Benha University

Benha Faculty of Engineering

Dept. of Electrical Engineering

Subject Name :

Communication system project

Submitted By :

Name : Mahmoud Saad Mostafa

Supervised By :

DR. Ayman Mostafa
Dept. of Electrical Engineering
Benha Faculty of Engineering
Benha University

May 2024

- Abstract

In the realm of modern communication systems, the efficient transmission of information is of paramount importance. Modulation techniques have emerged as indispensable tools for encoding and decoding signals, enabling reliable and high-quality communication in various domains.

The purpose of this report is to explore the fundamental principles of modulation, examining how it allows for the efficient transmission of data by imposing information on carrier signals. Furthermore, we will analyze the various modulation schemes employed in different communication systems, considering their strengths, limitations, and trade-offs.

1-Design part 1

1.1 functions

1.1.1. Audio read function

In MATLAB, the audioread function is used to read audio files. It can handle various audio file formats, including .wav, .mp3, and .flac. Here's a detailed explanation of how to use the audioread function
{name = audioread(filename)}.

1.1.2. Low pass filter (Butter & Filter)

Butterworth Filter: Characterized by a smooth, flat frequency response in the passband with a gradual roll-off. Preferred for applications where phase linearity and minimal passband distortion are crucial,

```
[b,a] = butter(6,Fc/(Fs));
```

The butter function designs a **6th-order** low-pass Butterworth filter with a specified cutoff frequency.
The normalized cutoff frequency **Fc/(Fs)** determines where the filter will start to attenuate higher frequencies

filter(m,n,input_signal): This function applies the designed Butterworth filter to the input signal input_signal.
M: The numerator coefficients of the filter's transfer function.
N: The denominator coefficients of the filter's transfer function.

1.1.3. Modulation DSB_LC (AM)

```
modulated_signal =
(1 + mod_index*(filtered_signal)).*carrier;
```

```
This relation is used to modulation by DSB_LC technic
```

The modulation index : is a key parameter in amplitude modulation (AM) that governs the extent of signal variation imposed on the carrier.

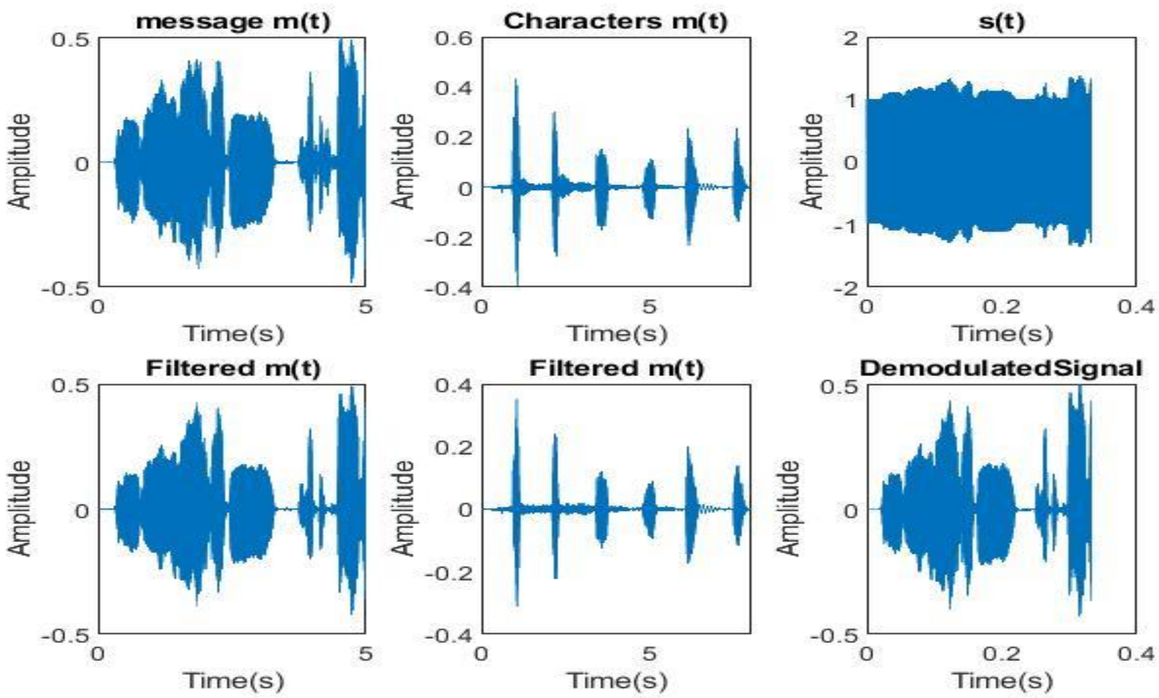
1.1.4. Envelop function

```
envelope = abs(1 + mod_index*(filtered_signal));
```

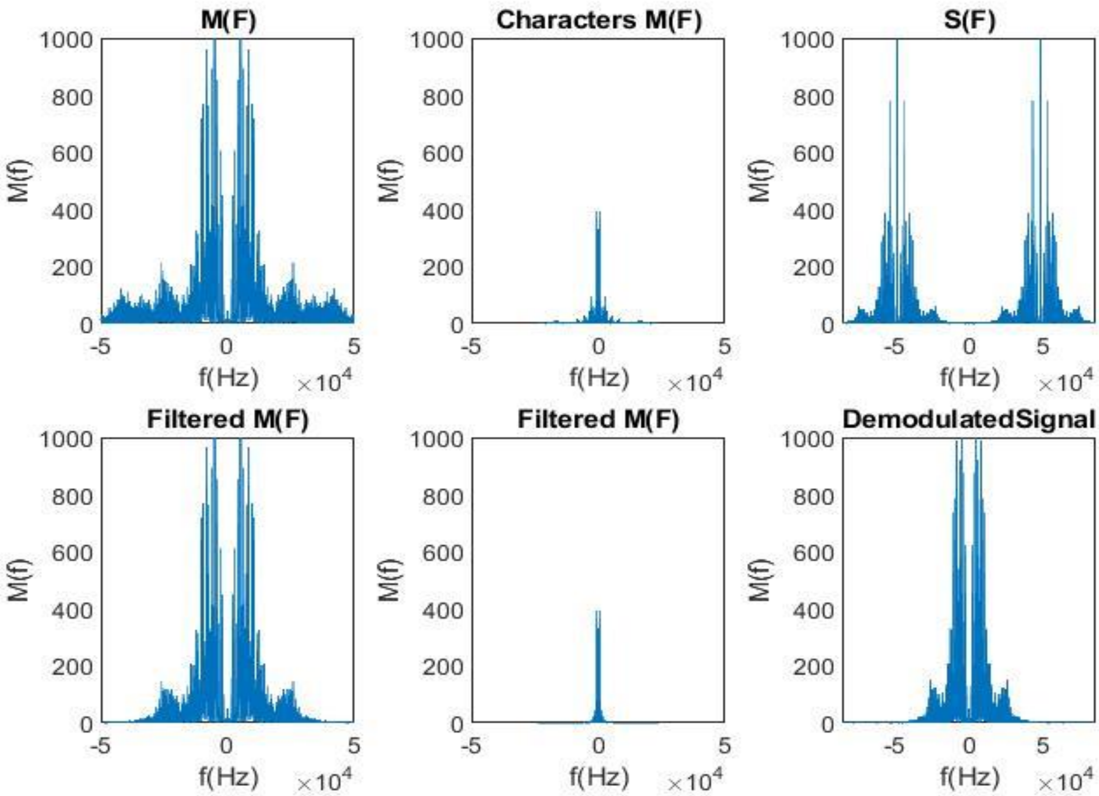
```
we can use this relation to reconstruct signal with dc value
this relation is used if modulation index less than or equal 1
then modulation is envelop - Dc value .
```

1.2. Plot

1.2.1. Signals plot in time domain



1.2.2. Signals plot in frequency domain



Result :

Cut off frequency after which the signal message becomes unintelligible is under **800Hz.**

Cut off frequency after which the characters signal becomes unintelligible is under **1500Hz**.

2- Design Part 2

2.1. Modulation by using DSB_SC

```
modulated_signal =
(mod_index*(filtered_signal)).*carrier;
```

This relation is used to DSB_SC technic .

This technic neglect carrier and transmit signal ,so power efficiency is 100% in this technic .

2.2. Demodulation for DSB_SC

We can demodulate by using multiply received signal by carrier and entered result on LOW_PASS_FILTER

```
demodulated_signal = modulated_signal .* carrier;

demodulated_signal=LPF(demodulated_signal,Fc,Fs);
```

2.3. modulation by using SSB technic

We can use function bult_in to modulate signal by using SSB technic

```
ssb_sc_signal =
modulate(filtered_signal,CarrierFreq,15*Fs,'amssb');
```

2.4. demodulation of SSB technic

We can use function bult_in to demodulate signal by using SSB technic

```
demodulated_ssb=demod(ssb_sc_signal,CarrierFreq,15*Fs
,'amssb');
```

2.5. Effect addition of several values of F to LO

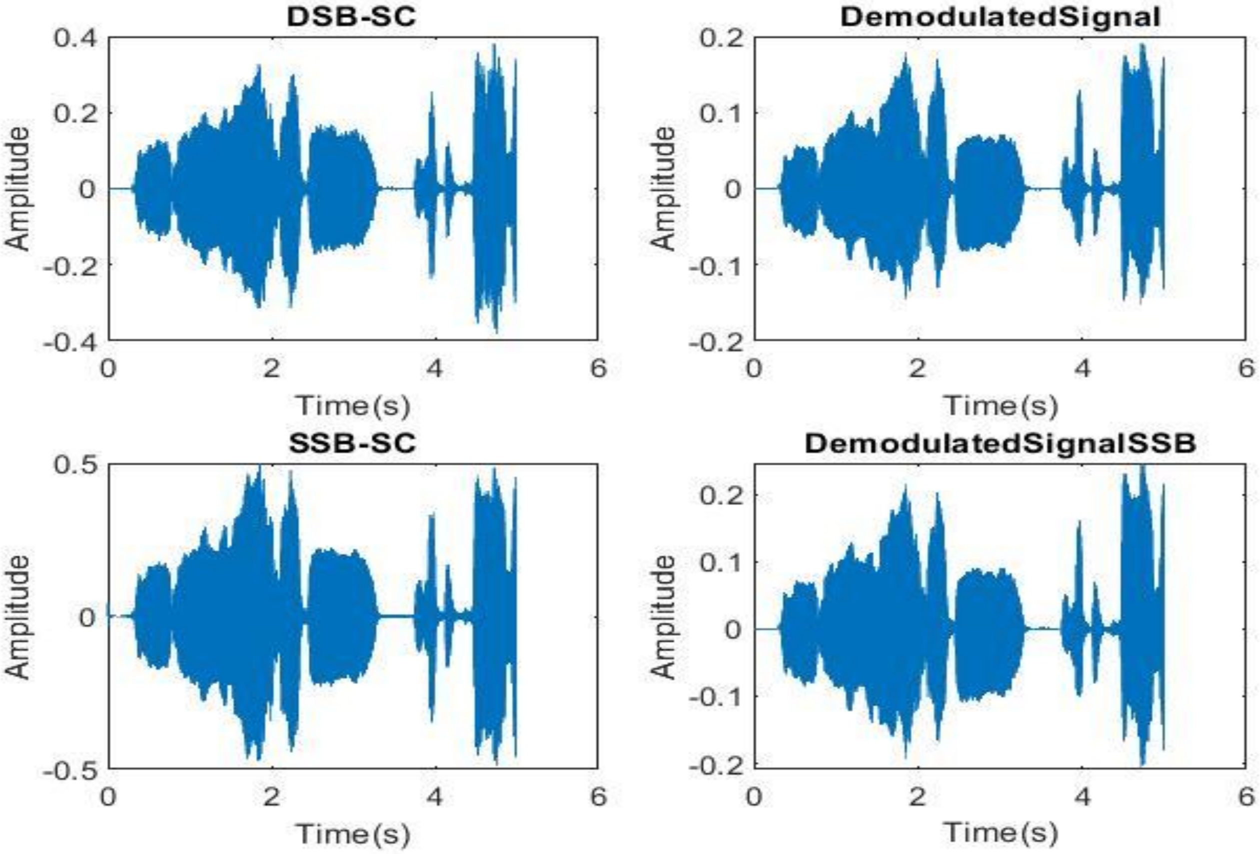
When I added several values of frequency offset to LO in both DSB-SC and SSB-SC the signal **got distorted**.

In DSB-SC there is **time varying phase**.

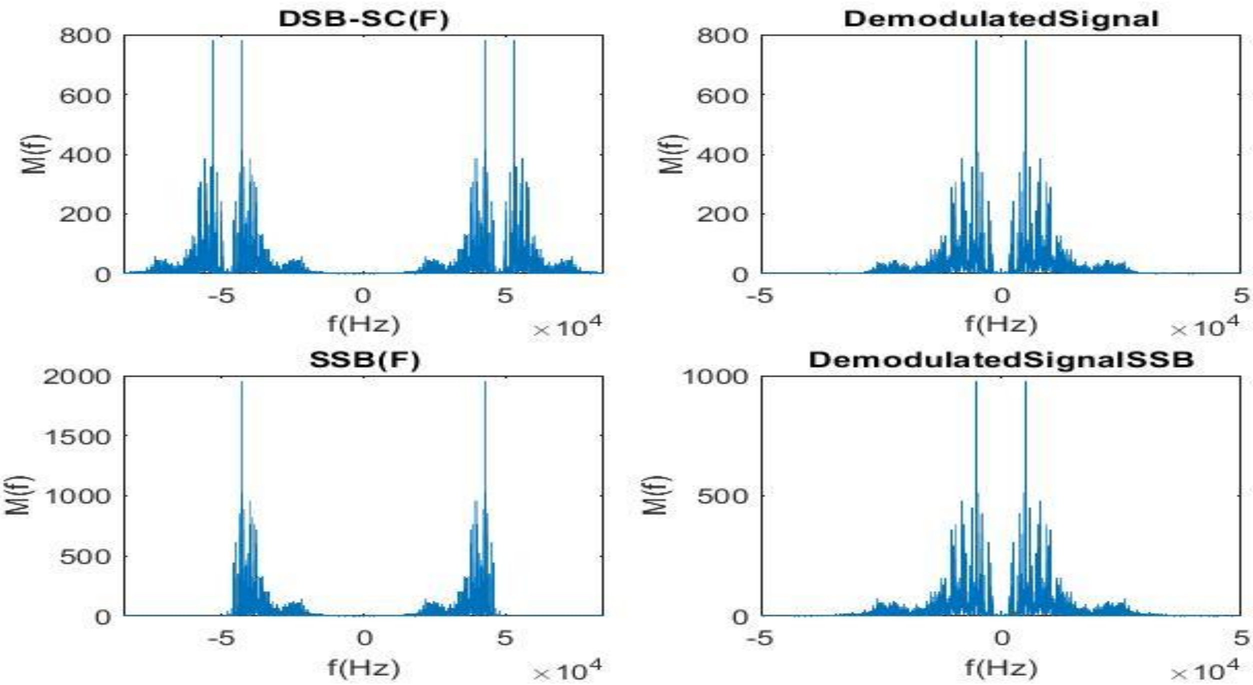
In SSB-SC there is a **phase shift for all frequencies**.

2.6. plot

2.6.1.Signals plot in time domain



2.6.2. Signals plot in frequency domain



3-design part 3

3.1. Frequency Modulation and Demodulation

```
kf=(beta1*2*pi*Fc)/(max(filtered_signal));

modulated_signal1 =
cos(2*pi*CarrierFreq*t1 + kf*cumsum(filtered_signal)/Fs);

envelope1 = abs(hilbert(diff(modulated_signal1)));

demodulated_signal1 = envelope1 - mean(envelope1);

demodulated_signal1 = LPF(demodulated_signal1,Fc,Fs);
```

kf1: Calculates the frequency deviation constant using the modulation index beta1.

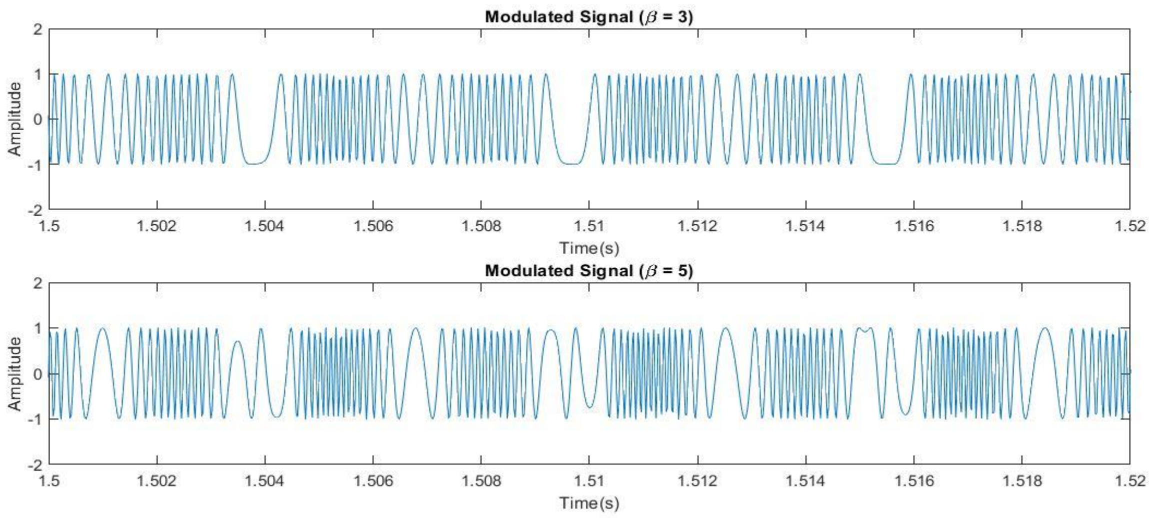
hilbert: Computes the analytic signal to find the envelope, a technique used in signal processing for demodulation.

diff: Computes the difference between successive elements of the modulated signal, enhancing the demodulation process.

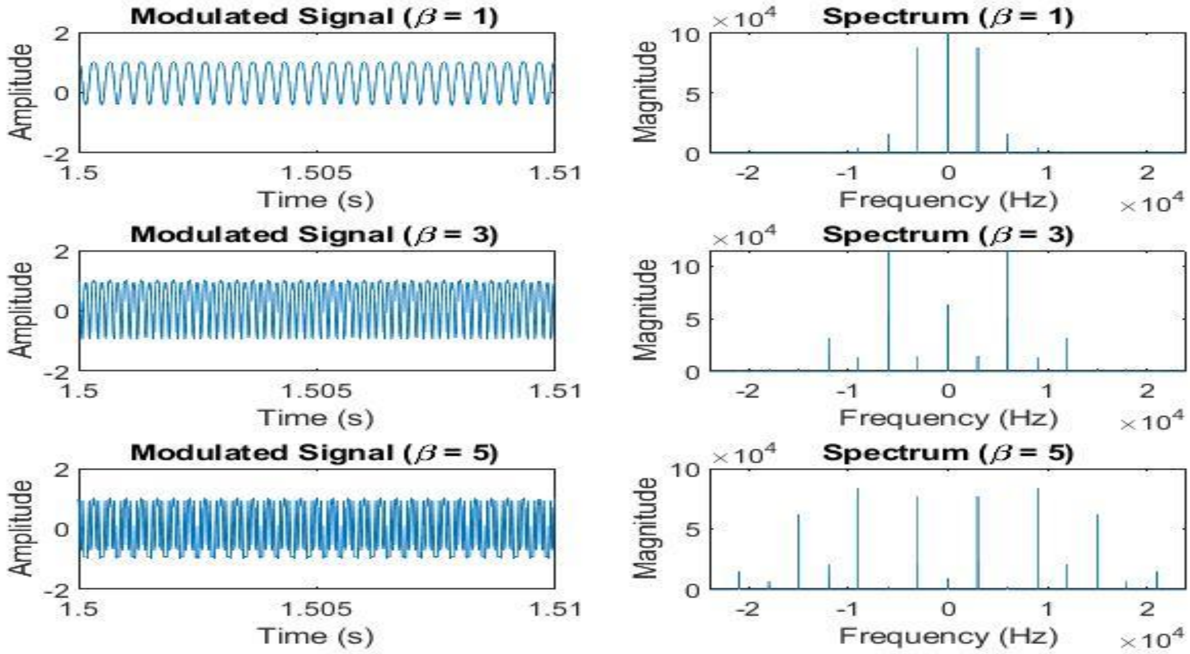
envelope1: The envelope of the differentiated signal, used to extract the original modulated message.

3.2. plot

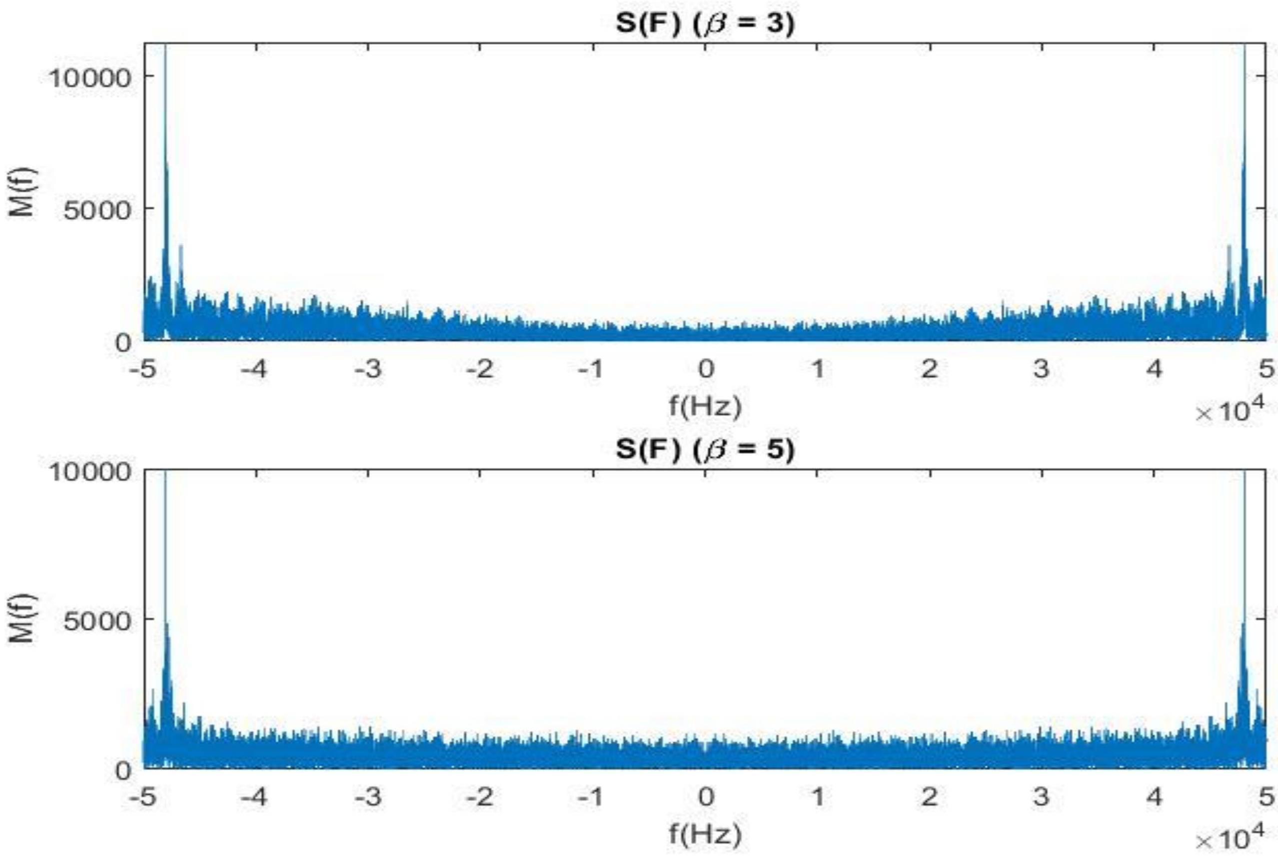
3.2.1. Signals plot in time domain



3.2.2. Single Tone signal versus different values of Beta



3.2.3. Signals plot in frequency domain



4.part 4

I tried different SNR levels on modulated signal and signal quality affected really bad under SNR 20 dB for $\beta = 3$ and under 25 dB for $\beta=5$ before adding LPF and after adding LPF the signal got really distorted at greater SNR level. I increased β from 10 to 60 and made SNR level constant = 30 dB, I noticed that the value of β at which threshold occurs is around 45

Conclusion

This project on communication systems covers a wide range of topics and implementation details, providing a comprehensive understanding of modulation techniques and signal processing.

The project concludes with a comprehensive analysis of each implementation part, offering insights into signal processing techniques and their applications in communication systems.

Overall, the project serves as a valuable resource for understanding modulation principles and their practical implementations.