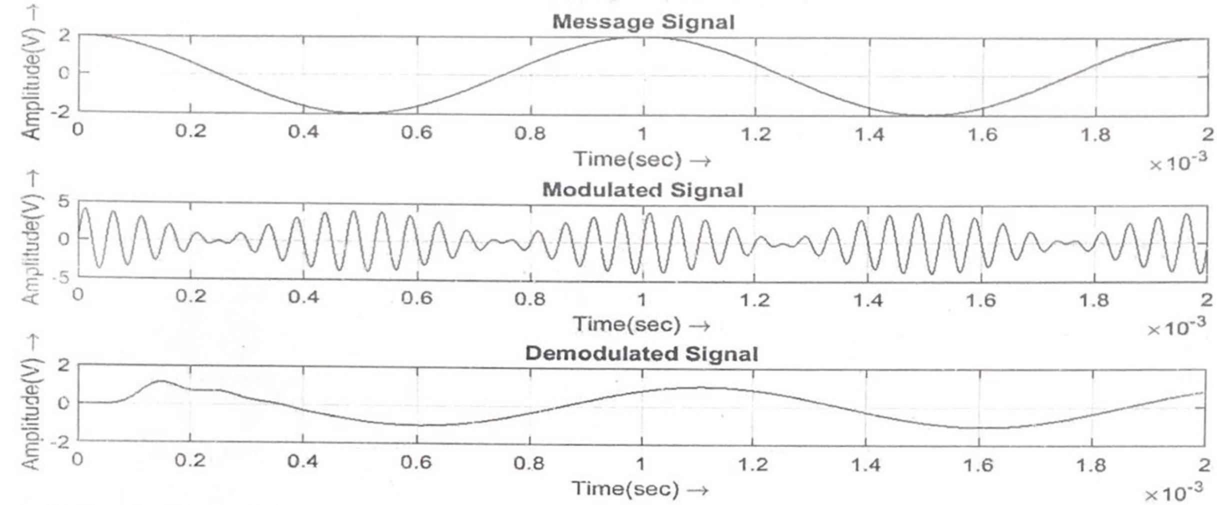
Code:

```
clear all
close all
clc
Am = input('Message Signal Amplitude: ');
fm = input('Message Signal Frequency: ');
Ac = input('Carrier Signal Amplitude: ');
fc = input('Carrier Signal Frequency: ');
t = 0 : 1/(50*fc) : 2/fm;
m = Am*cos(2*pi*fm*t);
c = Ac*sin(2*pi*fc*t);
%Modulation
x = m.*c;
%Demodulation
r = x.*sin(2*pi*fc*t);
[n,d] = butter(10, 1/50);
y = filter(n,d,r);
subplot(3,1,1);
plot(t,m, 'k');
grid on;
xlabel('Time(sec) \rightarrow',
'FontSize', 16);
ylabel('Amplitude(V) \rightarrow',
'FontSize', 16);
title('Message Signal',  'FontSize', 16);
set(gca, 'FontSize', 16);
subplot(3,1,2);
plot(t,x, 'k');
grid on;
xlabel('Time(sec) \rightarrow',
'FontSize', 16);
ylabel('Amplitude(V) \rightarrow',
'FontSize', 16);
title('Modulated Signal',  'FontSize',
16);
set(gca, 'FontSize', 16);
subplot(3,1,3);
plot(t,y, 'k');
grid on;
xlabel('Time(sec) \rightarrow',
'FontSize', 16);
ylabel('Amplitude(V) \rightarrow',
'FontSize', 16);
title('Demodulated Signal',  'FontSize',
16);
set(gca, 'FontSize', 16);
```

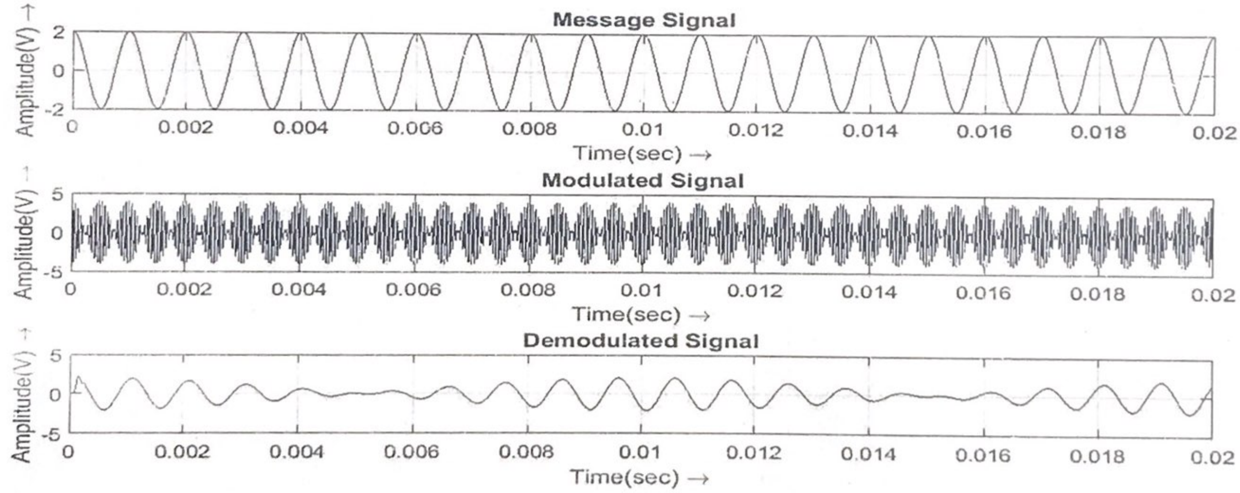
Case 1: Single Sinusoidal Input, Ideal Detection



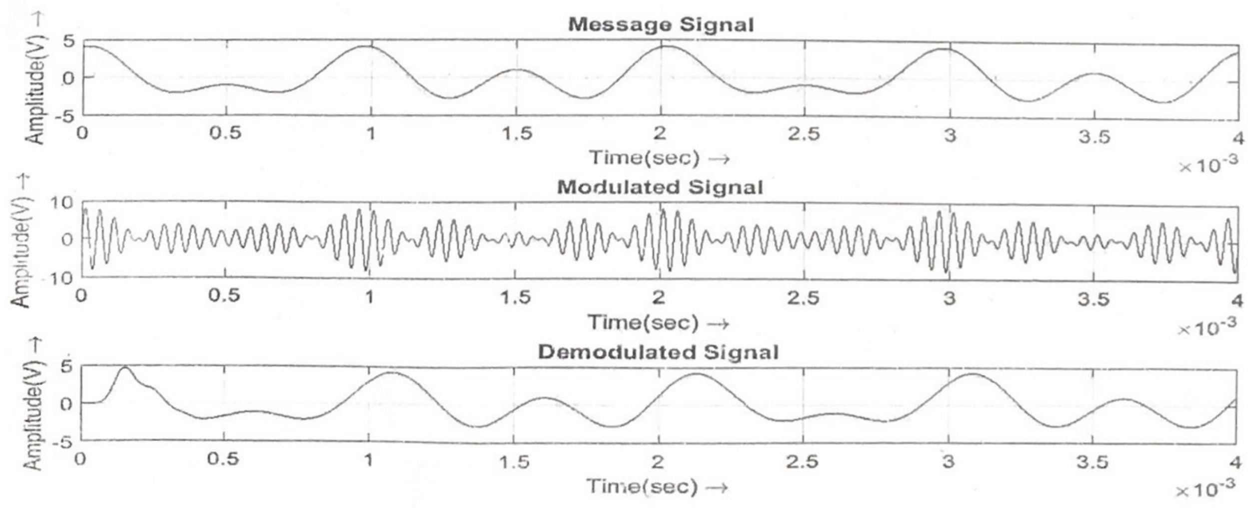
Case 2: Single Sinusoidal Input, Phase Shift Distortion



Case 3: Single Sinusoidal Input, Frequency Mismatch



Case 4: Multiple Sinusoidal Input, Ideal Recovery



Experiment No. 2

Aim:

To simulate Double sideband Full Carrier modulation scheme using MATLAB

Software Used:

MATLAB 2016 b

Theory:

An amplitude modulation scheme where transmission of signal contains a carrier along with two sidebands is termed as Double sideband Full Carrier or simply DSBFC.

In case of DSBFC

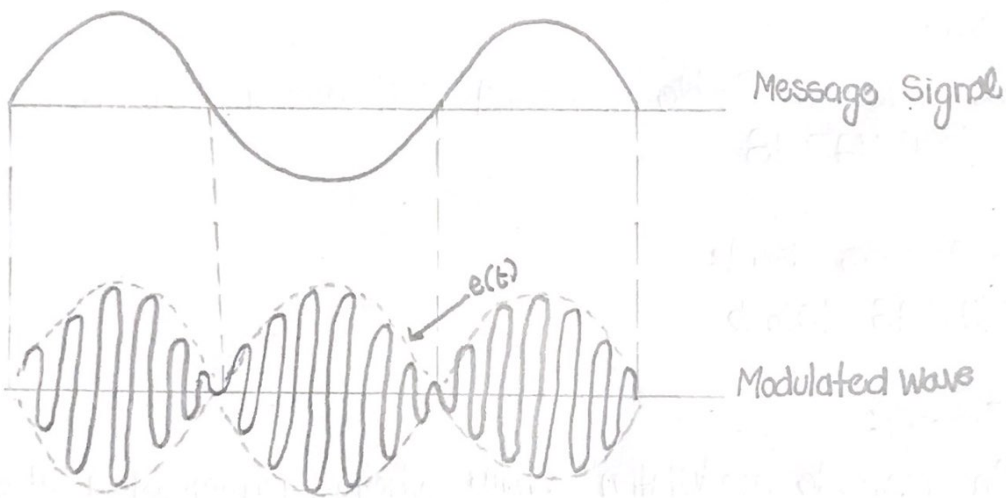
$$x_{DSBFC}(t) = A_c(1+km(t))\cos\omega_c t$$

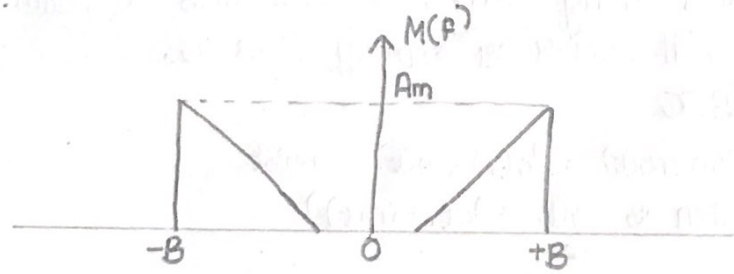
$$\text{written as} \quad e(t) = A_c(1+km(t))$$

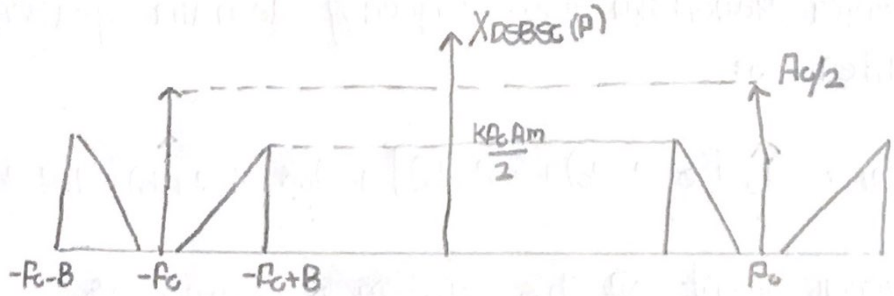
Applying Fourier Transform the frequency domain equation can be written as:

$$X_{DSBFC}(f) = \frac{A_c}{2}\left[\delta(f+f_c)+\delta(f-f_c)\right] + \frac{A_c k}{2}\left[M(f+f_c)+M(f-f_c)\right]$$

The band requirement of this system is same as DSB-SC i.e. 2B. Furthermore, the modulated wave transmission requires some extra power since carrier is also being sent.


Message Signal
e(t)
Modulated Wave


M(f)
Am
-B
0
+B


XDSBSC (f)
Ac/2
KAcAm
2
-fc-B
-fc
-fc+B
fc

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Since this extra power is not carrying any relevant message information, hence DSB-FC is power Inefficient.

Even though the system is both bandwidth & power inefficient, still the easy recover and cheaper demodulation circuits reduce the cost of receiver significantly

DSB-FC is the traditionally accepted modulation scheme for commercial broadcast AM Radio.

Discussion

First of all why DSB-FC when we have DSB-SC?
In case of DSB-SC we require coherent demodulation as the modulated waveform faces a 180° phase change everytime the modulating signal changes its polarity. This is because upon multiplication of carrier and message signal we get a wave proportional to mod of message signal.

Hence we can modify our message signal to always have a magnitude greatter than one. Therefore there won't be any phase shift and demodulation process becomes much easier.

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First of all we generate a message and carrier signal given by $m(t) = A_m \cos\omega_m t$ & $c(t) = A_c \cos\omega_c t$
In case of DSBFC

$$x_{DSBFC}(t) = A_c(1 + Km(t))\cos\omega_c t$$
$$x_{DSBFC}(t) = e(t)\cos\omega_c t$$

where $e(t) = A_c(1 + Km(t))$ can be called the envelop function of modulated waveform.

Scaling factor K ensures that $e(t) \geq 0 \ \forall t$
Envelope $e(t) \propto m(t)$ & $e(t) > 0$ which makes the recovery of $m(t)$ very simple.
However DSB-FC is power inefficient as the carrier transmission requires extra power which is not being used for communication of any useful information.
However this system also ensures easy recovery and cheaper demodulation circuits reduce the cost of receiver inefficiently

While working with DSBFC we come approx a term called modulation index. It is a measure of extend of modulation done on a carrier signal.

$$\mu = \frac{[e(t)]_{max} - [e(t)]_{min}}{[e(t)]_{max} + [e(t)]_{min}}$$

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$$\mu = \frac{A_c(1+Km(t)_{max}) - A_c(1+Km(t)_{min})}{A_c(1+Km(t)_{max}) + A_c(1+km(t)_{min})}$$

$$\mu = \frac{k[m(t)_{max} - m(t)_{min}]}{2 + k[m(t)_{max} + m(t)_{min}]}$$

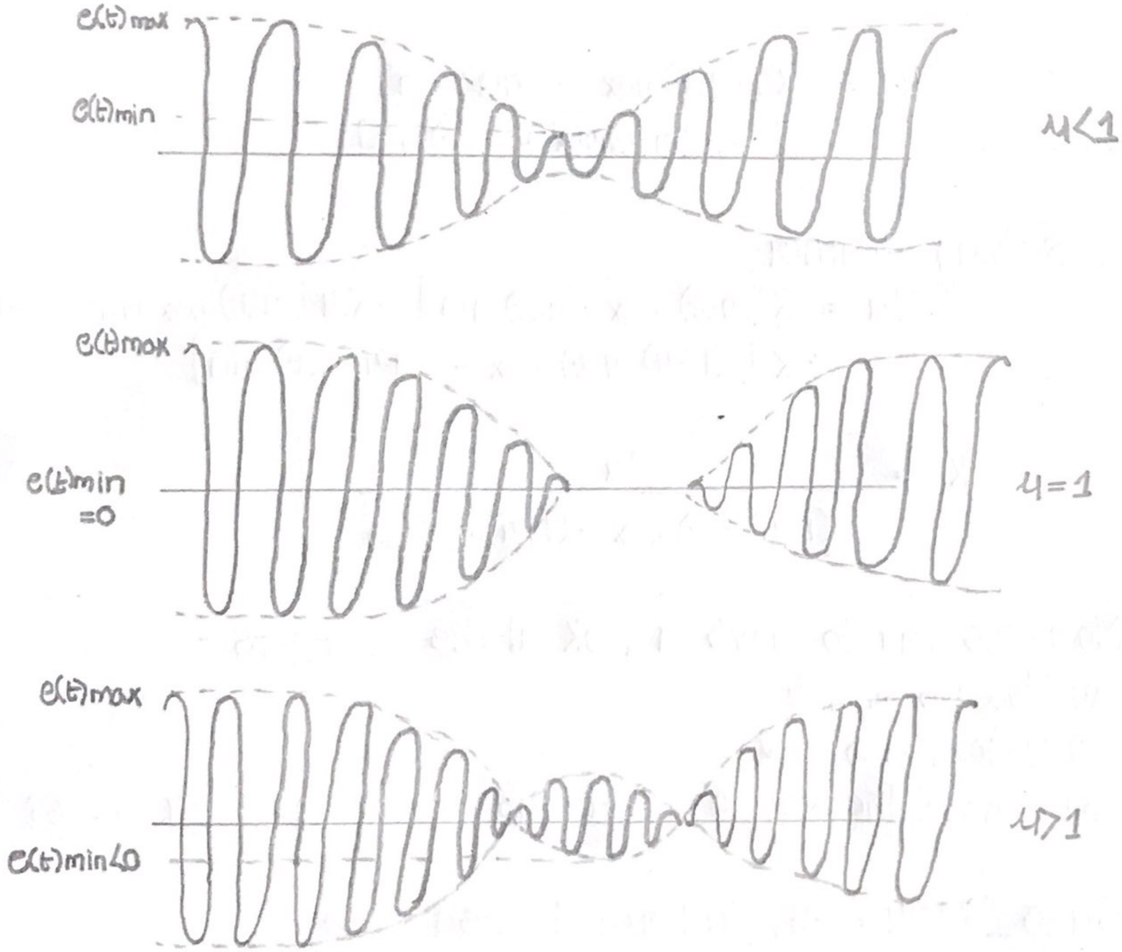
Solving Further

$$2\mu = k[m(t)_{max} - m(t)_{min}] - K\mu[m(t)_{max} + m(t)_{min}]$$
$$= k[(1-\mu)m(t)_{max} - (1+\mu)m(t)_{min}]$$

$$\boxed{K = \frac{2\mu}{(1-\mu)m(t)_{max} - (1+\mu)m(t)_{min}}}$$

Now we try to have a look at the outputs -
(1) Message Signal
(2) Modulated Signal
(3) Demodulated Signal for the following four cases:

Case I: $\mu = 0.5$, Undermodulated Case
For $\mu < 1$, the wave is said to be undermodulated. It has a valid DSB-FC but has comparitively smaller power in sidebands that carry the information reducing power efficiency of the system. The case is called undermodulation.


e(t)max
e(t)min
u<1
e(t)max
e(t)min
=0
u=1
e(t)max
e(t)min<0
u>1

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Case II: $\mu=1$, 100% Modulation
This case is perfect DSB-FC which is theoretically still recoverable, maintaining maximum permissible power in the side bands. This is therefore called fully modulated or perfectly modulated DSB-FC waveform.

Case III: $\mu=1.5$, Over modulated Case
This case validates the envelop condition making $e(t) \propto |m(t)|$ instead of simply $m(t)$ making simpler recovery impossible. This is called over modulation

Case IV: Multiple Sinusoidal inputs, $\mu=1$
The message signal is a combination of different frequencies rather than a signal of single frequency. Everything else is similar to case II.

Result:
We simulated double sideband full carrier using matlab and saw the results for various cases

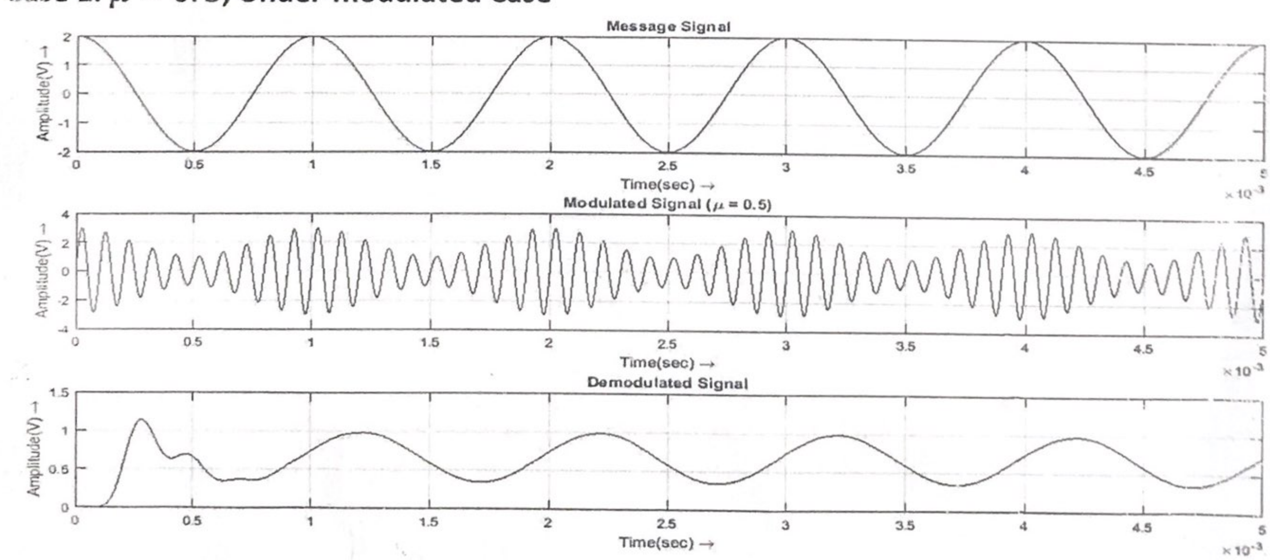
Experiment No.2

Aim: To simulate Double Side Band Full Carrier (DSBFC/ DSBAM) modulation scheme using MATLAB.

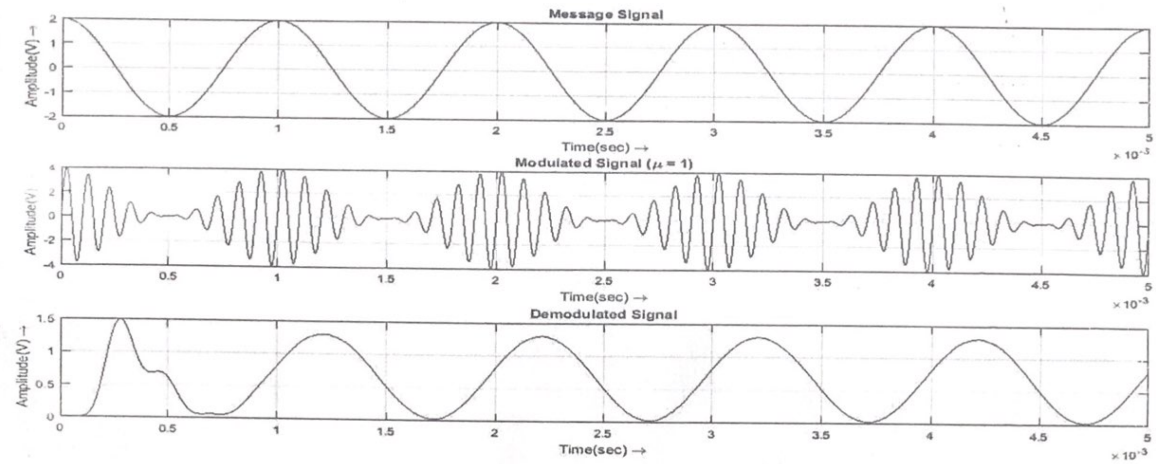
Code:

```
clear all
close all
clc
mu = input('Enter the Modulation Index: ');
Am = input('Enter Message Signal Amplitude: ');
fm = input('Enter Message Signal Frequency: ');
Ac = input('Enter Carrier Signal Amplitude: ');
fc = input('Enter Carrier Signal Frequency: ');
t = 0 : 1/(50*fc) : 2/fm;
m = Am*cos(2*pi*fm*t);
c = Ac*sin(2*pi*fc*t);
k = 2*mu/((1 - mu)*max(m) - (1 + mu)*min(m));
% Modulation
x = k*m.*c + c;
r = x;
r(r<0) = 0;
%Demodulation
[n,d] = butter(10, 1/50);
y = filter(n,d,r);
subplot(3,1,1);
plot(t,m, 'k');
set(gcf, 'Color', 'w');
grid on;
xlabel('Time(sec) \rightarrow', 'FontSize', 10);
ylabel('Amplitude(V) \rightarrow', 'FontSize', 10);
title('Message Signal (Multiple Sinusoids)', 'FontSize', 11);
set(gca, 'FontSize', 10);
subplot(3,1,2);
plot(t,x, 'k');
grid on;
xlabel('Time(sec) \rightarrow', 'FontSize', 10);
ylabel('Amplitude(V) \rightarrow', 'FontSize', 10);
title('Modulated Signal',  'FontSize', 11);
set(gca, 'FontSize', 10);
subplot(3,1,3);
plot(t,y, 'k');
grid on;
xlabel('Time(sec) \rightarrow', 'FontSize', 10);
ylabel('Amplitude(V) \rightarrow', 'FontSize', 10);
title('Demodulated Signal',  'FontSize', 11);
set(gca, 'FontSize', 10
```

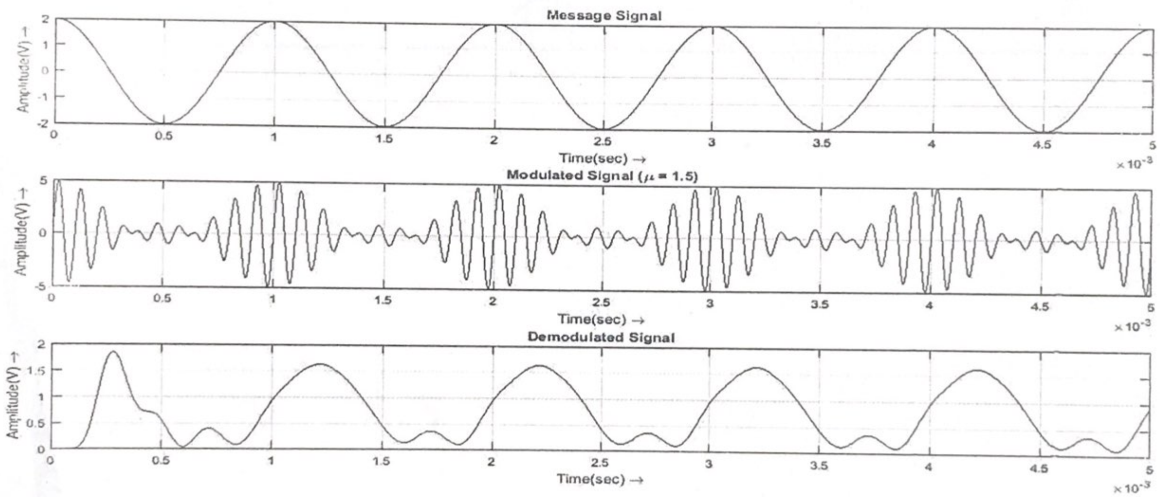
Case 1: $\mu = 0.5$, Under-modulated Case



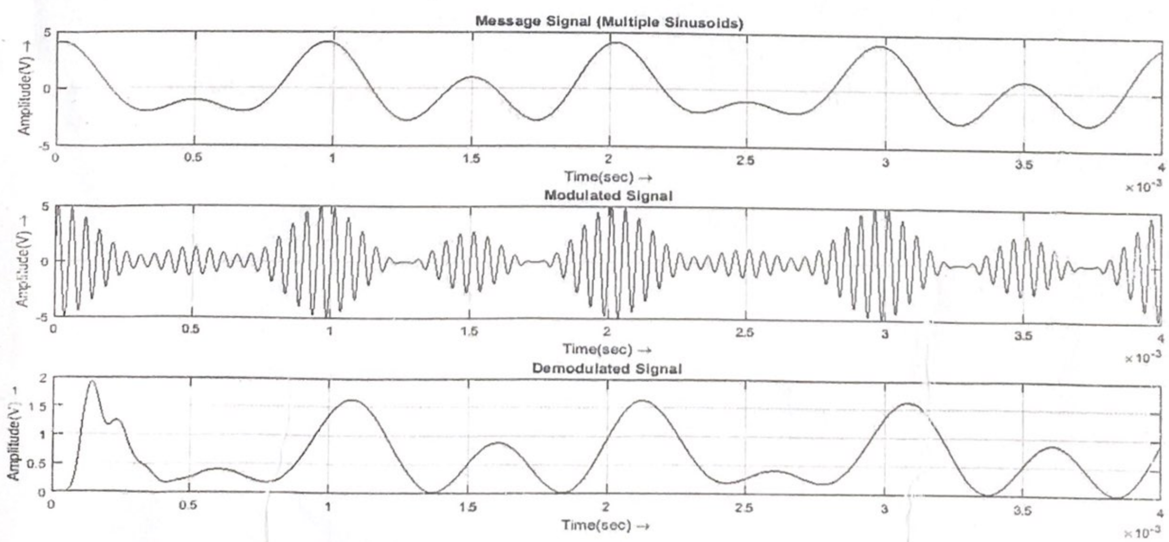
Case 2: $\mu = 1$, 100% Modulation



Case 3: $\mu = 1.5$, Over-Modulated Case



Case 4: Multiple Sinusoidal Inputs, $\mu = 1$



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Experiment No. 3

Aim:
To simulate SSB-SC ie. both LSB & USB form of modulation & study the effect of loss of coherence on their demodulation, for monotone & multitone signals.

Software Used:
MATLAB 2016b

Theory:
The DSBSC modulated signal has two sidebands. Since the two sidebands carry the same information, there is no need to transmit both sidebands. We can eliminate one sideband.
The process of suppressing one of the sidebands along with the carrier & transmitting a single sideband is called Single Sideband suppressed Carrier system or simply SSBSC.

We can suppress the carrier & the upper sideband while transmitting the lower sideband and vice versa

Bandwidth of SSBSC modulated wave $= \frac{2F_m}{2} = F_m$

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SSBSC Allows-

- Transmission of more number of signals
- Less Power consumption
- Capacity to transmit high power signal
- Reduced interference of noise
- Signal fading is less likely to occur

However generation & detection of SSBSC wave is quite a complicated process.

Discussion:

As we aldready discussed about DSBSC & DSBFC schemes of amplitude modulation we saw that the above two schemes are a waste of bandwidth since transmission bandwidth for both of them is equal to twice the bandwidth of message signal. One half of the transmission bandwidth is occupied by the upper side bands and the other half of the transmission band by lower side bands.

Moreover the message contained in the upper side bands is identical to that contain in lower side bands. Hence why do we need to transmit both the bands when while transmitting both the sidebands we are transmitting the same information twice? This is when SSB-SC comes into picture.

As far as the transmission is concerned, only one sideband is necessary. Thus, if the carrier and one of the two sidebands are suppressed at the transmitter no information is lost. Modulation of this type which provides a single sideband with suppressed carrier is known as single sideband suppressed carrier (SSB-SC) system. It reduces the transmission bandwidth by half. This means that given frequency band can accomodate twice the number of channels by using a single sideband in place of both the sidebands. This scheme of amplitude modulation is also power efficient.

In order to achieve SSB-SC Modulation we can use one of the two methods -

(I) The Filter Method

(II) Phase Shift Method.

Due to inherent disadvantages of physical filters we try using Phase Shift Method as done in this experiment.

We can write the sidebands as

$$x_{LSB}(t) = \frac{1}{2}\left[m(t)c(t) + \hat{m}(t)\hat{c}(t)\right]$$

$$x_{USB}(t) = \frac{1}{2}\left[m(t)c(t) - \hat{m}(t)\hat{c}(t)\right]$$

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Where $\hat{m}(t)$ & $\hat{c}(t)$ is the Hilbert's transform for message & carrier signal respectively. It can easily be obtained by introducing a phase difference of $-\frac{\pi}{2}$ to the signal whose Hilbert's transform is reqiured.

Now, in order to understand Phase error & Frequency error in coherent detection which is due to unavailability of ideal operating conditions for perfect synchronization, as follows -

Suppose

$$m(t) = A_m \cos w_m t$$

$$c(t) = A_c \cos w_c t$$

If local carrier generates $\cos(w_c t + \phi(t))$ where $\phi(t)$ is time varying function which can affect both phase & frequency performance in that case

$$r(t) = x(t) \cos(w_c t + \theta(t))$$

$$= \frac{1}{2} A_c [m(t) \cos (w_c t + \theta(t)) \mp \sin(w_c t) \cos(w_c t + \theta(t))]$$

$$= \frac{1}{2} A_c m(t) \left[\frac{\cos(2w_c t + \theta(t)) + \cos\theta(t)}{2} \right] \mp \frac{1}{2} A_c \hat{m}(t) \left[\frac{\sin(2w_c t + \theta(t)) - \sin\theta(t)}{2} \right]$$

Again LPF Blocks high freq components the resultant

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wave would simply be

$$y(t) = \frac{A_c}{4}\left[m(t)\cos\theta(t) \pm \hat{m}(t)\sin\theta(t)\right]$$

Therefore, when $\phi(t)=0$ then $y(t) = A_c/4\, m(t)$ is perfect replica. For any other value of $\phi(t) \neq 0$ the signal will be highly distorted.
Thus SSB will be even more sensitive to phase or frequency incoherence than DSB case.

Result:
We simulated SSBSC modulation for various cases using MATLAB

Experiment 3

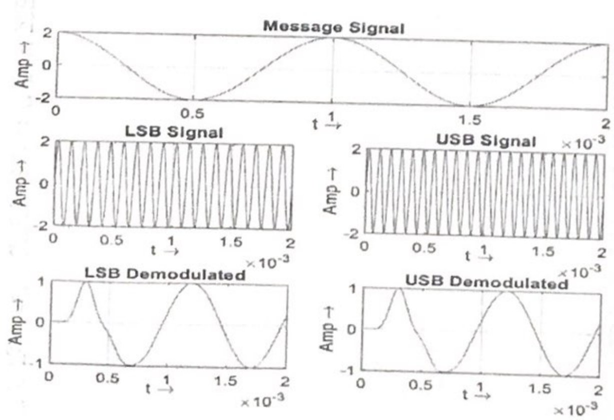
Aim: To simulate SSB-SC i.e. both LSB and USB form of modulation and study the effect of Loss of coherence on their demodulation, for monotone and multitone signals

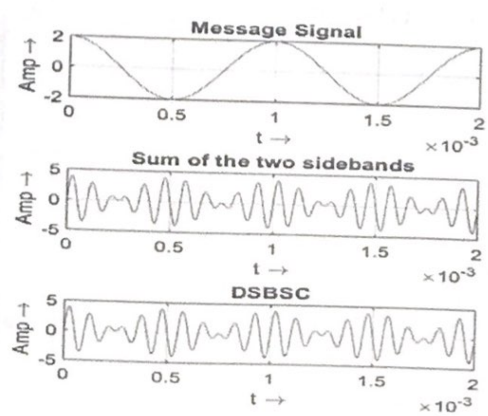
Code:

```
clc;
clear all;
close all;
Am = input('Message signal amplitude : ');
fm = input('Message signal frequency : ');
Ac = input('Carrier signal Amplitude : ');
fc = input('Carrier signal frequency : ');
t = 0:1/(50*fc):2/fm;
m = Am*cos(2*pi*fm*t);
c = Ac*sin(2*pi*fc*t);
m_h=Am*cos(2*pi*fm*t - pi/2);
c_h=Ac*sin(2*pi*fc*t - pi/2);
x_lsb = (m.*c + m_h.*c_h)/2;
x_usb = (m.*c - m_h.*c_h)/2;
r_lsb = x_lsb.*sin(2*pi*fc*t);
r_usb = x_usb.*sin(2*pi*fc*t);
[n,d] = butter(10,1/50);
y_lsb = filter(n,d,r_lsb);
y_usb = filter(n,d,r_usb);
figure(1);
set(gcf,'Color','w');
subplot(3,1,1);
plot(t,m);
xlabel('t \rightarrow');
ylabel('Amp \rightarrow');
title('Message Signal');
set(gca,'FontSize',10);
grid on;
subplot(3,2,3);
plot(t,x_lsb);
xlabel('t \rightarrow');
ylabel('Amp \rightarrow');
title('LSB Signal');
set(gca,'FontSize',10);
grid on;
subplot(3,2,4);
plot(t,x_usb);
xlabel('t \rightarrow');
ylabel('Amp \rightarrow');
title('USB Signal');
set(gca,'FontSize',10);
grid on;
subplot(3,2,5);
plot(t,y_lsb);
xlabel('t \rightarrow');
ylabel('Amp \rightarrow');
title('LSB Demodulated');
set(gca,'FontSize',10);
grid on;
subplot(3,2,6);
plot(t,y_usb);
xlabel('t \rightarrow');
ylabel('Amp \rightarrow');
title('USB Demodulated');
set(gca,'FontSize',10);
grid on;
figure(2);
set(gcf,'Color','w');
subplot(3,1,1);
plot(t,m);
xlabel('t \rightarrow');
ylabel('Amp \rightarrow');
title('Message Signal');
set(gca,'FontSize',10);
grid on;
subplot(3,1,2);
plot(t,x_lsb+x_usb);
xlabel('t \rightarrow');
ylabel('Amp \rightarrow');
title('Sum of the two sidebands');
set(gca,'FontSize',10);
grid on;
subplot(3,1,3);
plot(t,m.*c);
xlabel('t \rightarrow');
ylabel('Amp \rightarrow');
title('DSBSC');
set(gca,'FontSize',10);
grid on;
```

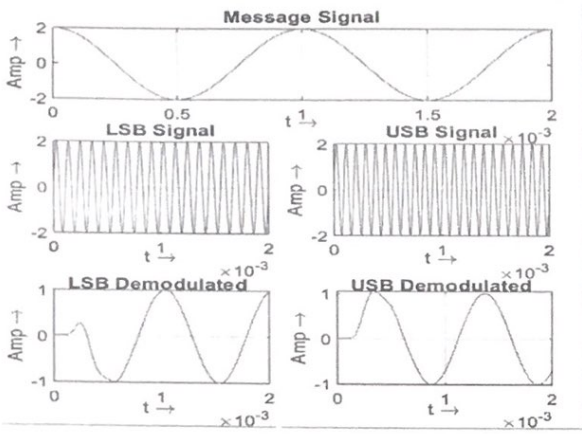
Outputs:

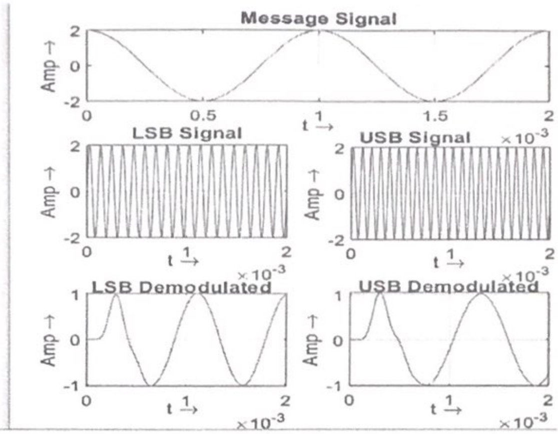
Case 1: Monotone signal, Ideal reception



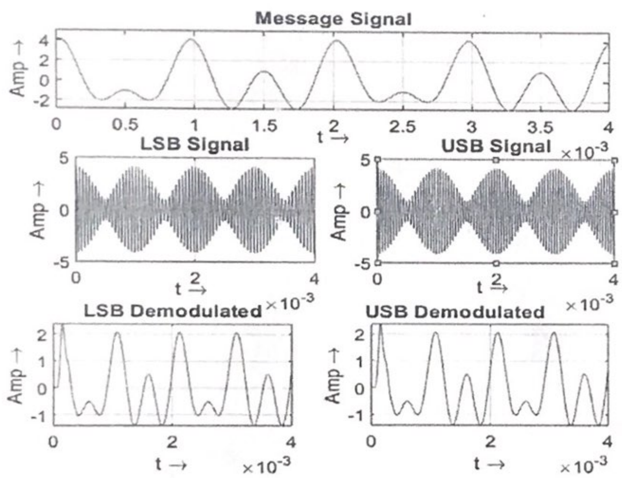


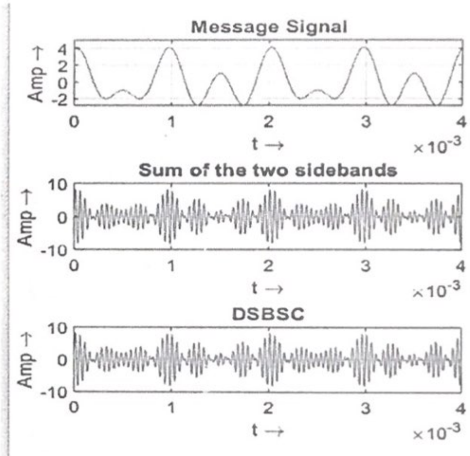
Case 2: Monotone Signal, Phase mismatch effect and frequency mismatch effect



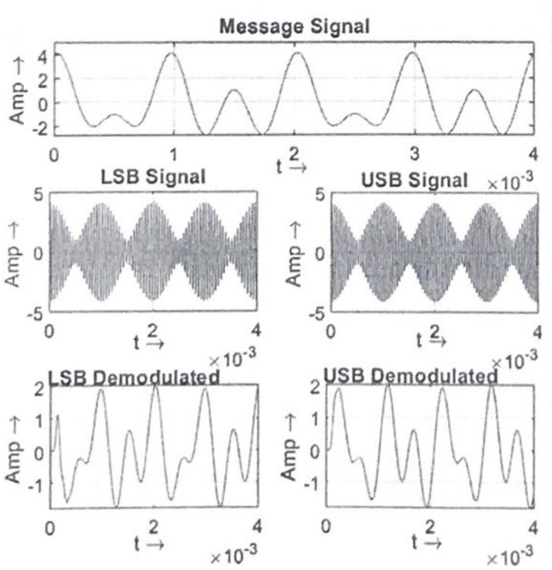


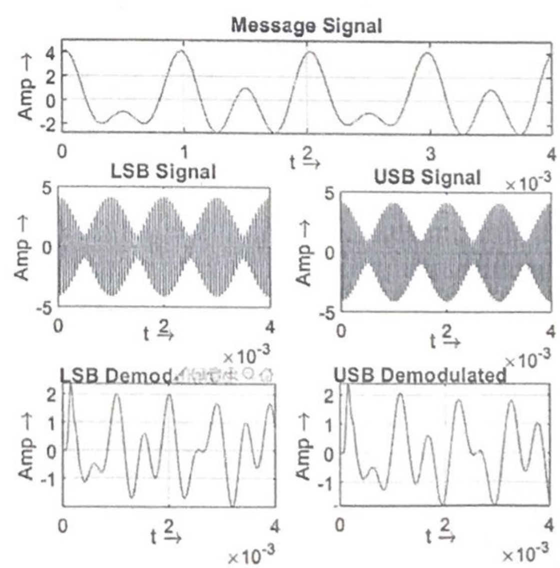
Case 3: Multitone signal, Ideal reception





Case 4: Multitone signal, Phase and frequency mismatch





Topic ______ Date ______

Experiment No 4

Aim:

To study the PDF & CDF Functions of different Discrete Random Variables & the effect of parametric changes

Software Used:

MATLAB 2016b

Theory:

A random variable associated with an experiment is a rule or relationship X(s) that assigns a real number X to every sample point 's'. If the sample space 'S' contains a countable number of sample points then X(s) will be a discrete random variable. A discrete random variable will thus have a countable number of distinct values.

CUMULATIVE DISTRIBUTION FUNCTION

The cumulative distribution function (CDF) of a random variable is defined as the probability that the random variable X takes value less than or equal to x.

ie. CDF $F_X(x) = P(X \leq x)$

Other properties are-

1.	$0 \leq F_X(x) \leq 1$	CDF is always bounded between 0 & 1
2.	$F_X(\infty) = 1$	As $X \leq \infty$ is a certain event
3.	$F_X(-\infty) = 0$	As $X \leq -\infty$ is a null event

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4. $F_X(x_1) \leq F_X(x_2)$ for $x_1 < x_2$ CDF is monotone non dec. function

PROBABILITY DENSITY FUNCTION

The PDF $f_X(x)$ is defined as derivative of cumulative distribution function. PDF: $f_X(x) = d(F_X(x))/dx$.

$f_X(x) \geq 0$ for all x. $\int_{-\infty}^{\infty} f_X(x) = 1$. $P(x_1 \leq X \leq x_2) = \int_{x_1}^{x_2} f_X(x)\,dx$

Discussion:

When solving problems, if we are able to recognise that the random variable fits one of the several distributions such as Binomial, Bernoulli, Poisson etc. we can simply use its precalculated probability mass function, expectation, variance & other properties. Random variables of this sort are called 'parametric' random variables. If we can argue that the random variable falls under one of the studied parametric types, we simply need to provide parameters.

(1) Bernoulli Random Variable is the simplest kind of random variable. It can take only two values 0 & 1. It takes on a 1 if an experiment with probability p resulted in a success & a zero otherwise.

Let us try to understand what happened in the experiment:

We consider probability of success as q

$q = 0.3$ here

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For an array x_be = -3:1:4

wiz. x_be = [-3, -2, -1, 0, 1, 2, 3, 4]

Bernoulli pdf is given by

$P(X=1) = p$ 'prob. of success'

$P(X=0) = 1-p$

In our case at first f_be = [0, 0, 0, 0, 0, 0, 0, 0]

PDF is f_be (x_be == 1) = q = 0.3 here

f_be (x_be == 0) = 1-q = 0.7

hence f_be = [0, 0, 0, 0.7, 0.3, 0, 0, 0]

wiz. Probability Density Function

We know that CDF $F_X(x) = P(X \leq x)$

here CDF F_be(1) = 0

F_be (2) = 0 + 0 = 0

F_be (3) = 0 + 0 + 0 = 0

F_be(4) = 0+0+0+0.7 = 0.7

F_be(5) = 0+0+0+0.7+0.3 = 1.0

F_be(6) = 1.0

F_be(7) = 1.0

F_be(8) = 1.0

hence F_be = [0, 0, 0, 0.7, 1.0, 1.0, 1.0, 1.0]

wiz. Cumulative Distribution Function.

Similarly we did for p = 0.5 & p = 0.7

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(2) Binomial Random Variable

A Binomial Random Variable is random variable that represents the number of successes in n successive independent trials of Bernoulli Experiment.

For a Binomial Random Variable

$$P(X=K) = {}^{n}C_{K}\, p^{K} (1-p)^{n-K} \quad 0 \leq K \leq n.$$

$$E(X) = np$$

$$Var(X) = np(1-p)$$

Where p is probability of success in a given trial

Let us consider $n = 20$

$p = 0.3$

$x\_bi = [0,1,2,3,4,5 --- 18,19,20]$

At first $f\_bi = [0,0,0 --- 0]$

but as we know $P(X=K) = {}^{n}C_{K}\, p^{k} (1-p)^{n-k}$

$f\_bi(\underbrace{K+1}) = {}^{n}C_{K}\, p^{K} (1-p)^{n-k}$

so $f\_bi(1) = {}^{20}C_{0}\, p^{0} (1-p)^{20}$ First element. Indexing begins with 1

$= 0.008$

Similarly we plot for diff values of K ranging from 0 to 20 Giving us the PDF as shown in Fig.

CDF is nothing but given by $P(X \leq x) = F_X(x)$

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(3) Poisson Random Variable:

Poisson's distribution is another standard probability distribution for discrete random variables. As the number 'n' increases, the binomial distribution becomes difficult to handle. If 'n' is very large, probability 'p' is very small & the mean value 'np' is finite, then the binomial distribution can be approximated by poisson's distribution given by

$$P(X=K) = \frac{m^k e^{-m}}{k!}$$

Mean Value $m_x = np$

Variance $\sigma_x^2 = np$

Standard deviation $\sqrt{\sigma_x} = \sqrt{np}$

Result:

We studied PDF & CDF Functions of diff. random variables for Bernoulli's, Binomial & Poisson's distribution.

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Experiment No. 5

Aim:

To study PDF & CDF functions of different continous Random Variables & the effect of parametric changes.

Software Used:

MATLAB 2016b

Theory:

A continous random variable is not restricted to a finite number of distinct values. Instead, it can have any values within a certain range. Thus, the continous random variable has an 'uncountable' number of possible values. Such a random variable is defined for systems which generate an infinite number of outputs within a finite period of time.

CUMULATIVE DISTRIBUTION FUNCTION

Cumulative Distribution Function of a random variable is defined as the probability that the random variable X takes value less than or equal to x

$$\text{CDF } F_X(x) = P(X \leq x)$$

1. CDF is always bounded between 0 and 1

$$0 \leq F_X(x) \leq 1$$

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2. $F_X(\infty) = 1$ as $X \leq \infty$ is a certain event
3. $F_X(-\infty) = 0$ as $X < -\infty$ is a null event.
4. $F_X(x_1) \leq F_X(x_2)$ for $x_1 < x_2$ hence CDF is a monotone non-decreasing function.

PROBABILITY DENSITY FUNCTION

The probability density function $f_X(x)$ is defined as the derivative of the cumulative distribution function. Thus we have

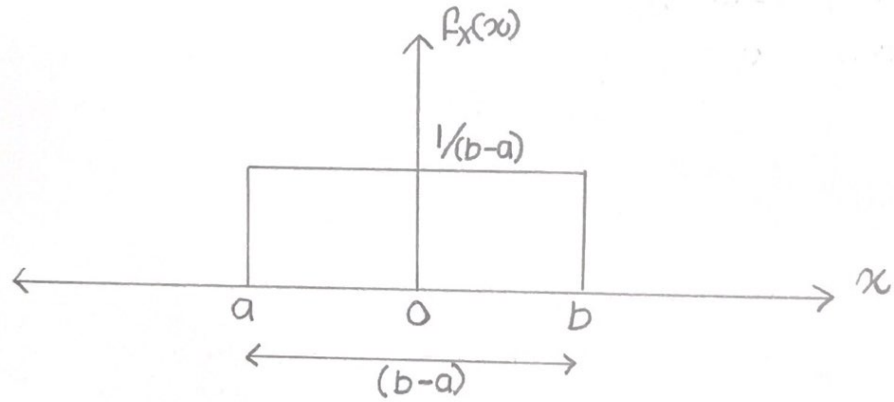
$$f_X(x) = \frac{d}{dx} F_X(x)$$

1. CDF is obtained by integrating PDF

$$F_X(x) = \int_{-\infty}^{x} f_X(x)\, dx$$

2. $f_X(x) \geq 0$ for all x. PDF is a non-negative function.
3. $\int_{-\infty}^{\infty} f_X(x) = 1$. Area under PDF curve is 1.
4. $P(x_1 < X \leq x_2) = \int_{x_1}^{x_2} f_X(x)\, dx$

Probability of obtaining X between x_1 & x_2 is equal to the area under the PDF curve between the values x_1 & x_2.



Uniform Distribution

Discussion:

1. Uniform Random Variable:

If a continous random variable X is equally likely to be observed in a finite range and is likely to have a zero value outside the finite range, then the random variable is said to have a uniform distribution. The PDF of a random variable having a uniform PDF is given by,

Uniform PDF:

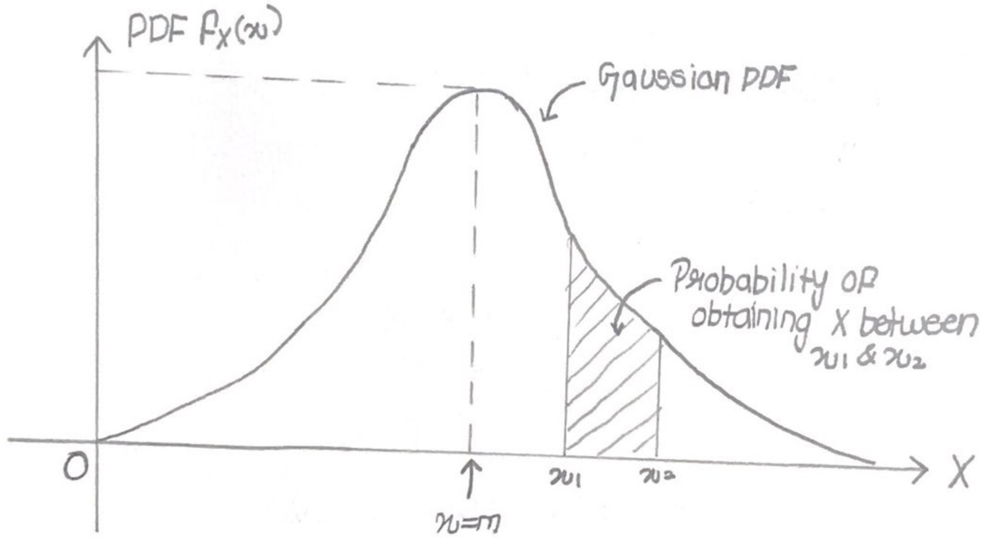
$$f_X(x) = \begin{cases} \frac{1}{b-a} & \text{for } a \leq x \leq b \\ 0 & \text{elsewhere} \end{cases}$$

Mean of uniform distribution $m_X = \frac{(a+b)}{2}$

Variance of uniform distribution $= \sigma_X^2 = \frac{(b-a)^2}{12}$

2. Gaussian Distribution (or Normal Distribution)

The Gaussian distribution is used for continous random variables. It is perhaps the most important PDF in the area of communication. The majority of noise processes observed in practice are Gaussian and many naturally occuring experiments are characterized by continous random variables with Gaussian PDF.



The Gaussian PDF

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The PDF of a continous random variable having Gaussian distribution is given by:

$$\text{Gaussian PDF - } f_X(x) = \frac{1}{\sigma\sqrt{2\pi}} e^{-(x-m)^2/2\sigma^2} \quad -\infty < x < \infty$$

Where m = Mean of the random variable
σ^2 = Variance of the random variable

The Gaussian PDF is a bell shaped function with peak at $x = m$ ie. corresponding to the mean value of the random variable X. The Gaussian PDF has an even symmetry about the peak

$$f_X(x = m-\sigma) = f_X(x = m+\sigma)$$

$$P(X \leq m) = P(X > m) = \frac{1}{2}$$

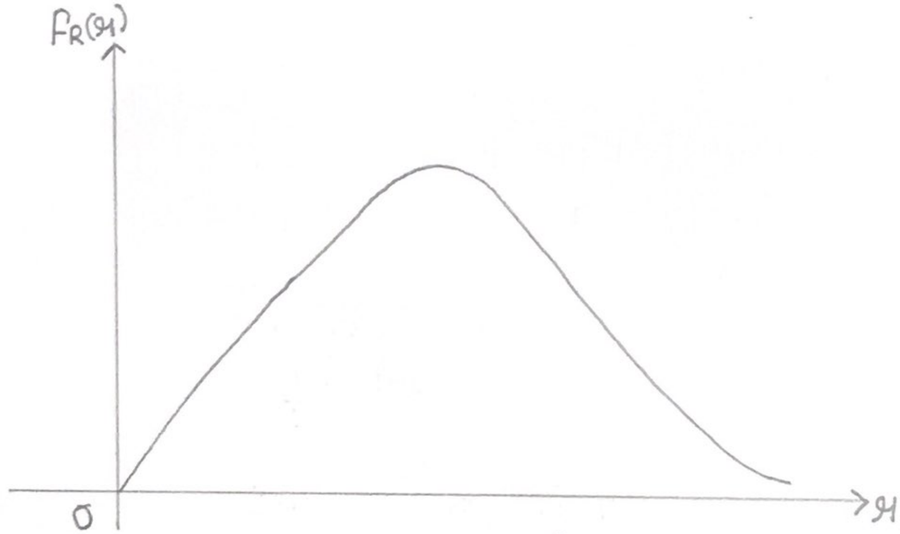
Area under Gaussian PDF is equal to 1

$$f_X(x)\,dx = 1$$

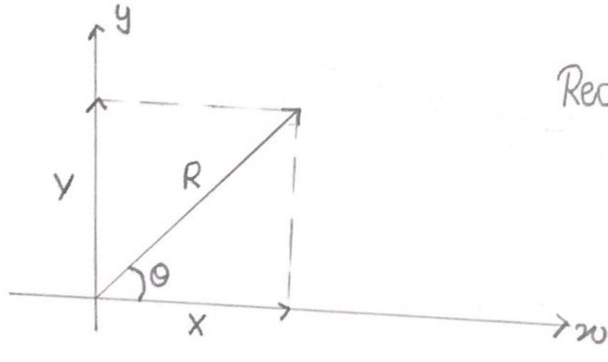
$$P(x_1 < X \leq x_2) = \int_{x_1}^{x_2} f_X(x)\,dx$$

$$= \int_{x_1}^{x_2} \frac{1}{\sqrt{2\pi\sigma^2}} e^{-(x-m)^2/2\sigma^2}\,dx$$

$$\text{CDF} \quad F_X(x) = \frac{1}{2}\left[\frac{2}{\sqrt{\pi}} \int_{\frac{m-x}{\sigma\sqrt{2}}}^{\infty} e^{-y^2}\,dy\right]$$



Rayleigh's PDF



Rectangular to Polar Conversion

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3. Rayleigh's Distribution

The Rayleigh's distribution is used for continous random variables. It describes a continous random variable produced from two Gaussian random variables.

Let X & Y be two independent Gaussian random variables having

$$m_x = m_y = m \quad \& \quad \sigma_x = \sigma_y = \sigma$$

The Rayleigh's continous random variable R is related to X & Y by the transformation in figure. Gaussian random variables X & Y are related to the Rayleigh's random variables R & ϕ as following:

$$R = \sqrt{X^2 + Y^2}$$

$$\phi = \tan^{-1}\left(\frac{Y}{X}\right)$$

The expression for Rayleigh PDF is given by

$$f_R(r) = \frac{r}{\sigma^2} e^{-r^2/2\sigma^2} \quad \text{for } r \geq 0$$

$$= 0 \quad \text{for } r < 0$$

Mean value for Rayleigh's PDF $m_x = \sqrt{\pi/2}\,\sigma$

Variance of Rayleigh's PDF $\sigma_x^2 = \left[2 - \frac{\pi}{2}\right]\sigma^2$

Result:

We studied PDF & CDF functions of continous random variables.

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Experiment No. 6

Aim:

To simulate the generation of frequency modulated signal for a single sinusoidal signal & for a multi-sinusoidal signal.

Software Used:

MATLAB 2016b

Theory:

The frequency modulation (F.M.) is a type of angle modulation in which the instantaneous frequency $f_i(t)$ is varied in linear proportion with the instantaneous magnitude of the message signal $x(t)$. This is expressed mathematically as follows:

For F.M. $f_i(t) = f_c + k_f\, x(t)$

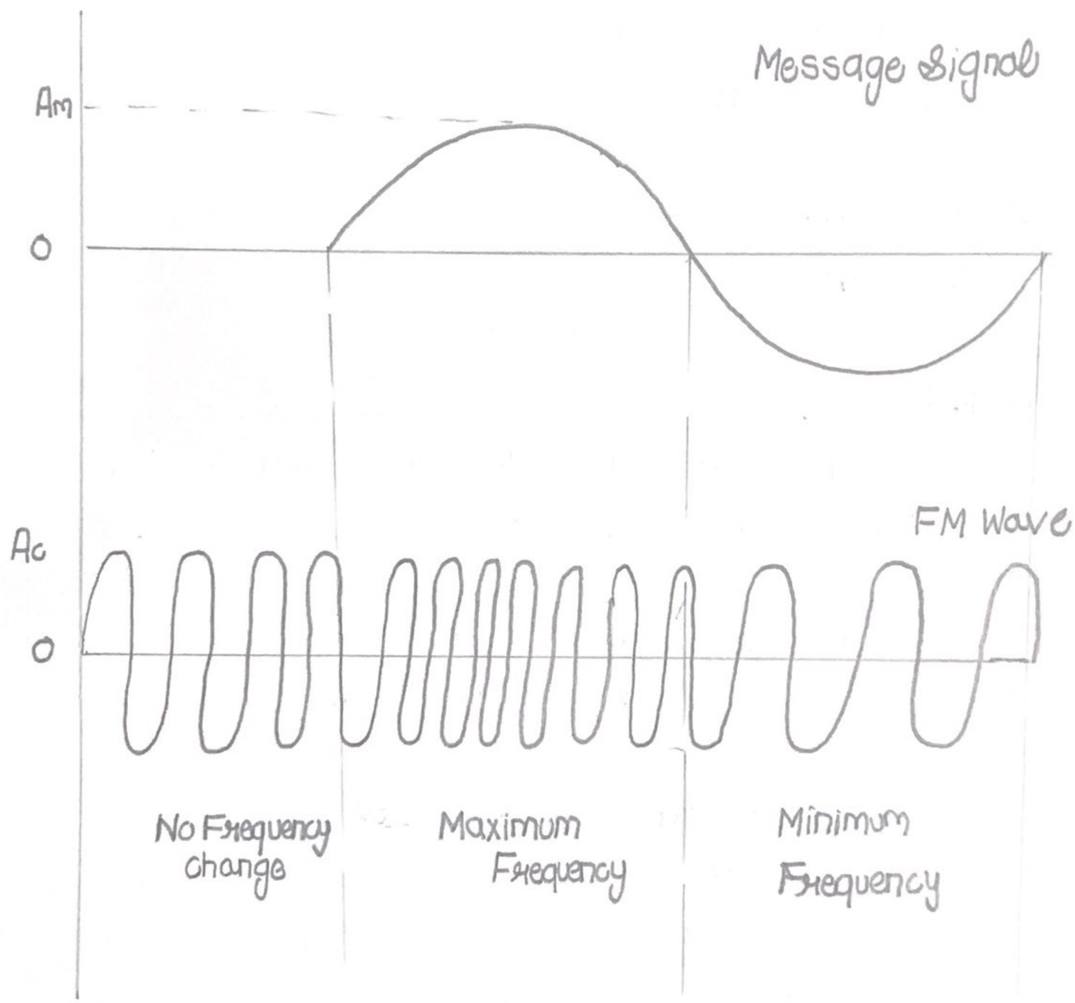
where $f_i(t)$ is Instantaneous Frequency

f_c is unmodulated carrier frequency

k_f is frequency sensitivity in Hz/volt

Since the instantaneous frequency $f_i(t)$ of FM is changing continously with time, we have to take integration of $x(t)$ over a duration of 0 to t.

$$\theta(t) = 2\pi f_c t + \cancel{2\pi k_f} \int_{-\infty}^{t} k_f\, x(\tau)\, d\tau$$



Message Signal
Am
0
FM Wave
Ac
0
No Frequency change
Maximum Frequency
Minimum Frequency

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Now,

$$x(t) = E_c \cos\theta(t)$$

F.M. Wave $x(t) = E_c \cos\left[2\pi f_c t + 2\pi k_f \int_{-\infty}^{t} x(\tau)\, d\tau\right]$

Discussion:

In sinusoidal frequency modulation the modulating signal, $m(t) = A_m \cos(2\pi f_m t)$ which is a pure sinusoidal signal.

FM is a system of modulation in which instantaneous frequency of the carrier is varied in proportion with the amplitude of the modulating signal. The amplitude of carrier signal remains constant. Thus the information is conveyed via frequency changes.

The amount by which the carrier frequency varies from its unmodulated value is called the deviation. The deviation (Δf) is made proportional to the instantaneous value of modulating voltage. The rate at which this frequency variation or oscillation takes place is equal to the modulating frequency (f_m). The amplitude of the FM wave always remains constant. This is the biggest advantage of FM.

Let us analyze frequency modulation for single sinusoidal signal & for multi-sinusoidal signal.

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Advantages of Frequency Modulation

- Resilience to Noise:
  The modulation is carried only as variations in frequency. This means any signal level variations will not effect the audio output, provided the signal doesn't fall to level where the receiver couldn't cope.

- Easy to apply modulation at a low power stage of transmitter. It is possible to apply modulation to a low power stage of the transmitter.

- It is possible to use efficient RF Amplifiers with frequency modulated signals

Disadvantages of Frequency Modulation

- FM has poorer spectral efficiency than some other modulation formats

- It requires more complicated demodulator

- Sidebands extend to infinity either side

Results:
We successfully generated FM signals for single & multi sinusoidal signals.

Experiment No. 7

Aim:

To simulate the generation of Phase Modulated signal for a single sinusoidal signal & for a multi-sinusoidal signal.

Software Used:

MATLAB 2016b

Theory:

Phase modulation is the type of angle modulation in which the angular argument $\theta(t)$ is changed in linear proportion with the instantaneous magnitude of message signal $x(t)$.

The general form of modulated signal can be written as

$$x_{PM}(t) = A\cos[\omega_c t + \phi(t)]$$

Where $\phi(t)$ captures the behaviour of message $m(t)$

The overall argument of cosine term may be written as

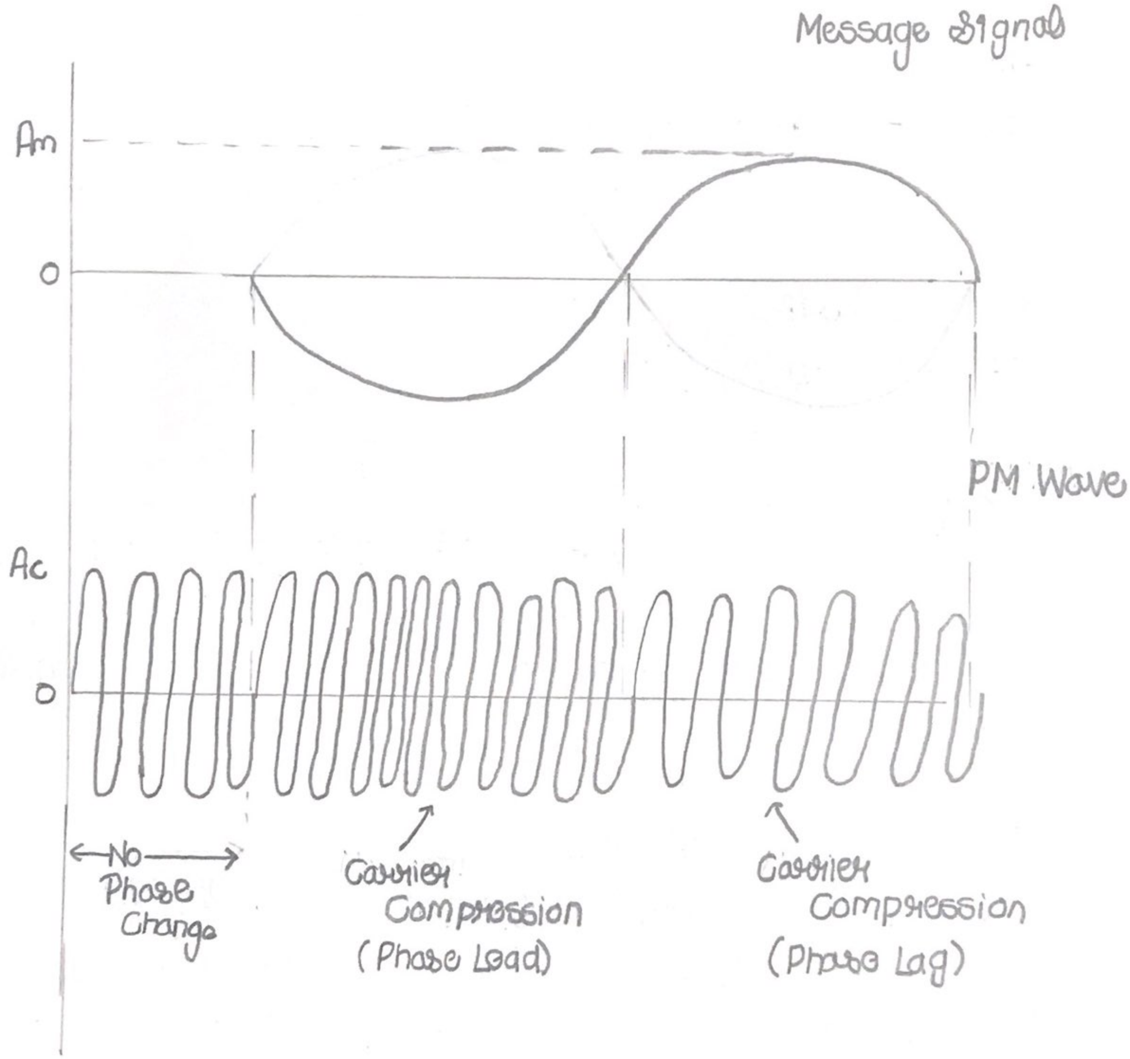
$$\theta_i(t) = \omega_c t + \phi(t)$$

as the instantaneous phase of modulated signal.

$$\omega_i(t) = \frac{d\theta_i(t)}{dt}$$

$$\omega_i(t) = \omega_c + d\phi(t)/dt$$

Here $\phi(t)$ in this case is defined as instantaneous phase deviation,


Message Signal
Am
0
PM Wave
Ac
0
No Phase Change
Carrier Compression (Phase Lead)
Carrier Compression (Phase Lag)

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Now if in such a case, phase deviation is made directly proportional to message signal variation, then such a modulation is called phase modulation.

$$\phi(t) = k_p m(t)$$

Where k_p = Phase sensitivity of modulator (rad/v)

hence

$$x_{PM}(t) = A\cos[\omega_c t + k_p m(t)]$$

Discussion:

Phase modulation is very similar too the frequency modulation. The only difference is that the phase of the carrier is varied instead of varying the frequency. The amplitude of the carrier remains constant.

As the modulating signal goes positive, the amount of phase lead increases with the amplitude of modulating signal. The effect is that the carrier signal is compressed or its frequency is increased. When the modulating signal goes negative, the phase shift becomes lagging. This causes the carrier wave to be effectively stretched up. The effect is as if the carrier frequency is reduced.

Thus, phase modulation produces frequency modulation

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Advantages of Phase Modulation:

- Phase Modulation is simple contrasted to Frequency Modulation
- Can be used to find the velocity of target by removing doppler effect. This needs constant carrier which is achievable
- using phase modulation however not in FM
- The main benefit of this modulation is signal modulation because it permits computer for communicating on highspeed using a telephone system
- When the information is being transmitted without instrusion then the speed rates can be observed
- PM also has improved immunity towards noise

Disadvantages of Phase Modulation:

- Phase modulation needs two signals by a phase variation among them. Through this, both the two patterns are recquired like a reference as well as a signal

- PM generally recquires hardware which obtains more complex due to its conversion technique

- Phase modulation index can be enhanced by employing frequency multiplier

Result:

We simulated the generation of phase modulated signal successfully.

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Experiment No. 8

Aim:

To simulate the different parts of Super Hetrodyne Receivers for DSBAM Modulation

Software Used:

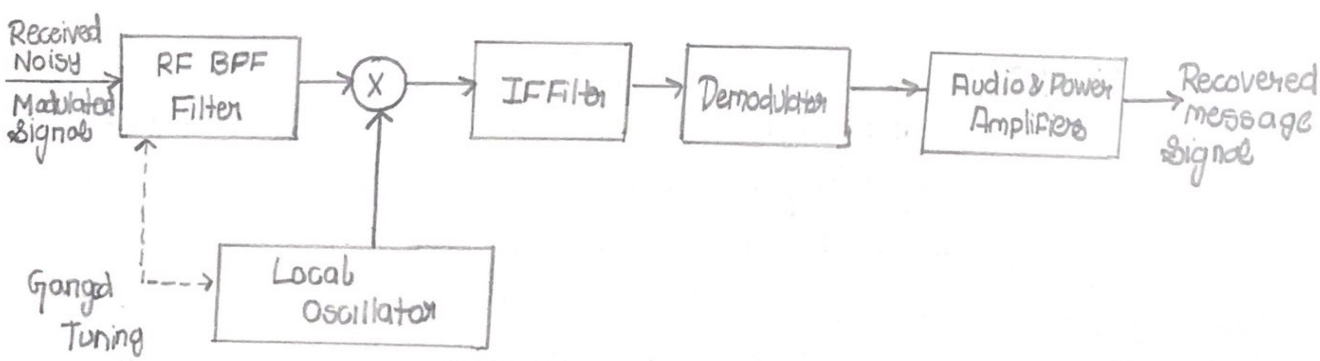
MATLAB 2016b

Theory:

The term superhetrodyne refers to creating a beat freq. that is lower than original signal. What superhetrodyning does is to purposely mix in another frequency in the receiver, so as to reduce the signal frequency prior to processing.
In a radio application we are reducing the AM or FM signal which is centered on the carrier frequency to some intermediate value, called the IF (Intermediate Frequency).
For Practical Purposes, the superhetrodyne always reduced to the same value of IF. This requires we are able to continously vary frequency being mixed into the signal so as to keep diff. the same.

Advantages of Using Superhetrodyne Receiver:

- It reduces the signal from very high freq. sources when ordinary components wouldn't work.
- It allows many components to operate at a fixed frequency (IF Section). & therefore can be optimized or made inexpensively.
- It can be used to improve signal isolation by Arithmetic Selectivity.



Super Hetrodyne Receiver

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Discussion:

In case of a TRF Receiver before shifting the signal, boasting of the signal was done.
However in case of a superhetrodyne receiver we focus only on the portion required, then extracting it to be amplified after which it is sent to the detector.
A detector is nothing but a demodulator. Once the signal has been demodulated, the audio and power amplifiers come into picture, we hear a sound.

We start our experiment by getting the amplitude and frequency of message signal from the user. However in our case we took predefined values as 1V and 4KHz respectively. 4KHz is taken in order to signify transmission of voice.
In the analysis of superhetrodyne receiver we come across something known as Sampling frequency denoted by fs. Sampling frequency is reoquired because our system is of type digital. Sampling is the only thread that connects Analog & Digital platform.
While Working with Receivers Carrier frequencies of different stations are available at the Receiver end. Therefore we take multiple values for Fc.
Now we need to tell our system which frequency station do we want to tune to. Fc(2) signifies Second station ie 700 KHz

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Our next step is to generate an AM Wave. Inorder to do that we use inbuilt function of modulate(). This function takes message signal, Frequency of carrier, sampling frequency and type of modulation as its parameters.

amdsb-tc is keyword for DSB-AM Total Carrier.
amdsb-sc is for DSBSC AM.
amssb is for SSB Amplitude Modulation
fm for Frequency modulation & pm for phase modulation

We create a for loop to get an AM signal with multiple Frequencies i.e. 600 KHz, 700 KHz & 800 KHz all jumbled up together. We do add an Image Frequency to the desired AM Signal to check if the filtering process is working properly.

In order to plot given system's Frequency response:
fft is used to generate the Fourier transform later scaled down by N. To assign frequency range we take from 0 to Fs/2, the positive part of spectrum.
fft gives us value in terms of complex numbers. We plot the magnitude beginning from 0 to Fs/2 +1.
In Received Signal Spectrum we get three Frequencies of the stations along with their side bands & a Image Frequency.

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Next Step is RF Stage, band pass Filter which removes the Image Station off.

In order to increase the selectivity of RF-Stage, we see that For lower Frequency to be $F_c - F_{if}$ & higher Frequency to be F_{if}. For any Frequency of $F_c \pm 2F_{if}$, it won't lie in above range. It makes the system highly selective or resistant to image Frequencies associated with it.

L_cutoff_Freq = Ft - F_if

U_cutoff_Freq = Ft + F_if

The coeff. being provided as parameters to the butter Filter must lie between range 0 to 1. Hence L & U-cutoff are scaled down by Fs/2 it being the maximum value. Butterworth Filter here is 5th order band pass Filter which leads us to RF band pass Filter response where image Frequency has vanished.

In Next Step we transfer the entire system to Intermediate Frequency For which local oscillator is used For third graph i.e. response after mixing signals with $F_t + F_{if}$.

The last step is to remove all the Frequencies not required. For that we need a highly selective model.

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The highly selective part of hetrodyne receiver is the IF Filter or Intermediate Frequency band pass Filter. IF Filter removes all other stations from its purview. Some buffer is also required so the signal is not attenuated in any way. 50Hz Buffer means extra 100 Hz Bandwidth in the system.

IF Filter Response gives us a single freq at 455 kHz associated with its side bands.
All that is demodulated using demod() command to get the original message signal but in sampled form.
Very few samples are plotted as shown in recovered message signal diagram.

Result:
We simulated different parts of Super Hetrodyne Receivers for DSBAM Modulation.