ISTANBUL AYDIN UNIVERSITY ★ FACULTY OF ENGINEERING

VISUALISATION OF MODULATION METHODS IN COMMUNICATION THEORY

BACHELOR DEGREE THESIS

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JUNE 2014

<u>İSTANBUL AYDIN ÜNİVERSİTESİ ★ MÜHENDİSLİK FAKÜLTESİ</u>

HABERLEŞME TEAORİSİNDE KULLANILAN MODULASYON METODLARININ GÖRSELLEŞTİRİLMESİ

LİSANS TEZİ

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Mustafa DEMİRTAŞ, a Bachelor Degree**.** student of IAU **Faculty of Electric and Electronic Engineering ,**student ID B1205 020024, successfully defended the **thesis** entitled "Visualisation of Modulation Methods in Communication Theory", which he prepared after fulfilling the requirements specified in the associated legislations, before the jury whose signatures are below.

Thesis Advisor : **Prof. Dr. Hasan Hüseyin BALIK** ....
İstanbul Aydın University

Date of Defense : **13 June 2014**

FOREWORD

I would like to thank my supervisor Prof. Dr. Hasan Hüseyin BALIK for the suggestions, ideas and advise during the project. He encouraged me and talked to keep me on track many times.

I would also like to thank my friend Mr.Mehmet Ilgaz for his prominent support with his communication backround.

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JUNE 2014 Mustafa DEMİRTAŞ

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ABBREVIATIONS

AM	**:** Amplitude modulation
FM	**:** Frequency Modulation
ASK	**:** Amplitude Shift Key
PSK	**:** Phase Shift Key
BPSK	**:** Binary Phase Shift Key
PWM	**:** Pulse Width Modulation
PCM	**:** Pulse Code Modulation

VISUALISATION OF MODULATION METHODS IN COMMUNICATION THEORY

SUMMARY

This thesis includes implemented visual graphics through MATLAB, MATLAB codes and general information about heartland of digital communication systems, modulation types which directly assigns the bit rate of transmission. Modulation is a process of combining low-frequency message signal with higher-frequency carrier signal to produce a new one. The message signal modulates the carrier one. To emphasize the aim of process, antenna height becomes a noticeable parameter. If unmodulated signals had been transmitted, the wavelength of signals would be extremely huge due to the low frequencies so that antenna size would be measured by kilometers. Also, higher frequency means higher energy, so long distance transmissions are enabled by modulation. Modulation can be performed by several different forms. Each different form offers users several advantages such as higher data speed, lower bit error rate and noiseless transmission. Thus, modulation type should be well-chosen with respect to demand.

HABERLEŞME TEAORİSİNDE KULLANILAN MODULASYON METODLARININ GÖRSELLEŞTİRİLMESİ

ÖZET

Bu tez, dijital haberleşme sistemlerinin can damarı haline gelen ve iletimin hızını doğrudan belirleyen bazı modülasyon çeşitlerinin MATLAB'de uygulanmış görselleri, MATLAB kodları ve bu modülasyonlar hakkında genel bilgi içermektedir. Modülasyon, düşük frekanslı mesaj işaretinin yüksek frekanslı taşıyıcı işaret ile birleştirilip yeni bir işaret üretilmesidir. Mesaj işareti, taşıyıcı işareti modüle eder. Bu işlemin yapılmasındaki temel amaç, anten boyudur. Eğer işaretler modülasyona uğramadan iletilseydi, mesaj işareti düşük frekanslı olduğu için dalga boyu çok büyük seviyelerde olacaktı ki, bu da anten boylarının kilometrelerce ölçülmesi anlamına geliyor. Ek olarak, yüksek frekanslı işaretler yüksek enerjiye sahip olduğu için uzun mesafeli haberleşme de modülasyon sayesinde mümkün hale gelmiş oluyor. Modülasyon, 4 farklı şekilde gerçekleştirilebilir. Her bir modülasyon çeşidi, kullanıcıya yüksek bilgi hızı, düşük hata oranı ve gürültüsüz iletim gibi çeşitli avantajlar sunar. Bu sebepten ötürü, sistemde kullanılacak modülasyon çeşidi, ihtiyaca göre doğru tercih edilmelidir.

1. INTRODUCTION

One way to communicate a message signal whose frequency spectrum does not fall within that fixed frequency range, or one that is otherwise unsuitable for the channel, is to change a transmittable signal according to the information in the message signal. This alteration is called *modulation*, and it is the modulated signal that is transmitted. The receiver then recovers the original signal through a process called *demodulation*. Modulation is a process by which a *carrier signal* is altered according to information in a *message signal*. The *carrier frequency,* denoted *Fc*, is the frequency of the carrier signal. The *sampling rate, Fs,* is the rate at which the message signal is sampled during the simulation.

The frequency of the carrier signal is usually much greater than the highest frequency of the input message signal. The Nyquist sampling theorem requires that the simulation sampling rate *Fs* be greater than two times the sum of the carrier frequency and the highest frequency of the modulated signal, in order for the demodulator to recover the message correctly.

For a given modulation technique, two ways to simulate modulation techniques are called *baseband* and *passband*. Baseband simulation requires less computation. The

MATLAB_ Communication toolbox supports baseband simulation for digital modulation and passband simulation for analog modulation. In this tutorial, baseband simulation will be used.

Here are the some explanations and brief information about some modulation technics. Besides, some script codes which are being used in related modulatin process in MATLAB

2. AMPLITUDE (AM) MODULATION

Modulation is the process of varying a higher frequency carrier wave to transmit information. Though it is theoretically possible to transmit baseband signals (or information) without modulating it, it is far more efficient to send data by modulating it onto a higher frequency "carrier wave." Higher frequency waves require smaller antennas, use the available bandwidth more efficiently, and are flexible enough to carry different types of data. AM radio stations transmit audio signals, which range from 20 Hz to 20 kHz, using carrier waves that range from 500 kHz to 1.7 MHz. If we were to transmit audio signals directly we would need an antenna that is around 10,000 km! Modulation techniques can be broadly divided into analog modulation and digital modulation. Amplitude modulation (AM) is one form of analog modulation.

The carrier signal is generally a high-frequency sine wave. There are three parameters of a sine wave that can be varied: amplitude, frequency, and phase. Any of these can be modulated, or varied, to transmit information. A sine wave can be mathematically described by a sine or cosine function with amplitude A_c, frequency f_c, and phase φ.

$$\underbrace{A_c}_{\text{Amplitude}}\cos(2\pi \underbrace{f_c}_{\text{Frequency}} t + \underbrace{\phi}_{\text{Phase}})$$

The carrier signal is modulated by varying its amplitude in proportion to the message, or baseband, signal. The message signal can be represented by

$$m(t) = M_b \cos(2\pi f_b t + \phi)$$

and the carrier signal can be represented by

$$c(t) = A_c \cos(2\pi f_c t + \phi)$$

To make the equations simpler, assume that there is no phase difference between the carrier signal and the message signal and thus φ = 0.

The modulated signal can be represented by multiplying the carrier signal and the summation of 1 and the message signal, as shown below.

$$A_c\left(1+m\left(t\right)\right)\cos\left(2\pi f_c t\right)$$

<u>Matlab code for AM:</u>

```
clc;
clear all;
close all;
t=0:0.001:1;
set(0,'defaultlinelinewidth',2);
A=5;%Amplitude of signal
fm=input('Message frequency=');%Accepting input value
fc=input('Carrier frequency=');%Accepting input value (f2>f1)
mi=input('Modulation Index=');%Modulation Index
Sm=A*sin(2*pi*fm*t);%Message Signal
subplot(3,1,1);%Plotting frame divided in to 3 rows and this fig appear at 1st
plot(t,Sm);
xlabel('Time');
ylabel('Amplitude');
title('Message Signal');
grid on;
Sc=A*sin(2*pi*fc*t);%Carrier Signal
subplot(3,1,2);
plot(t,Sc);
xlabel('Time');
ylabel('Amplitude');

title('Carrier Signal');
grid on;
Sfm=(A+mi*Sm).*sin(2*pi*fc*t);%AM Signal, Amplitude of Carrier changes to
(A+Message)
subplot(3,1,3);
plot(t,Sfm);
xlabel('Time');
ylabel('Amplitude');
title('AM Signal');
grid on;
```

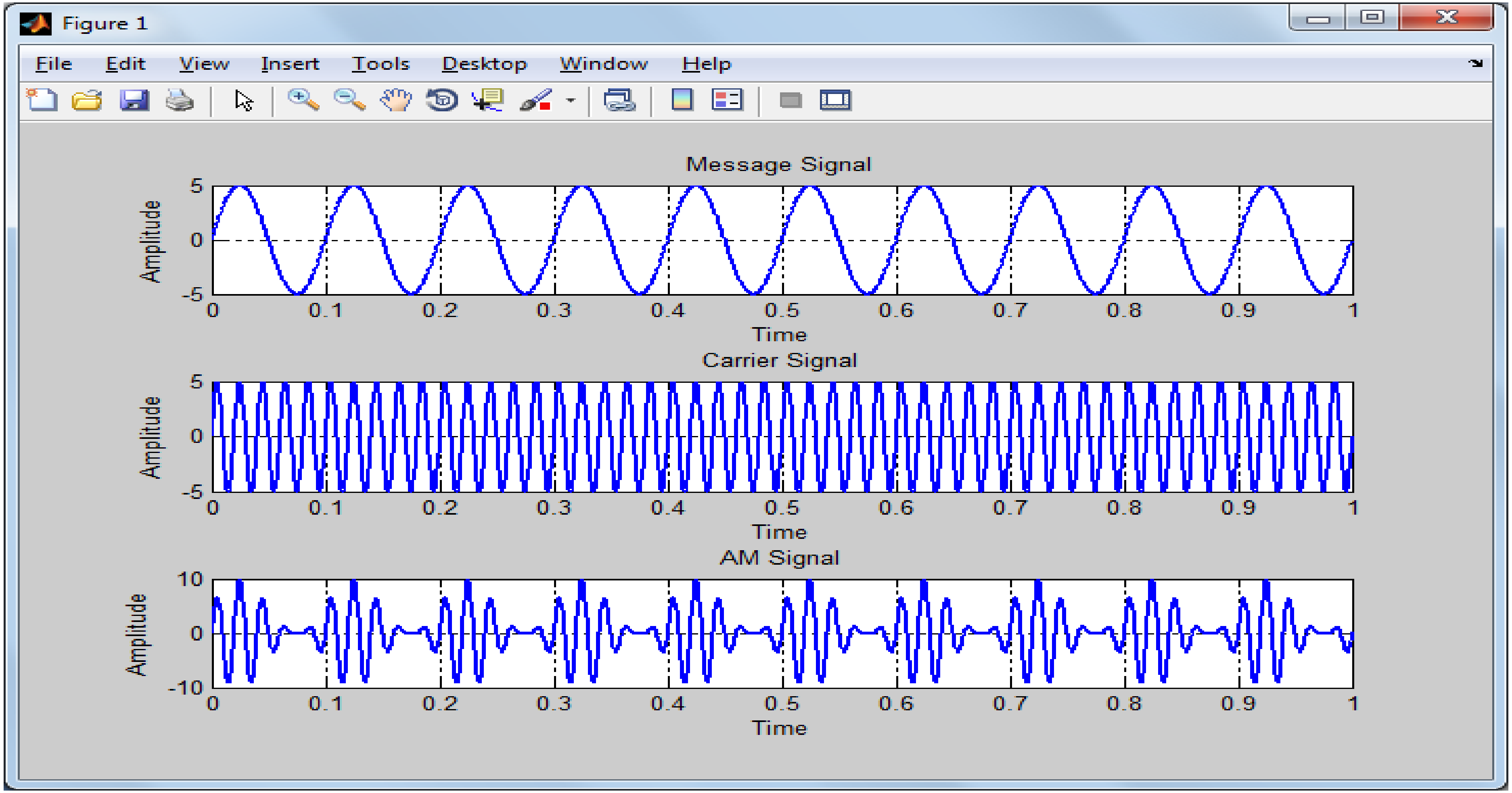


Figure 2.1 : AM Modulatin for the values : Message Frequency: 10 Carrier Frequency:50 Modulation Index:1

3. FREQUENCY (FM) MODULATION

Frequency Modulation (FM) is a form of modulation in which changes in the carrier wave frequency correspond directly to changes in the baseband signal. FM is considered an analog form of modulation because the baseband signal is typically an analog waveform without discrete, digital values

Let the modulating signal be

$$e_m(t) = E_m Sin(w_m t)$$

and the carrier signal be

$$e_c(t) = E_c Sin(w_c t)$$

then the modulating signal e(t) is expressed as

$$e(t) = E_c sin\big((w_c t) + msin(w_m t)\big)$$

where ‘m’ is the modulation index.

Matlab Code for FM

```
clc;
clear all;
close all;
fm=input('Message Frequency=');
fc=input('Carrier Frequency=');
mi=input('Modulation Index=');
t=0:0.0001:0.1;
m=sin(2*pi*fm*t);
subplot(3,1,1);
plot(t,m);
xlabel('Time');
ylabel('Amplitude');

title('Message Signal');
grid on;
c=sin(2*pi*fc*t);
subplot(3,1,2);
plot(t,c);
xlabel('Time');
ylabel('Amplitude');
title('Carrier Signal');
grid on;

y=sin(2*pi*fc*t+(mi.*sin(2*pi*fm*t)));%Frequency changing w.r.t Message
subplot(3,1,3);
plot(t,y);
xlabel('Time');
ylabel('Amplitude');
title('FM Signal');
grid on;
```

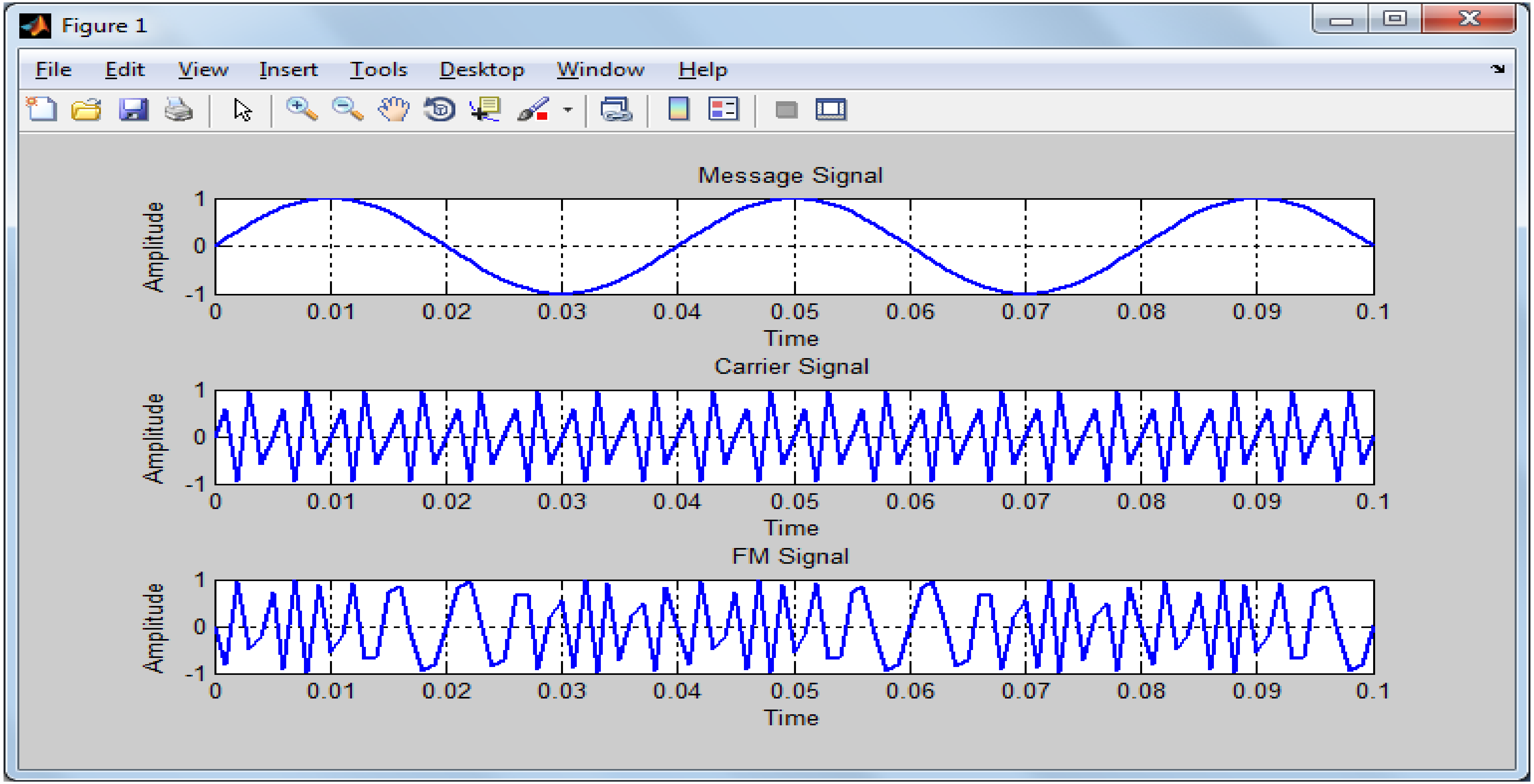


Figure 3.1 FM Modulatin for the values : Message Frequency: 25 Carrier Frequency:40 Modulation Index:1

4.AMPLITUDE SHIFT KEY (ASK) MODULATION

In this method the amplitude of the carrier assumes one of the two amplitudes dependent on the logic states of the input bit stream. A typical output waveform of an ASK modulator is shown in Fig. 1.

A binary amplitude-shift keying (BASK) signal can be defined by

$$s(t) = A\, m(t) \cos 2\pi fct \qquad 0 < t < T$$

where A is a constant, m(t) = 1 or 0, fc is the carrier frequency, and T is the bit duration. It has a power P = A2/2, so that A = 2P.

Thus equation can be written as

$$s(t) = \sqrt{} 2P \cos 2\pi fct, \qquad 0 < t < T$$

In ASK modulation, discontinuities may be seen at transition points. This is an undesired case because discontinuities result in signals having an unnecessarily wide bandwidth. Removing discontinuities are possible by using an envelope detector however, ASK system unfortunately does not include a constant envelope. This makes the process more difficult since envelope detection is also necessary for linearity in modulation. By the lack of constant envelope, linearity must be provided by using a power amplifier which increases the power consumption. Also, in order for removing discontinuities band limiting is necessarily introduced before transmission. The digital message or the modulated signal may be band limited. If linearity had not be obtained and the signal was passed through non-linear amplifier, sidelobes of signals could be larger enough to interfere with adjacent signals. So, a wider bandwidth or a higher power will be required for transmission to prevent interference

Matlab Code for ASK

```
clear all;
clc;
close all;
F1=input('Enter the frequency of carrier=');
F2=input('Enter the frequency of pulse=');
A=3;%Amplitude
t=0:0.001:1;
x=A.*sin(2*pi*F1*t);%Carrier Sine wave
u=A/2.*square(2*pi*F2*t)+(A/2);%Square wave message
v=x.*u;
subplot(3,1,1);
plot(t,x);
xlabel('Time');
ylabel('Amplitude');
title('Carrier');
grid on;
subplot(3,1,2);
plot(t,u);
xlabel('Time');
ylabel('Amplitude');
title('Square Pulses');
grid on;subplot(3,1,3);
plot(t,v);
xlabel('Time');
ylabel('Amplitude');
title('ASK Signal');
grid on;
```

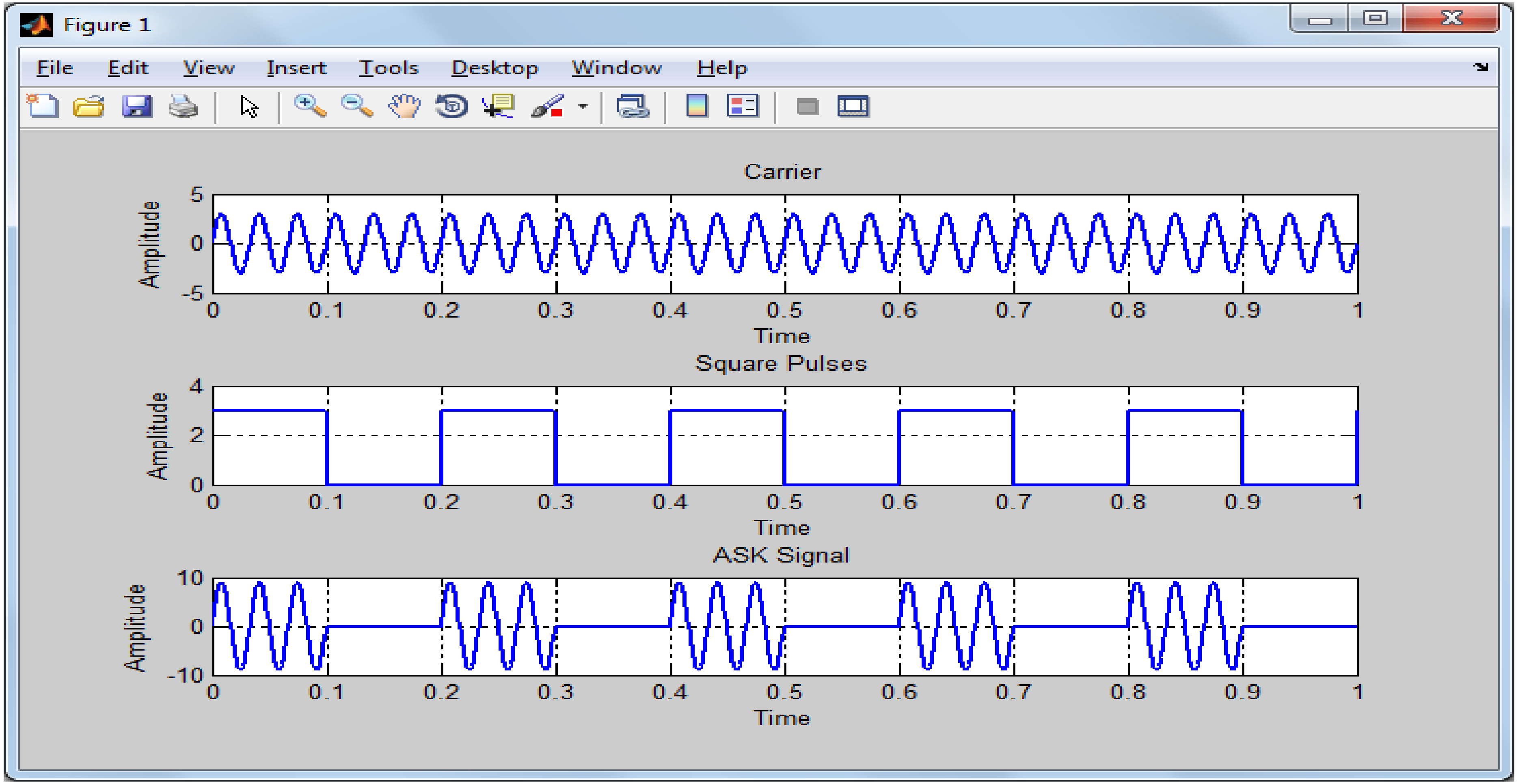


Figure 4.1 ASK Modulatin for the values : Message Frequency: 5 Carrier Frequency:30

5. **PHASE SHIFT KEY (PSK) MODULATION**

Phase shift keying is a digital modulation scheme which is extensively used in communication systems. PSK conveys data by changing the phase of the carrier signal, while amplitudes and frequency remain constant.
The message signal changing between one and zero cases creates a phase reversal of 180^o phase shifts.

The above figure is the modulation of "1 1 0 1 0 0 1 1 1 0".
However, a difficulty is noticed here in which the receiver can not determine the exact phase of transmitted signal to sign whether it is space or one condition. To overcome it, the phase changes for binary one case, while remaining constant for the case zero. Moerover, this idea was improved for special PSK forms, in which the phase changes 90 degree for a direction in one case and turns 90^o to another direction in zero case. So, this retains totally a 180^o phase reversal between one and zero cases which gives a distinct change for zero as shown obviously in figure.
PSK contains several sub-groups explained below.

.

Matlab code for PSK

```
clear all;
clc;
close all;
set(0,'defaultlinelinewidth',2);
A=5;
t=0:.001:1;
f1=input('Carrier Sine wave frequency =');
f2=input('Message frequency =');
x=A.*sin(2*pi*f1*t);%Carrier Sine
subplot(3,1,1);
plot(t,x);
xlabel('time');
ylabel('Amplitude');
title('Carrier');
grid on;
u=square(2*pi*f2*t);%Message signal
subplot(3,1,2);
plot(t,u);
xlabel('time');
ylabel('Amplitude');
title('Message Signal');
grid on;

v=x.*u;%Sine wave multiplied with square wave
subplot(3,1,3);
plot(t,v);
axis([0 1 -6 6]);
xlabel('t');
ylabel('y');
title('PSK');
grid on;
```

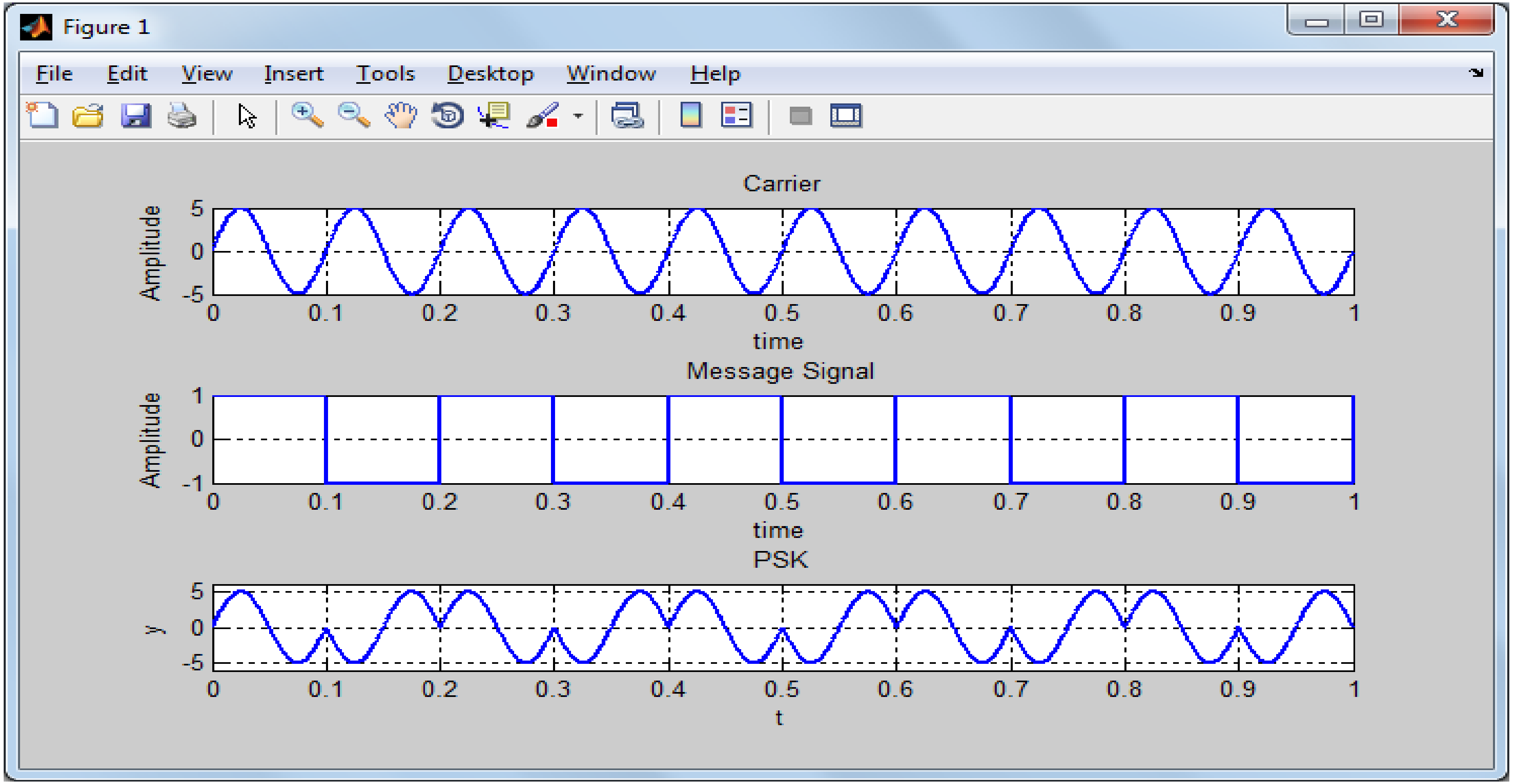


Figure 5.1 PSK (BPSK) Modulatin for the values : Message Frequency: 5 Carrier Frequency:10

5.1 BPSK MODULATION

Binary phase shift keying is the simplest form of PSK. As understood from the binary term, just two phases by a 180^{o} separation are used for transmission, which is also named 2PSK. By a brief reminding, number of phases were equal to 2^{bits}. Thus, only one bit can be transmitted in BPSK as shown below mathematically and demonstrated by constellation diagram;

$$S_{BPSK}(t)=\sqrt{\frac{2E_b}{T_b}}\cdot\cos\left(2\pi f_c t+\pi(1-n)\right), \qquad n=0,1 \tag{3.5}$$

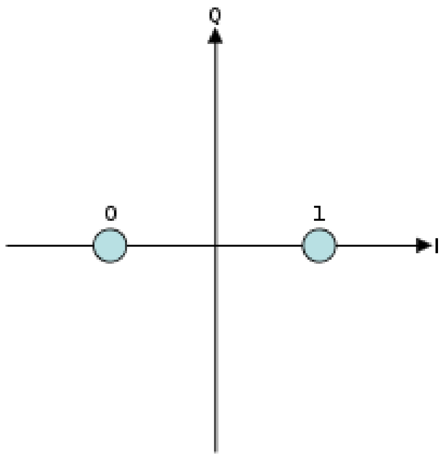


Figure 3.7. Constellation diagram for BPSK [11]

BPSK is the strongest form of PSK, since by taking the highest level of noise and distortions, it prevents the demodulator to reach a correct decision. Demodulator hereby, can not tell which constellation point is which, so data is always differently encoded prior to modulation.

However, the bit rate of BPSK is a dark side of system. To remember that only one bit can be transmitted by a phase, this type is too low in which users may suffer from..

BPSK has the strongest BER performance compared to others. The reason is that just 2 phases are used for transmission which can be separated greatly. As expected, the noise ratio will reduce while the distance between the symbols are increased, so bit error rate will be also at least. However, the system is inefficient because of the low bit rate. Despite its BER performance, data transmission is very slow here, as 1 bit per phase which causes to be unpreferred in new technology systems.

6. PULSE WIDTH MODULATION(PWM)

Pulse width modulation (PWM) is a powerful technique for controlling analog circuits with a microprocessor's digital outputs. PWM is employed in a wide variety of applications, ranging from measurement and communications to power control and conversion.

An analogue signal has a continuously varying value, with infinite resolution in both time and magnitude. A nine-volt battery is an example of an analog device, in that its output voltage is not precisely 9V, changes over time, and can take any real-numbered value. Analogue signals are distinguishable from digital signals because the latter always take values only from a finite set of predetermined possibilities, such as the set {0V, 5V}. Analogue voltages and currents can be used to control things directly, like the volume of a car radio. In a simple analogue radio, a knob is connected to a variable resistor. As you turn the knob, the resistance goes up or down. As that happens, the current flowing through the resistor increases or decreases. This changes the amount of current driving the speakers, thus increasing or decreasing the volume. An analogue circuit is one, like the radio, whose output is linearly proportional to its input. By controlling analogue circuits digitally, system costs and power consumption can be drastically reduced.

In a nutshell, PWM is a way of digitally encoding analogue signal levels. Through the use of high-resolution counters, the duty cycle of a square wave is modulated to encode a specific analogue signal level. The PWM signal is still digital because, at any given instant of time, the full DC supply is either fully on or fully off. The voltage or current source is supplied to the analog load by means of a repeating series of on and off pulses. The PWM signal is still digital because, at any given instant of time, the full DC supply is either fully on or fully off. The voltage or current source is supplied to the analog load by means of a repeating series of on and off pulses.

To illustarte the point, in figure 6.1, when sine is greater than sawtooth, PWM is high. On the other hand, when sine is less than sawtooth,PWM is low.PWM toggles when sine equals to sawtooth.

Matlab code for PWM

```
clc;
clear all;
close all;
F2=input('Message frequency=');
F1=input('Carrier Sawtooth frequency=');
A=5;
t=0:0.001:1;
c=A.*sawtooth(2*pi*F1*t);%Carrier sawtooth
subplot(3,1,1);
plot(t,c);
xlabel('time');
ylabel('Amplitude');
title('Carrier sawtooth wave');
grid on;
m=0.75*A.*sin(2*pi*F2*t);%Message amplitude must be less than Sawtooth
subplot(3,1,2);
plot(t,m);
xlabel('Time');
ylabel('Amplitude');
title('Message Signal');
grid on;
n=length(c);%Length of carrier sawtooth is stored to ‘n’
for i=1:n%Comparing Message and Sawtooth amplitudes
if (m(i)>=c(i))
    pwm(i)=1;
else
    pwm(i)=0;
end
end
subplot(3,1,3);
plot(t,pwm);
xlabel('Time');
ylabel('Amplitude');
title('plot of PWM');
axis([0 1 0 2]);%X-Axis varies from 0 to 1 & Y-Axis from 0 to 2
grid on;
```

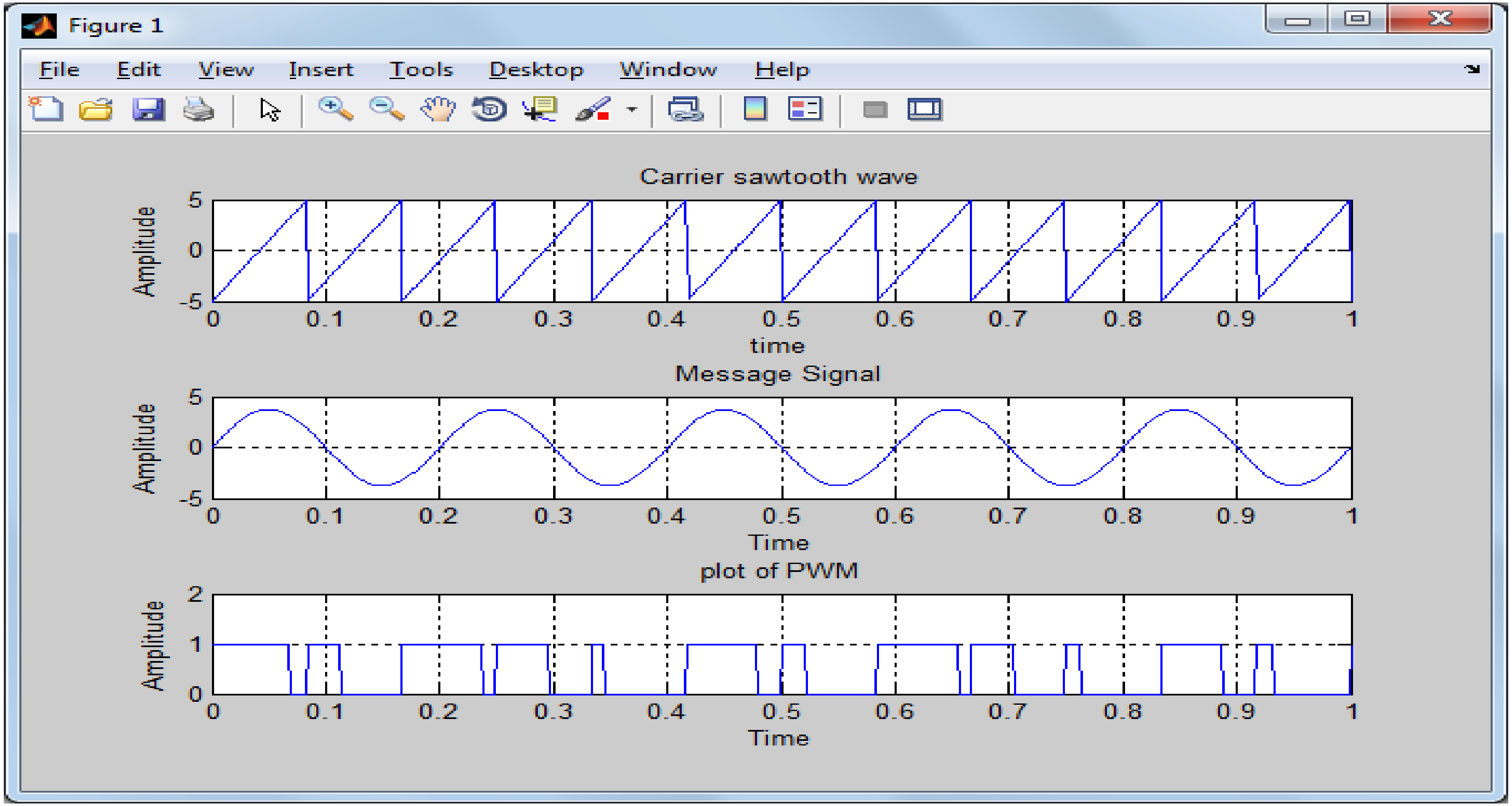


Figure 6.1 PWM Modulatin for the values : Message Frequency: 5 Carrier Frequency:12

7. PULSE CODE MODULATION(PCM)

Pulse Code Modulation, referred to as PCM. It is a kind of encoding that changes analog voice signal into digital signal. In the late 1970 s, with the appearance of pulse code modulation encoder and decoder for super-large-scale integrated circuit as well as the development of optical fiber communication, digital microwave communications, satellite communications, PCM has been gradually widely used. At present, the PCM has become a key technology in digital communication.

PCM modulation mainly includes sampling, quantization and encoding process. Sampling changes the continuous analog signals into the discrete time continuous amplitude sampling signals; Quantification changes the discrete time continuous amplitude sampling signals into the discrete time discrete amplitude digital signals; Coding makes the quantified signals into the output binary code groups. International standard PCM code groups (telephone voice) adopt eight-level codes represents a sampling value [1]. From the view of modulation concept in communication, it can be considered that, the PCM encoding process is analog signal modulating a binary pulse sequence, the carrier is pulse sequence, and modulation changes pulse sequence as none or "1", "0", therefore PCM is made into pulse code modulation. Pulse code modulation process as shown in Figure 1.Encoded PCM code groups, via digital channels, can be directly transmitted by baseband or microwave, light wave carrier modulated pass band. At the receiving end, the binary code group inversely transforms into the reconstruction analog signal $\hat{x}(t)$. In the demodulation process, generally uses the sampling hold circuit, so the low pass filter adopts $x/\sin x$ type frequency to response to compensate for frequency distortion $x/\sin x$ introduced by sampling hold circuit is.

Pre-filtering is to limit the original speech signal frequency band within 300-3400 Hz standard long-distance analog telephone frequency band. Due to the original speech band is around 40-10000 Hz, so pre-filter can made out certain band distortion.

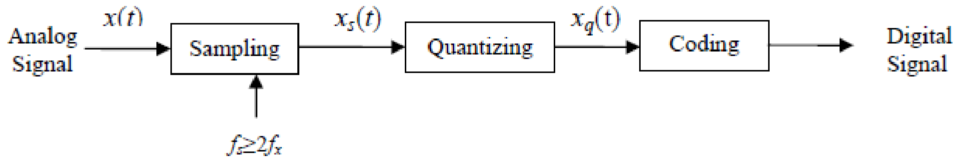


Figure 7.1 PCM Transmitter Block Diagram

7.1 Sampling:

It is the process of obtaining an instantaneous value of the input analog signal amplitude at regular intervals. The signal $m(t)$ entering the sampler is band-limited to B Hz. Usually there exists a filter with bandwidth B Hz prior to the sampler to cutoff the out-of-band components. The sampling rate must be sufficiently large so that the analog signal can be reconstructed from its samples $m_s(n)$ with sufficient accuracy. The input analog signal is sampled at a rate higher than the Nyquist rate to allow for some guard-band. Thus, the sampling frequency, f_s, can be restricted by

$$f_s \geq 2B$$

The sampling period is T_s second/sample and it is related to the sampling frequency by

$$f_s = 1/\,T_s$$

7.2 Uniform Quantization

It is the process of converting the voltage level of the sampled amplitude to the voltage value of the nearest standard level, or quantization level. At the end of this stage, the signal $m_q(n)$ will be represented discretely in both time and amplitude. In uniform quantization, the quantization regions are chosen to have equal length. Assumed that the range of the input samples is $[-m_{max}, +m_{max}]$. In uniform quantization, all quantization regions except the first and last ones are of equal length, which is denoted by Δ, and the number of quantization levels L is an integer power of 2. From this, the length of the quantization region is given by

$$\Delta = \frac{2m_{\max}}{L}$$

The quantization levels are chosen to be the midpoints of the quantization regions and therefore, the quantization error at the n^{th} sample is given by

$$e(n) = m(t) - m_q(n)$$

where $t = nT_s$ and $e(n)$ has a uniform probability density function on the interval $[-\Delta/2, +\Delta/2]$.

Matlab code for SAMPLING AND QUANTIZING

```
% MATLAB script for Illustrative Example uniform_ex1.m
clc;clear;close all
% echo on
global q

t      = [0:pi/100:2*pi];
m      = 10 * sin(t);

t_samp = [0:pi/4  :2*pi];
m_samp = 10 * sin(t_samp)

figure(1)
plot (t,m,'-b','linewidth', 2)
hold on
stem (t_samp,m_samp, '--ro')
hold on
legend('original signal','sampled signal');
axis([0 2*pi -15 15])

L=16;
[sqnr,m_quan,code]=uniform_pcm(m_samp,L)

figure(2)
plot (t,m/max(m),'-b','linewidth', 2)
hold on
stem (t_samp,m_samp/max(m), '--ro')
hold on
for i=1:L
    plot (t,q(i)* ones(1,length(t)),'--g');hold on
end
legend('normalized original signal','normalized sampled signal');
axis([0 2*pi -1.5 1.5])
echo off
```

Matlab code for mutual function used in SAMPLING and PCM

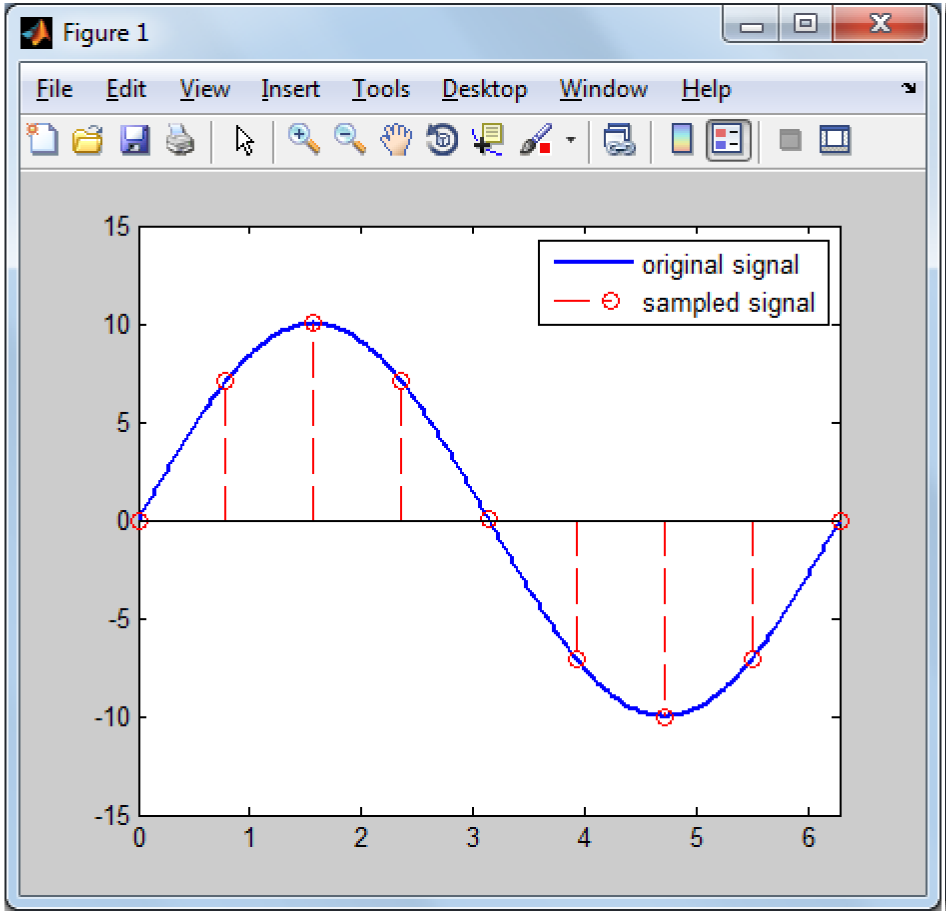
```
function [sqnr,m_quan,code]=uniform_pcm(m_samp,L)
global q
%UNIFORM_PCM    uniform PCM encoding of a sequence
%               [sqnr,m_quan,code]= uniform_pcm(m_samp,L)
%               m_samp   = input sampled sequence.
%               L        = number of quantization levels (even).
%               sqnr     = output SQNR (in dB).
%               m_quan   = quantized output before encoding.
%               idx_quan = index of quantized output.
%               code     = the encoded output.

m_max    = max(abs(m_samp));     % Find the maximum value of m_samp.
m_quan   = m_samp/m_max;         % Normalizing m_samp.
idx_quan = m_quan;               % Quantization index.
delta    = 2/L;                  % Quantization step.
q        = delta.*[0:L-1];       % Define quantization regions.
q        = q-((L-1)/2)*delta;    % Centralize all quantization levels
                                 % around the x-axis.

for i=1:L
  m_quan(find((q(i)-delta/2 <= m_quan) & (m_quan <= q(i)+delta/2)))=...
  q(i).*ones(1,length(find((q(i)-delta/2 <= m_quan) & ...
  (m_quan <= q(i)+delta/2))));
  idx_quan(find(m_quan==q(i)))=(i-1).*ones(1,length(find(m_quan==q(i))));
end

m_quan;
idx_quan;

m_quan = m_quan * m_max;    % Release normalization for quantized values.
R      =ceil(log2(L));      % Define no. of bits per codeword.
code   = de2bi(idx_quan', R, 'left-msb'); % Generate codewords.
sqnr   = 20 * log10(norm(m_samp)/norm(m_samp - m_quan)); % Estimate SQNR.
```



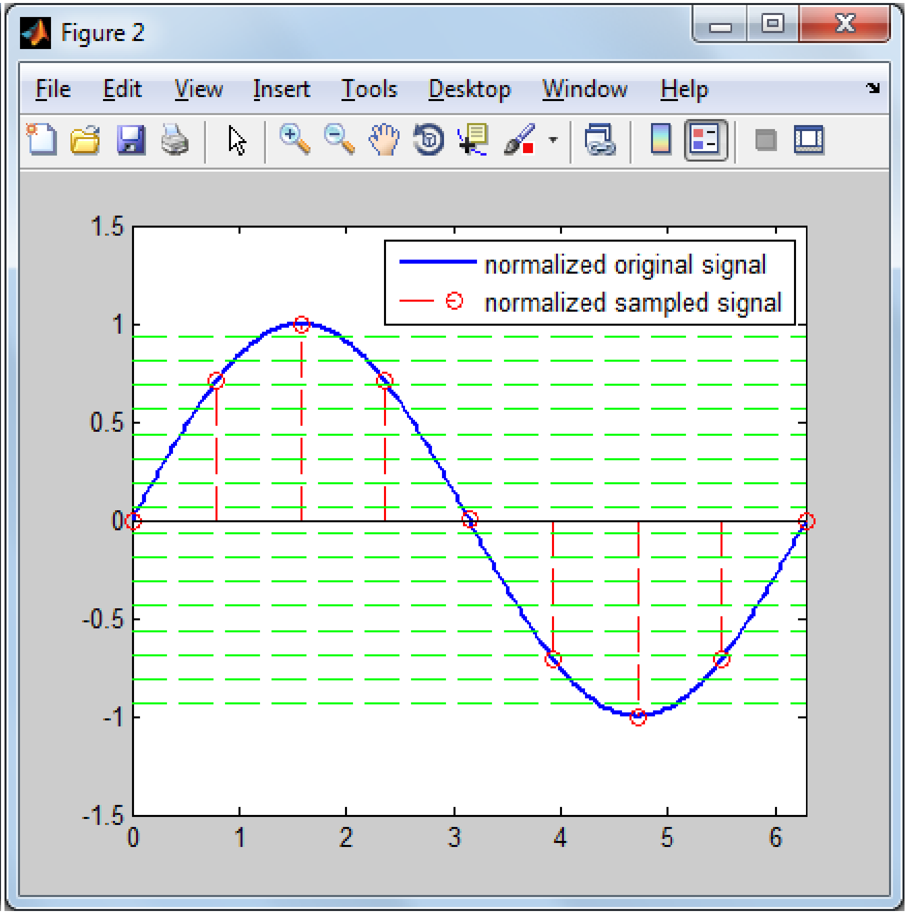


Figure 7.2 Sampled Signal

Figure 7.3 Normalized signal with its quantized values

7.3 Encoding:

It is the process of representing a particular quantization level of the analog signal with a *binary codeword.* After quantization, the quantized levels are encoded using *R* bits for each quantization level. The encoding scheme that is usually employed is *natural binary coding* (NBC), meaning that the lowest quantization level is mapped into a sequence of all 0's and the highest level is mapped into a level of all 1's.

Matlab code for PCM

```
% MATLAB script for Illustrative Example uniform_ex2.m
clc;clear;close all
echo on

t_samp = [0:0.01:10];
m_samp = sin(t_samp);

[sqnr_8L,m_quan_8L,code_8L]=uniform_pcm(m_samp,8);
[sqnr_16L,m_quan_16L,code_16L]=uniform_pcm(m_samp,16);

pause   % Press a key to see the SQNR for L = 8.
sqnr_8L

pause   % Press a key to see the SQNR for L = 16.
sqnr_16L

pause   % Press a key to see the plot of the signal and its
        % quantized versions.
t = t_samp; m = m_samp;
plot(t,m,'-',t,m_quan_8L,'-.',t,m_quan_16L,'-',t,zeros(1,length(t)), ...
   'linewidth', 2)
legend('Original','Quantized (L=8)','Quantized (L=16)','Location', ...
   'SouthEast');

% pause % Press a key to see the first 5 samples, corresponding quanited
%       % values, and corresponding codewords with 8 quantization levels
% m(1:5)
% m_quan_8L(1:5)
% code_8L(1:5,:)
%
% pause % Press a key to see the first 5 samples, corresponding quanited
%       % values, and corresponding codewords with 16 quantization levels
% m(1:5)
% m_quan_16L(1:5)
% code_16L(1:5,:)
echo off
```

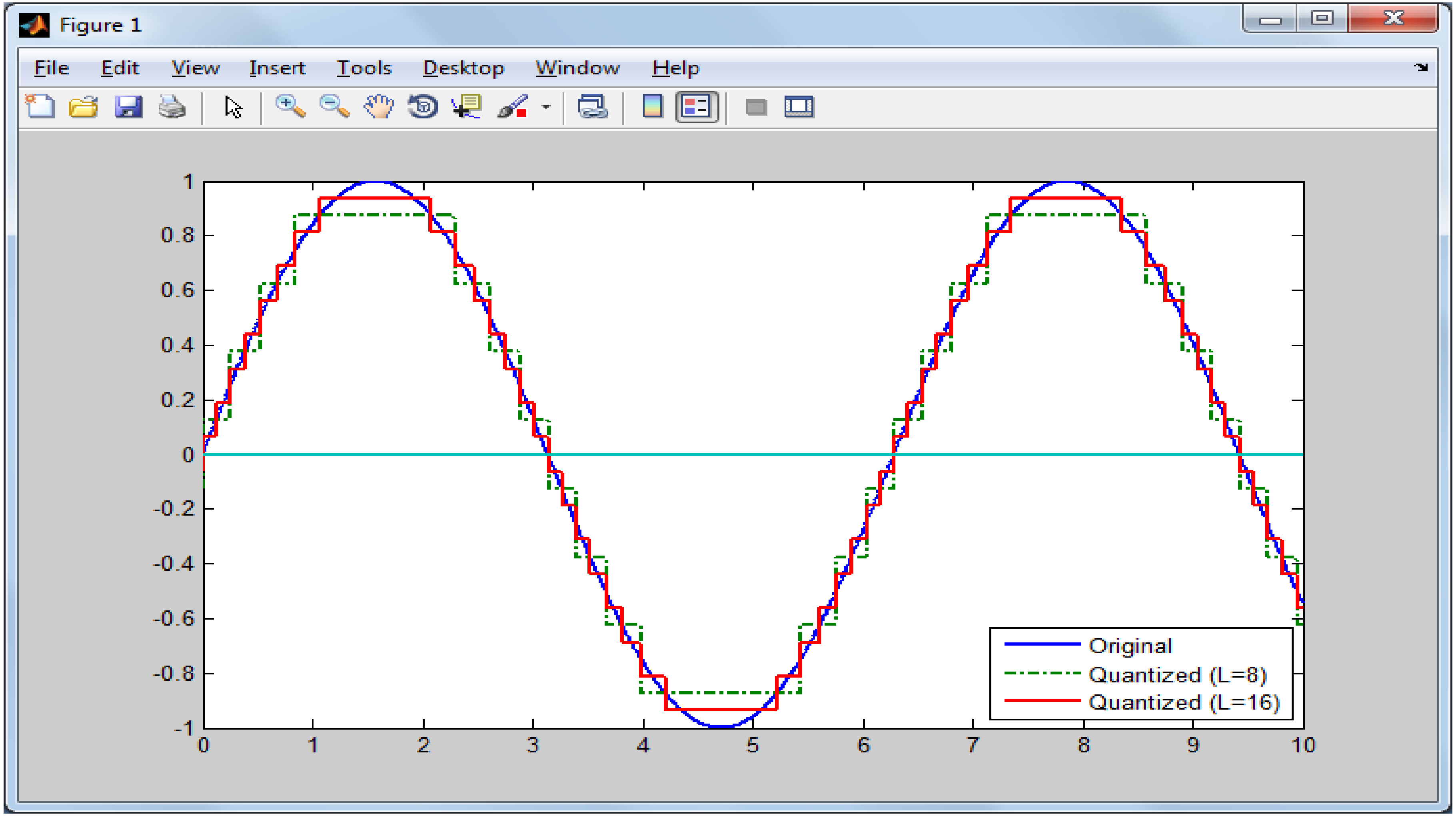


Figure 7.4 Quantized signal using 8 and 16 quantization levels.

REFERENCES

Simon Haykin (2001). Digital Communication,4th edition, McMaster University

Mohammed N. Sabet, Assc Prof Erhan Ince. 2010). Pulse Code Modulation Coding.Eastern Mediterranean University

Prof.V.Venkata Rao. (2004). Digital Transmission of Analog Signals.

Lou Frenzel (2010).Electronics Explained

Ola Jetlund. (2009). Digital Signal Processing Lab,Oslo University College.

Hewlett Packard. (1997). Digital Modulation in Communication Systems

Van Trees, H. L., *Detection, Estimation, and Modulation Theory, Part I,* New York: John Wiley

Fatima Faydhe AL-Azawi, Zainab Faydh AL-Azawi, Rafed shaker AL- Fartosy (2014). Design and simulation of PCM

Proakis, J., *(1989)Digital Communications*, 2nd ed., New York: McGraw-Hill

Lucky, R., J. Salz, and J.Weldon,(1968) *Principles of Data Communications*, New York: McGraw-Hill,

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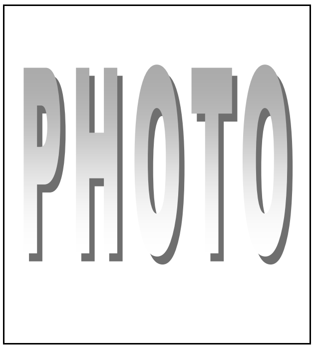

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