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Improving design of High Pass Sigma Delta modulation.

By

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ABSTRACT

Analog to digital conversion have been widely used in the measurement, telecommunications, digital signal processing. Sigma delta modulation is a digital signal processing, or DSP method for encoding analog signals into digital signals as found in an ADC with high resolution and low cost. So sigma delta modulation has become a very useful and widely applied technique for high performance analog to digital (**A/D**) conversion of narrow band signals. Through the use of over sampling and feedback, the quantization error which is suppressed in a narrow band signal at the output of the modulator an analog-to-digital converter based on the sigma delta modulation is capable of providing a very high resolution (**HPSDM**).

This thesis describes the theory and design of continuous- time (**CT**) **HPSDM** for **A/D** conversion. The essential idea of the work is the design of the 2nd and 4th order **HPSDM** and get the output using fast Fourier transform (**FFT**) by an applied software .

الملخص

تحويل التناظري إلى رقمي أستخدم علي نطاق واسع في الاتصالات ومعالجة الإشارة الرقمية. تعديل سيقما دلنا هو أحد طرق المعالجة الرقمية للإشارة وطريقة لترميز الإشارة التناظرية إلى رقمية كما هو موجود في تحويل التناظري إلى رقمي مع دقة عالية وتكلفة منخفضة. أصبحت هذه التقنية مفيدة جدا وفي نطاق واسع للحصول علي أداء عالي في تحويل التناظري إلى رقمي من خلال استخدام نظرية أكثر من أخذ العينات (OSR) والتعددية العكسية وبهذا خطأ التكمية يجمع في نطاق الخرج الضيق.

توضح هذه الأطروحة نظرية تصميم تعديل دلنا سيقما لتحويل التناظري إلى رقمي. والفكرة الأساسية للعمل مبنية علي تصميم الرتبة الثانية والرابعة للدلتا سيقما والحصول علي خرج باستخدام تحويل فورير السريعة عن طريق تطبيق برنامج بلغة الماتلاب.

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List of Acronyms

Analog –to- Digital	A/D
Adaptive Delta Modulation	ADM
Amplitude Modulation	AM
Adaptive Sigma Delta Modulation	ASDM
Amplitude Shift Keying	ASK
High Pass Filter	HPF
High Pass Sigma Delta Modulation	HPSDM
Continuous Wave	CW
Delta Modulation	DM
Differential Pulse Code Modulation	DPCM
Dynamic Range	DR
Double Side Band -Suppressed Carrier	DSB-SC
Effective resolution	ER
Frequency Modulation	FM
Frequency Shift Keying	FSK
Low Pass Filter	LPF
Low Pass Sigma Delta Modulation	LPSDM
Low Pass Filter	LPF
Matrix Laboratory	MATLAB
Noise Transfer Function	NTF
Over Sampling Ratio	OSR
Pulse amplitude modulation	PAM

Pulse Code Modulation	PCM
Phase Modulation	PM
Phase Shift Keying	PSK
Pulse Position Modulation	PPM
Pulse Time Modulation	PTM
Pulse Width Modulation	PWM
Radio Frequency	RF
Signal-to-Noise Ratio	SNR
Signal Transfer Function	STF
Transfer Function	TF

List of Symbols

Peak amplitude of input signal	A
Bandwidth	BW
Quantization error	e
Quantization noise power	e_{rms}
Difference signal(error signal)	$e(t)$
Quantization noise	$E(s)$
Signal bandwidth	f_m
Sampling frequency	f_s
The loop filter transfer function	$H(s)$
Integrator constant	K_1, K_2
Original signal	$m(t)$
Approximation signal	$\tilde{m}(t)$
Additive white noise	$N(t)$
Power of quantization error	P_q
Analog input signal	$X(s)$
Quantization level	Δ

INTRODUCTION

1.1 Introduction

Analog-to-Digital Converters (ADC) are key components in almost all modern communication systems. High speed and high resolution ADCs are required in many applications. Nowadays, there exist a lot of techniques that allow a very high precision conversion, such as the analog-to-digital conversion based on the sigma delta modulation and oversampling [1]. However, the process of converting the analog signal to a digital one often limits the overall system speed. In contrast, there are other techniques that provide a very high speed conversion but have just an average or even low resolution. Associating the high speed and the high resolution in one ADC has been long time a tough challenge in the research of analog-to-digital conversion techniques. The principle of sigma delta modulation was discovered long time ago and during the last ten years the converters based on this principle have become very popular. However, this popularity does not concern the "high-pass sigma delta modulator". The reason may lie in its difficult implementation and the uncertainty about its Stability and performance in the presence of circuit non idealities. So over sampling converters, or those that use a sampling frequency much larger than the Nequist frequency of the signal being converted have become very popular during the last few years. So sigma delta modulation it has several application such as industrial process , audio application and temperature and pressure measurements. In this chapter clarify , the main objective of the thesis, general outline of the thesis and organization of this work are presented[1,6,10].

1.2 Aim of the thesis

The main objective of this thesis is summarized in the following points:

- Review on digital communication/ modulation systems.
- Theory and basic theoretical analyses design of high order HPSDM and block analysis.
- Implement and simulate high order HPSDM.
- Conclusion and Recommendation of thesis.

1.3 Organization of the work.

In addition to an introductory chapter this thesis contains five chapters, which are briefly summarized below:

Chapter II " Review on Communication Systems". this chapter describes the over all review on communication systems and modulation techniques, and this types.

Chapter III "Review on Low Order Low Pass Sigma Delta Modulation" this chapter introduce the theory behind of the low pass sigma-delta modulation (**LPSDM**), **LPSDM** parameters, transfer function of **LPSDM** . finally, presents **LPSD** modulator / demodulator.

Chapter IV " High Pass Sigma Delta Modulation" this chapter presents the theory behind of High Pass Sigma- Delta Modulation(**HPSDM**) and introduces first order **HPSDM** and it is transfer function.

Chapter V "Investigation of HP High Order Sigma- Delta Modulation" this chapter is the main chapter of this thesis, it gives brief explanation about the Matlab software simulation using fast forward transform to get speedup the output.

Chapter VI " Conclusion and Future Work" this chapter includes the main conclusion of this work and suggests some points as a future work.

Review of communication system

2.1 introduction.

Communication technique is known as the process, in which information is transferred from one point to another point in the free space Figure (2.1). Shows the general communication system block diagram .[1,2,5]

Which contains three main parts such as transmitter, communication channel and receiver. The function of these parts are summarized in the following points.

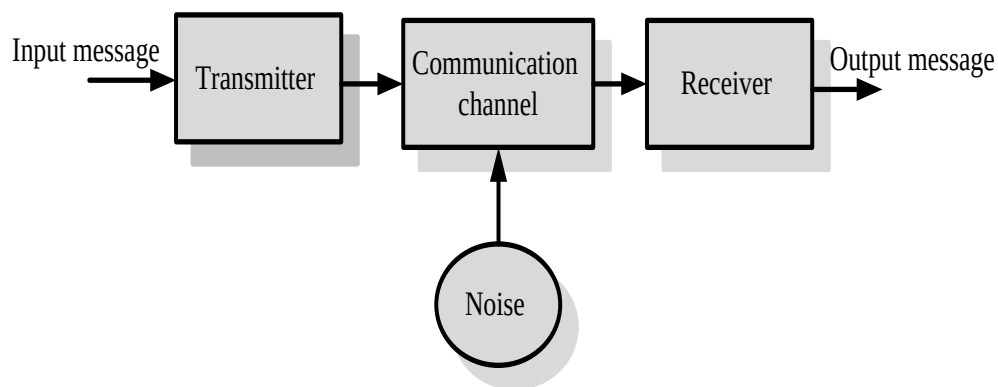


Figure (2.1) Schematic Diagram of General Communication System

- **Transmitter:** the main function of transmitter is to sending (transmitting) an analogue or digital information signal, over a physical media point-to-point or point-to -multipoint transmission. medium, either wired, optical fiber or free space.
- **Communication Channel :** The communications channel is the physical medium that, is used to passing the signal from the transmitter to the receiver. it may be pair or wire , a coaxial cable and optics....etc.

- **Receiver** : The function of the receiver is to recover the message signal contained in the received signal. The message signal is transmitted by carrier modulation, the receiver extract the message from the sinusoidal carrier. Besides performing the primary function of signal demodulation, the receiver also, signal filtering and noise suppression[3].

2.2Types of Communication System:

There are two main types of communication systems such as digital communication and analog communication . In this work only digital communication are explained briefly.[1]

The digital communication system are classified in two main type:

- Digital transmission.
- Digital radio.

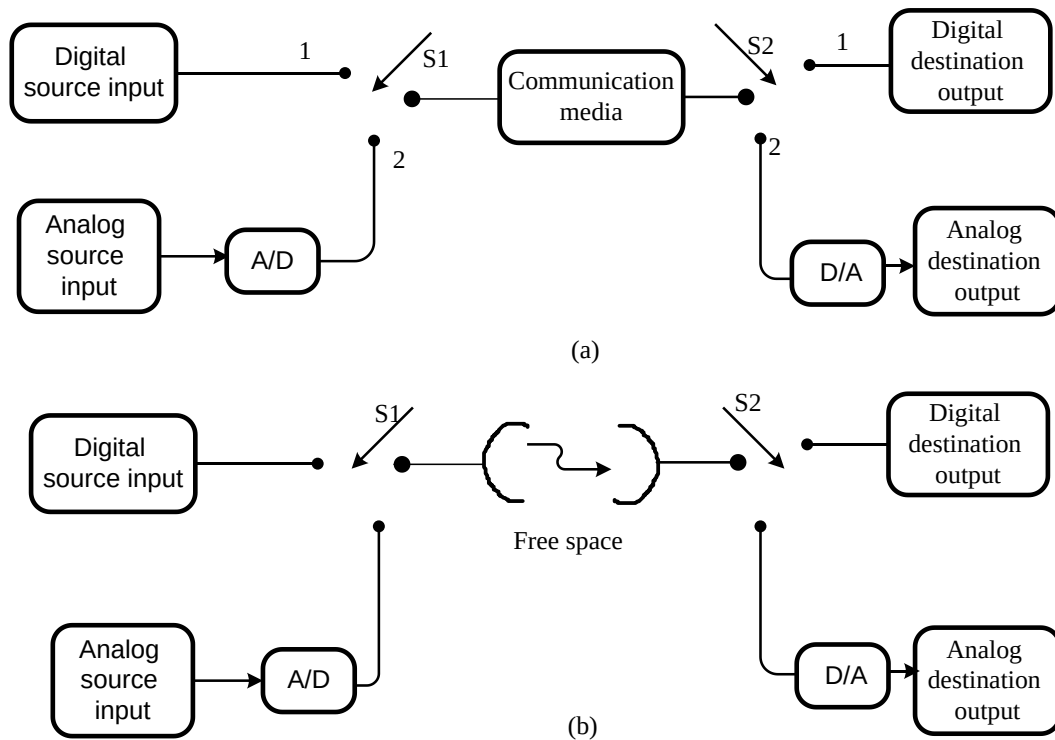
These types will be explained briefly in the following subsections:

2.2.1 Digital Transmission:

Figure (2.2a) represent the block diagram of digital transmission in that scheme the digital pulses is transmitted between two or more points in communication system . It require transmission medium between source and destination just like coaxial cable or pair of wire . If the information source analog then must be converted to digital pulse at transmission side and back to analog at receiver side[4,9] .

2.2.2 Digital Radio:

The digital radio shown in figure (2.2b) is require free space medium between source and destination. In this type of system the modulating input signal and demodulating out put signal are digital pulses[8].



Figure(2.2) Digital Communication System

(a) Digital Transmission.

(b) Digital Radio.

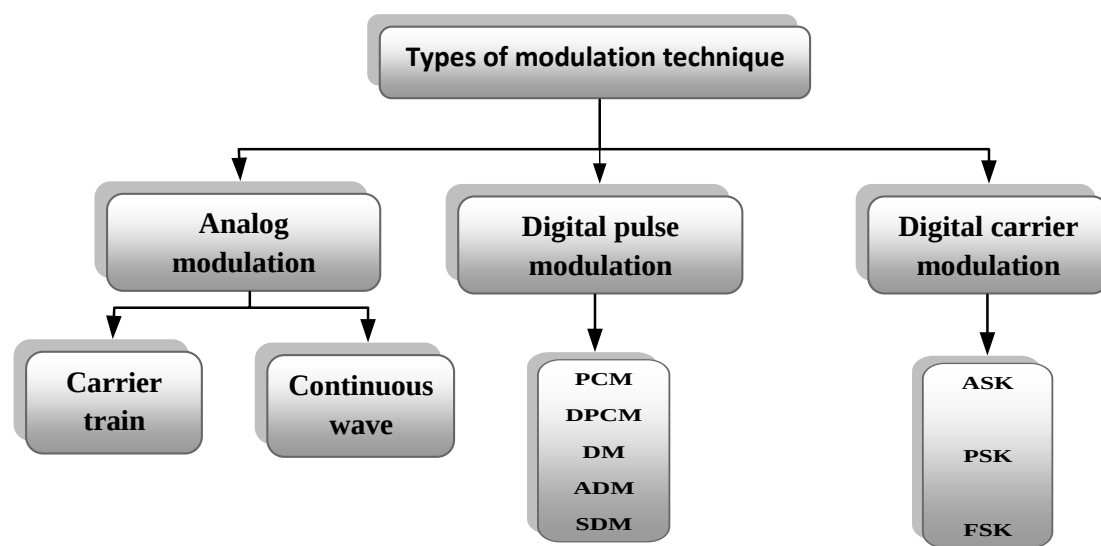
2.3 Modulation Techniques:

modulation is the process of varying characteristic of the carrier signal according to the message signal . In telecommunications, modulation is the process of conveying a message signal, for example a digital bit stream or an analog audio signal, inside another signal that can be physically transmitted .The modulation technique needed for several reasons such as Reduce the noise and interference, For frequency assignment , For multiplexing to over come equipment limitation andetc [7].

2.4 Types of Modulation Techniques:

There are several types of modulation technique according to the carrier signal it is analog modulation , digital pulse modulation and digital carrier modulation as shown in figure(2.3) .

If the type of modulation analog then the carrier is sinusoidal in the other hand if the modulation is pulse (digital) then the carrier is digital pulse .



Figure(2.3) Types of modulation technique.

The modulation technique can be classified in to three main types such as analog modulation ,digital pulse modulation and digital carrier modulation. These types will be described briefly in the following subsection.

2.4.1 Analog Modulation:

In this type of modulation, a sine wave is used as a carrier signal, that is modulates the message signal or data signal. In which, three parameters can be altered to get modulation they are amplitude modulation(AM), frequency modulation(FM) and phase modulation(PM)

In the analog pulse modulation the information represented by code group of digital continuous pulse and this types of modulation include pulse width modulation (**PWM**) ,pulse amplitude modulation (**PAM**) and pulse position modulation (**PPM**).

2.4.2 Digital Carrier modulation:

The digital signal is used to modulate one of the parameters of the carrier wave, This basic process is given the name "shift-keying" to differentiate it from the purely analog systems like AM and FM. As with analog modulation, there are three parameters of the carrier wave to vary and therefore three basic types of shift keying Amplitude Shift Keying (ASK) Frequency Shift Keying (FSK) and Phase Shift Keying (PSK).

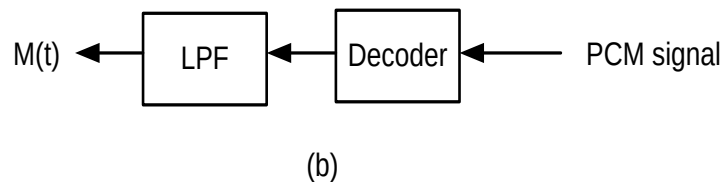
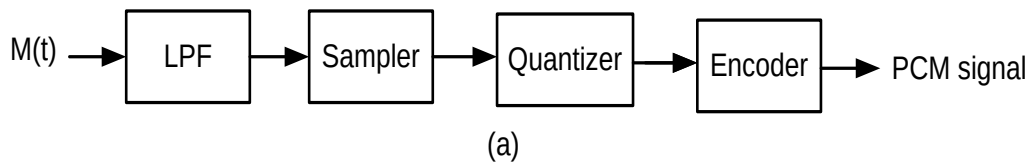
2.4.3 Digital Pulse modulation.

In the digital pulse modulation the information represented by code group of digital pulses and this types of modulation include the following types:

Pulse code modulation(PCM), Differential pulse code modulation (DPCM),Delta modulation(DM), adaptive delta modulation(ADM) and Sigma delta modulation(SDM).

(i) Pulse Code Modulation (PCM):

PCM is a system used to convert analog signal into digital form for transmission figure(2.4) shows block diagram of **PCM** modem.



figure(2.4) PCM Modem

(a)PCM Modulator

(b) PCM Demodulator

The analogue signals are first sampled using pulse amplitude modulation. The pulse amplitude modulation pulses are quantized and then encoded into a binary code that is transmitted as a digital stream. At the receiver these pulse code modulation (PCM) codes are decoded into pulse amplitude modulation (PAM) pulses that are then used to reconstruct the analogue waveform.

Quantization introduce error in the amplitude of sample .this quantization error is reduced as the number of quantization level increased .

There are two main reason to produce this type of noise:

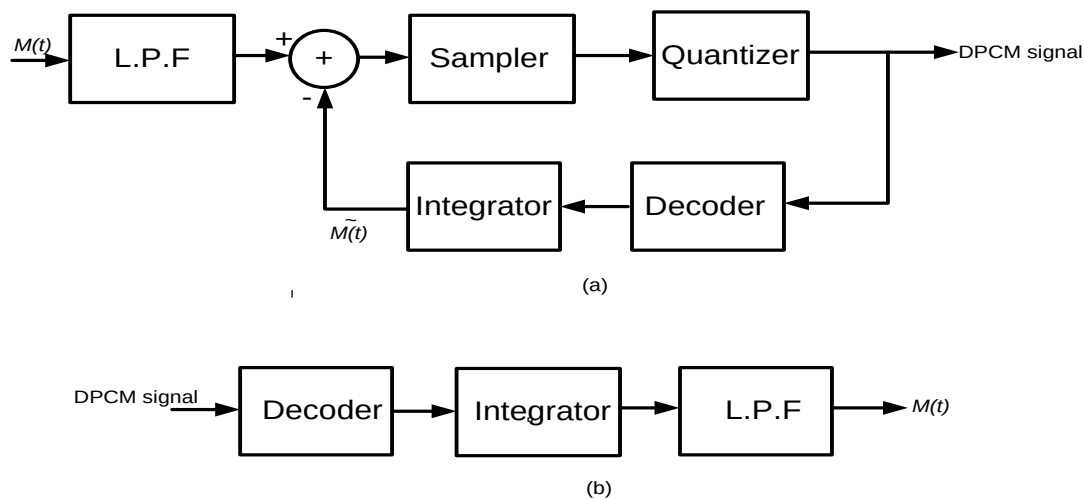
- The number of quantized level at the PCM transmitter tends to occur quantizing noise.
- The bit error are caused by channel noise[1,2,3].

(ii) Differential Pulse code modulation(DPCM).

The PCM system that uses differential quantizing schemes is known as differential PCM (DPCM) system. A differential PCM system that is

particularly simple to implement results when the difference signal is quantized into two levels after sampling, the output of the quantizer is represented by a single binary digit, which indicates the sign of the sample to sample difference.

Differential pulse code modulation employs increased circuit complexity to improve system performance. The improvement is achieved by using the idea of prediction to remove redundant symbols from incoming data stream. Figure (2.5) shows DPCM modem.



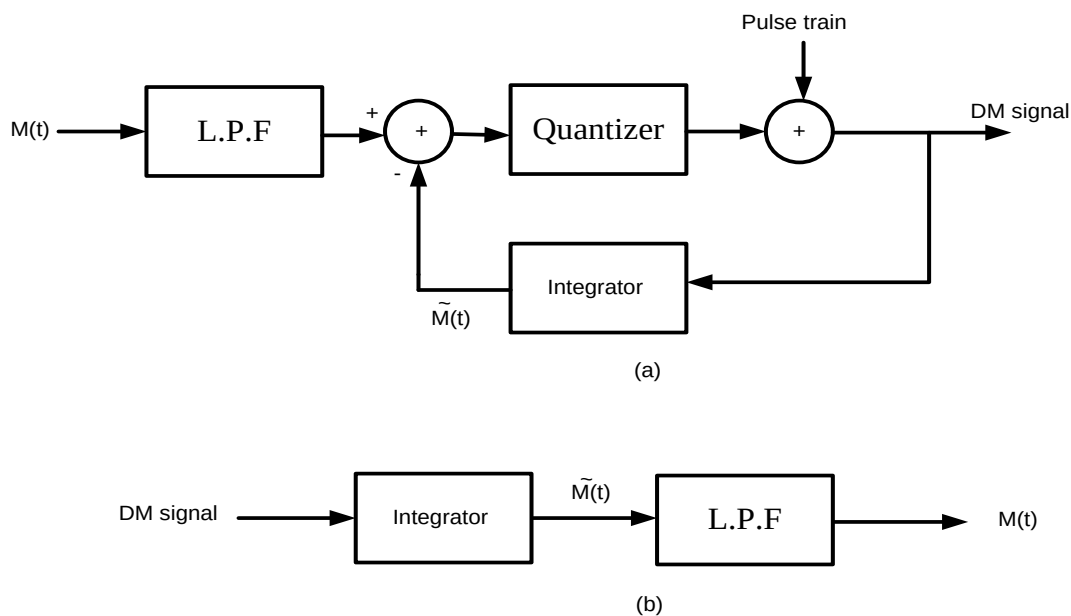
Figure(2.5)DPCM Modem
(a) Modulator
(b) Demodulator

The original signal $m(t)$ is first passed via a **LPF** to limit its bandwidth and get rid of any combined noise with the information signal. The difference signal is sampled, quantized, and coded, which are then transmitted. The coded signal is also fed back and used to generate an approximation signal. [1,2]

(iii) Delta Modulation(DM)

Delta modulation (DM) is one type of modulation technique in which the message signal is encoded into a sequence of binary digits. Delta modulation transmits only one bit per sample. That is the present sample

value is compared with the previous sample value and the indication, whether the amplitude is increased or decreased is sent. The input signal is approximated to step signal by the delta modulator. This step size is fixed. The difference between the input signal and the staircase approximated signal confined to two levels, If the difference is positive then the approximated signal is increased ,if the difference is decreased then the approximated signal is decreased. Figure (2.6) shows DM modem[11] .



Figure(2.6) DM Modem
(a) Modulator
(b) Demodulator

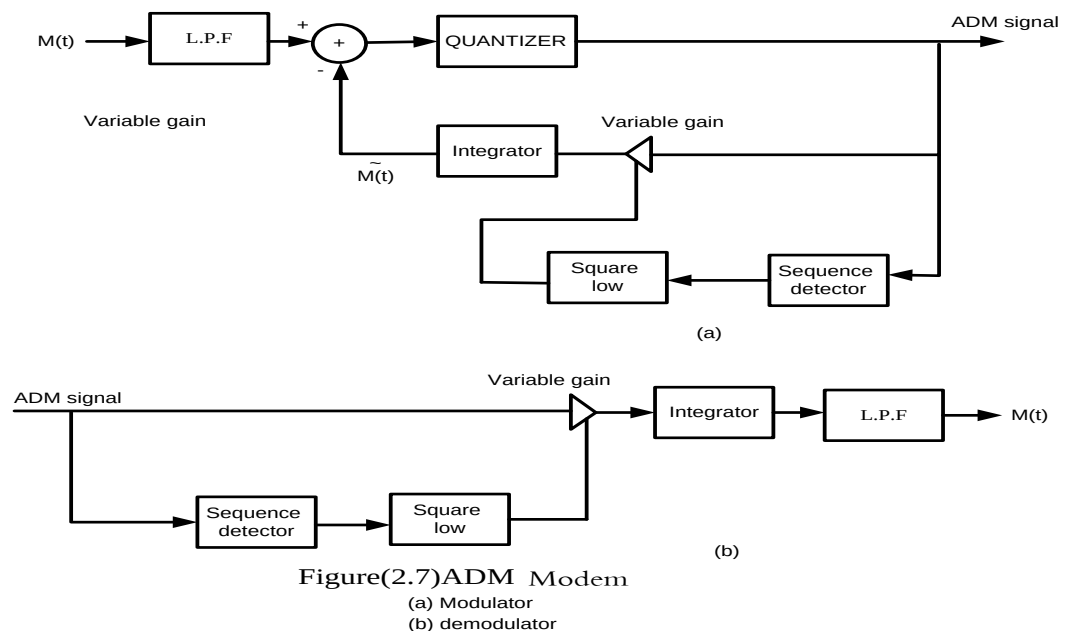
The transmitted signal have a large derivative then modulated signal can not follow the input signal which cause slope overload .

(iv) Adaptive Delta Modulation(ADM).

The DM suffers from one series drawback which is a dynamic range of amplitude of message signal $m(t)$ is too small because of the threshold and overload effects. This drawback can be getting rid of it by making a

DM modulator adaptive which is known as ADM as shown in the figure(2.7).

In ADM the step size Δ is varied according to the level of the input signal. A variation of DM which achieves better performance is adaptive delta modulation (ADM). it uses a variable quantum 'step size' which is small for a slowly changing waveforms and increase to a large value as the waveform changes steeply. The adaptive technique produces small quantization and over-load errors. The use of an ADM modulator needs that the receiver side be adaptive also. So that step size at the receiver change to match the change in Δ at the modulator[13,14].



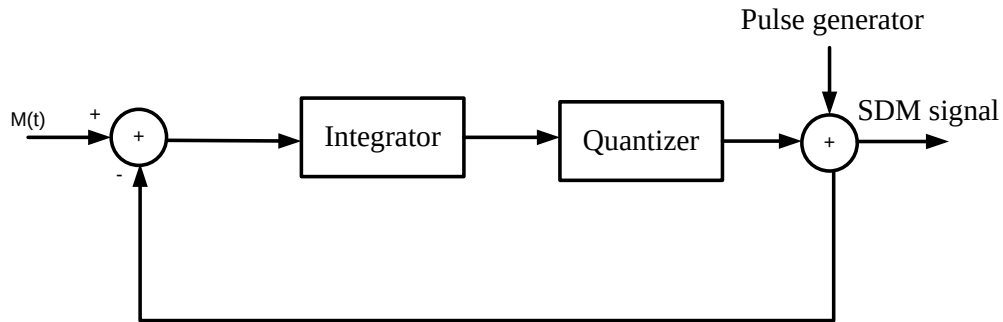
Figure(2.7)ADM Modem
(a) Modulator
(b) demodulator

(v) Sigma Delta Modulation(SDM).

Delta modulation has several disadvantages such as slope over load and cannot transmitting D.C signal to get rid of these disadvantage DM is modified to sigma delta modulation (**SDM**). Signal to noise ratio dependent on modulating frequency and so on. This technique is capable of transmitting D.C signal and more stability performance. Also to produce SDM signal can be done through modification of DM, by

moving the integrator from feedback to feed forward path prior to quantizer . Figure (2.8) shown the SDM modem.

In **SDM** more details will be given in next chapters.



(a)



(b)

Figure(2.8) SDM Modem
(a) Modulator
(b) Demodulator

REVIW ON SIGMA DELTA-MODULATION

3.1 introduction

Sigma delta modulation is a method for encoding analog signals into digital signals as found in an ADC. It is also used to transfer higher-resolution digital signals into lower-resolution digital signals as part of the process to convert digital signals into analog. a analog signal is integrated, or sampled, with a sampling frequency and subsequently quantized in a multi-level Quantizer into a digital signal. This process introduces quantization error noise. The first step in a delta-sigma modulation is delta modulation. By other word Delta-sigma modulation converts the analog voltage into a pulse frequency and is alternatively known as pulse density modulation or Pulse Frequency modulation.[13,14]

3.2 Sampling.

sampling is the reduction of a continuous signal to a discrete time signal. A common example is the conversion of a sound wave (a continuous signal) to a sequence of samples (a discrete-time signal).Also refers to a value or set of values at a point in time and/or space. By other word Sampling is an operation that is basic to digital signal processing and digital communications. Through use of the sampling process, an analog signal is converted into a corresponding sequence of samples that are usually spaced uniformly in time. Clearly, for such a procedure to have practical utility, it is necessary to choose the sampling rate properly, so that the sequence of samples uniquely defines the original analog signal. This is the essence of the sampling theorem, which is derived in what

follows.

If the signal $m(t)$ is band limited such as the highest frequency component is f_m and is sampled periodically with the frequency f_s , then it is possible to reconstruct the signal $m(t)$ from its samples $m_s(t)$ without any distortion if the sampling frequency is higher than or equal to $2f_m$, i.e., $f_s \geq 2f_m$, where the minimum theoretical sampling frequency $2f_m$ is called Nequist frequency .Figure (3-1) shows the analog, sampling pulses and sampled signals[11].

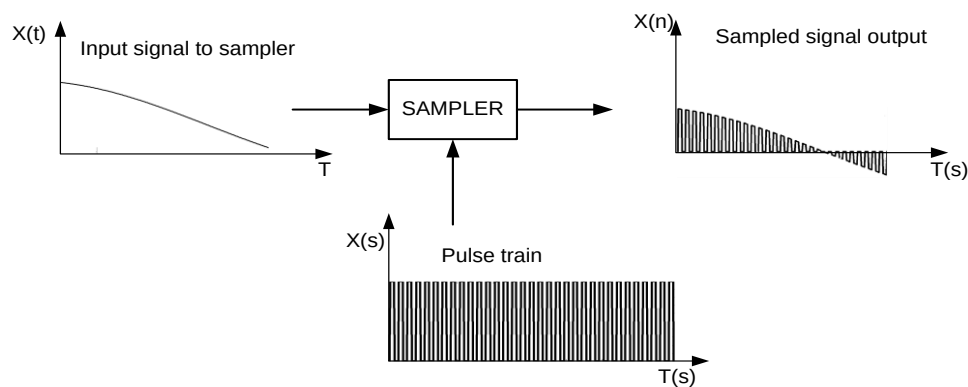


Figure (3.1) Sampling process

3.3 Quantization.

The Quantization is the process of converting a continuous analog audio signal to a digital signal with discrete numerical values. On other hand Quantization makes the range of a signal discrete, so that the quantized signal takes on only a discrete, usually finite, set of values as shown in figure (3.2.) Unlike sampling ,quantization is generally irreversible and results in loss of information . It therefore introduces distortion into the quantized signal that cannot be eliminated. One of the basic choices in quantization is the number of discrete quantization levels to use.

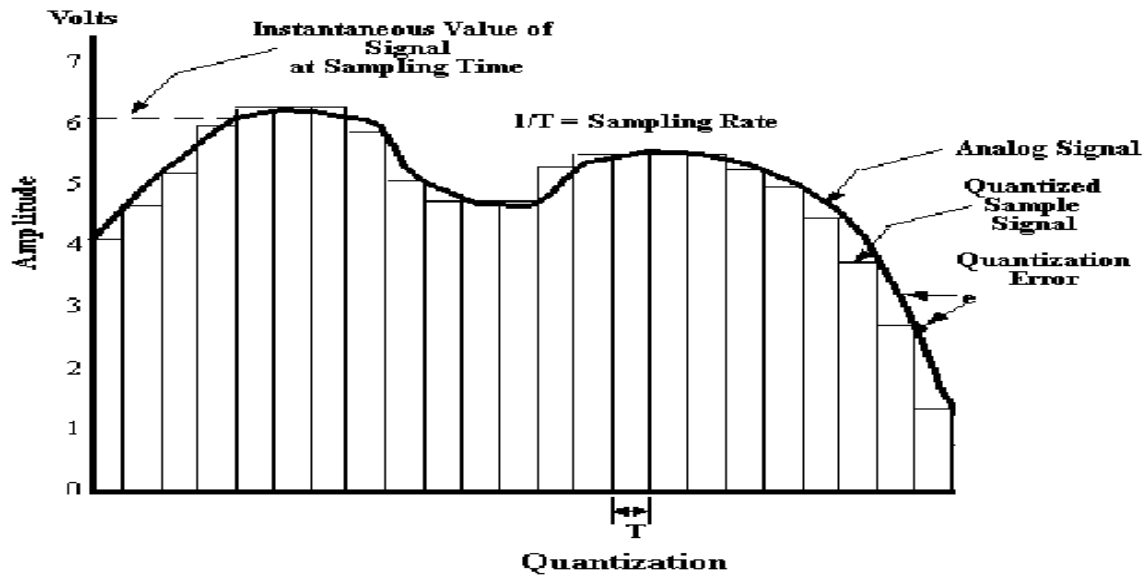


Figure (3.2) Quantization Process.

The fundamental trade off in this choice is the resulting signal quality versus the amount of data needed to represent each sample. ADC converter it has various method. SDM one form of A/D converter by over sampling the signal using high sampling frequency it is possible to trade off signal to noise ratio (**SNR**). for bandwidth, thus decreasing number of quantizing level needed to represent a signal. **SDM** technique digitizes the amplitude of the signal to provide similar representation, of **PCM** except with very high resolution at a cost of much higher sampling rate[14] .

It should be obtained that in conversion of analog signal to digital signal there will losses occur in the information. since it is impossible to exactly reconstruct the original signal from quantizing version . The quantization error accurse can be reduced in the case of **PCM** it is achieved by increasing the number of level, creating a high number of level. In **SDM** the resolution can be increased the sampling rate or increasing the number of quantization levels .[17]

3.4 Over Sampling and Noise Shaping.

oversampling is the process of sampling a signal with a sampling frequency, significantly higher than the Nyquist rate. Theoretically a bandwidth limited signal can be perfectly reconstructed if sampled at or above the nyquist rate, which is twice the highest frequency in the signal. Oversampling improves resolution, reduces noise , helps avoid aliasing and phase distortion, by relaxing anti-aliasing filter performance requirements. And noise shaping Its purpose is to increase the apparent signal to noise ratio of the resultant signal. It does this by altering the spectral shape of the error that is introduced by quantizer[13].

Noise shaping works by putting the quantization error in a feedback loop. Any feedback loop functions as a filter, so by creating a feedback loop for the error itself, the error can be filtered as desired. **SDM** use oversampling and noise shaping the input signal to achieve their performance.

A common parameter used for **SDM** converters is the over sampling ratio (**OSR**) , defined as:

$$\mathbf{OSR} = \frac{f_s}{f_n} = \frac{f_s}{2f_m} \quad (3.1)$$

Where:

OSR: Over sampling ratio.

f_s : Sampling frequency.

f_n : Nyquist frequency

f_m : Signal bandwidth.

The Nyquist frequency , defined as a frequency twice the highest frequency component of the input signal.

The other property common to **SDM** converters is noise shaping, accomplished with the **SDM**. Figure (3.3) shows simplified model of continuous time **SDM**.

The out put $Y(s)$ of modulator is determined by :

$$Y(s) = \frac{H(s)}{1+H(s)} X(s) + \frac{1}{1+H(s)} E(s) \quad (3.2)$$

Where

$X(s)$: Analog input signal.

$Y(s)$: Output signal.

$H(s)$: The loop filter transfer function.

$E(s)$: Quantization noise.

The first term of the right hand side of equation (3.2) is the signal transfer function (**STF**). The second term is noise transfer function (**NTF**). If $H(s)$ has low pass filter characteristic with high DC gain ,then for low frequencies the signal transfer function is closed to 1, while the quantization error tends toward zero (NTF is 0). For frequencies close to half the sampling frequency ,the input signal is filtered and quantization error becomes large. [10,13]

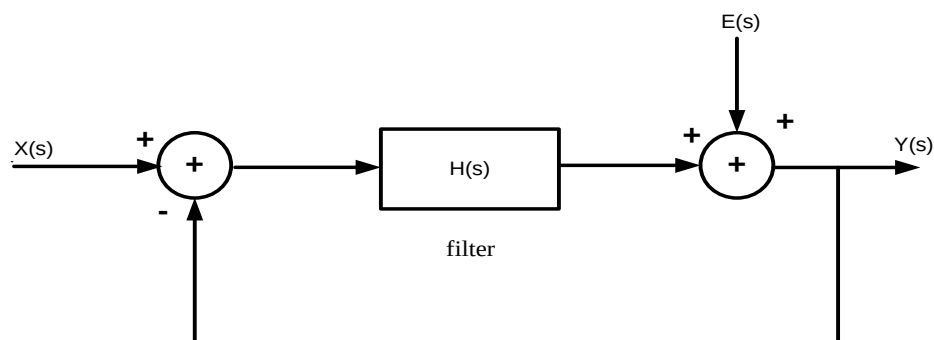


Figure (3.3) SimplifiedContinuousTime SDM Model

3.5 Sigma Delta Modulation Theory

Delta modulation (DM) has been used for several decades to convert analog signals into digital pulse streams for improved immunity noise during transmission. Unlike the PCM procedure where sampling is performed every 2^N clock cycles, the input waveform is Delta Modulator is a differentiator. Each pulse is a quantized difference between the input signal and the accumulated output. In order to recover the signal, the pulse stream must be filtered to remove the clock frequency and integrated to counteract the differentiation effect of the modulator. By extension, problems that affect DC signals may affect other signal frequencies. When the frequency is too slow for a delta modulator, the output suffers from granularity problems. When the output changes too fast, the delta modulator can not track it either, and the modulator output suffers from a slope overload. For these reasons delta modulator is useful only for band limited high frequency AC signals, which is inadequate for most ADC systems. SDM and DM swap the position of the integrator and quantizer. The net effect is a simpler implementation that has the added benefit of shaping the quantization noise away from signals of interest. This effect becomes more dramatic with increased oversampling, which allows for quantization noise to be somewhat programmable. [13,14,15].

3.6 Slope Over Load in Sigma Delta Modulation.

If the input signal is rising or falling with a slope larger than $\frac{\Delta}{T_s}$, where T_s is the sampling time. It observes that the sampler is suffering from **Slope Overload**. the SDM system may be overloaded, just as the DM system if the slope of the signal emerging from the input integrator is too large. The slope of the analog signal is greater than the sigma delta

modulator can maintain. This is called slope overload. If the modulating signal is $m(t)$ then the reconstructed modulating signal $m_q(t)$ can never equal the original modulating signal $m(t)$ because of the one-bit quantization involved.

$$e_q(t) = m(t) - m_q(t) \quad (3.3)$$

However, since the receiver may be thought of as recovering $m(t)$ with an error $e_g(t)$. Where $e_q(t)$ was called the quantization error. In SDM there are two types of non noise error. One, called granular error $e_g(t)$, corresponds to the error due to quantization in the one-bit coding process. The second is called slope overload error $e_{so}(t)$. The composite error due to both granular and slope overload errors is called quantization error in SDM[19].

$$e_q(t) = e_{so}(t) + e_g(t) \quad (3.4)$$

Slope overload errors occur when the input message at the sigma delta modulator has a slope exceeding the slope capability of the feedback signal $m_q(t)$. However, the maximum generated slope capability is $\frac{\Delta}{T_s}$.

Thus, to prevent slope overload, the following condition is required

$$\frac{\Delta}{T_s} \geq \left| \frac{dm(t)}{dt} \right|_{\max}$$

(3.10) Where Δ is the quantizer step size and T_s is sampling period .

Slope overload may be corrected for a given message, by increasing the step-size Δ , or by decreasing T_s .

3.7 Sigma Delta Modulation Parameters.

The most important parameters of sigma delta modulation **(SDM)** Are signal to noise ratio, dynamic range **(DR)** and effective resolution**(ER)**.

These parameters will be described briefly in the following points:

(i) Signal To Noise Ratio(SNR).

In general ,**SNR** can be defined as the ratio of the signal average power to the noise average power in **SDM** system.**SNR** depends on the input signal power and the power of the quantization error. The **SNR** of **SDM** system in case of sinusoidal signal can be written as : [20]

$$P(x)=A\sin^2(2\pi f t) \quad (3.5)$$

$$P(e) = \frac{\Delta^2}{12} \quad (3.6)$$

$$P(N)= \int_{-FB}^{FB} S_e(f)df \quad (3.7)$$

$$S/N \text{ (dB)} = 10 \log \frac{P(x)}{P(N)} \quad (3.8)$$

By substituting (3.5),(3.7) in equation (3.8) get it:

$$S/N \text{ (dB)} \cong 20 \log A + 6.02n + 9.03L - 3.411.$$

Where:

$P(x)$: The power of input sine wave.

$P(e)$: Quantization noise power.

$P(N)$: Noise power.

$S(e)$: noise power spectral density.

Δ : Quantization step size.

A : peak amplitude of the input signal.

L : The order of integrator

(ii) Dynamic range(DR).

Dynamic range is the ratio between maximum signal power and minimum detectable signal power with in specified bandwidth.[9,10]

So the dynamic range (DR) mathematically can be written as:

$$DR = \frac{3}{2} OSR^{2L+1} \frac{2L+1}{\pi^{2L}} \quad (3.9)$$

Where

Δ :Quantization level Step Size.

L : The order of integrator

(iii) Effective Resolution (ER).

The dynamic range of an A/D converter is given by

$$DR = 3 * 2^{(2ER-1)} \quad (3.10)$$

Where :

DR : Dynamic range.

ER : Effective Resolution.

Manipulating this expression yield the effective number of bits or effective resolution of the sigma delta modulator as a function of its dynamic range in **(dB)** .[10]

$$ER(bit) = \frac{DR(dB) - 1.76}{6.02} \quad (3.11)$$

3.8 Transfer Function of the SDM System.

The transfer function of a linear system is the ratio of the Laplace transform of the output to the Laplace Transform of the input. considering its initial conditions and equilibrium point to be zero. If we

have an input function of $X(s)$, and an output function $Y(s)$. therefore the transfer function $H(s)$ is given by:

$$H(s) = \frac{Y(s)}{X(s)} \quad (3.12)$$

It is often helpful to represent the transfer function by means of a technique known as block diagram. Closely associated with the use of block diagrams to represent the input- output characteristic of a plant a set of procedures commonly referred to as block diagram algebra as used. In essence, these manipulations are nothing more than a graphical procedure for manipulating algebra equations. These procedures are often helpful in finding the transfer function of a complex plant by combining the block diagrams of the various parts of the plant in order to find the overall input-output block diagram and hence the transfer function[20,21].

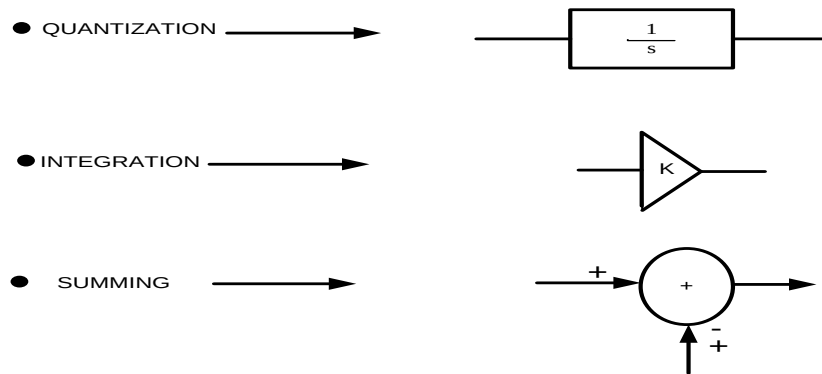
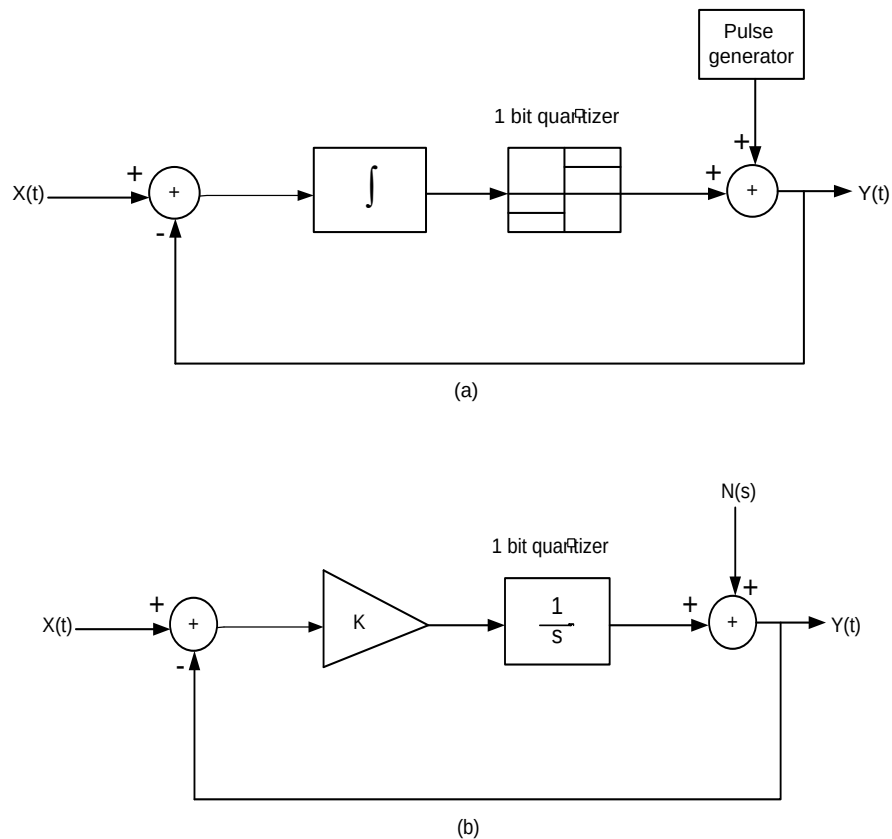


Figure (3.5) The transfer function of mathematical operations

3.8.1 Signal Transfer Function of(SDM).

Firstly will define the transfer function of the signal and noise. the signal transfer function (STF) is the transfer function of the output $Y(s)$ with respect to the plant input $X(s)$. While, the noise transfer function (NTF)

is the transfer function of the output $Y(s)$ with respect to the noise input $N(s)$. The low order low pass sigma delta modulator shown in figure(3-6a), can be drawn as block diagram by using Laplace transform of each block, as shown in figure (3-6b).



Figure(3.6) First Order LPSDM

(a) Block Diagram of First Order LPSDM Modulator.

(b) Block Diagram of First LPSD Modulator Using Laplace Transfer Function.

To find either STF or NTF, it must be assumed that the other input equal zero, so that when STF is desired, the noise input is assumed to be zero as shown in figure (3-7).

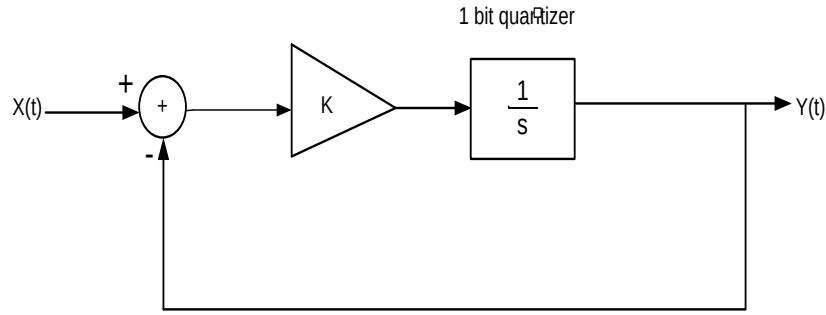


Figure (3.7) The STF of the First Order LPSDM

$$\text{STF} \quad \frac{Y(s)}{X(s)} = \frac{\frac{k}{s}}{1 + \frac{K}{s}} = \frac{k}{s + k} \quad (3.13)$$

Equation (3-12) is . So that the first order LPSDM low passes the input signal within the band of interest.

3.8.2 Noise Transfer Function of SDM.

And when NTF is required the signal input is assumed zero, as shown in figure (3-8).

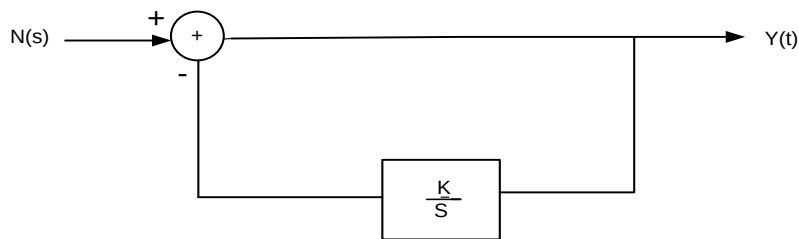


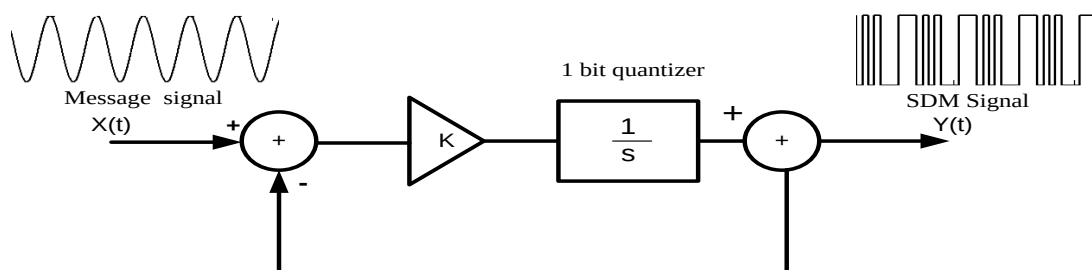
Figure (3.8) The NTF of First Order LPSDM

$$\text{NTF} \quad \frac{Y(s)}{N(s)} = \frac{1}{1 + \frac{k}{s}} = \frac{s}{s + k} \quad (3.14)$$

Equation (3-13) shows that the SDM high passes the noise out the band of interest, so that the noise energy is transferred to higher frequencies and small a mount of noise remains in the baseband region.

3.9 Generation Of Low Pass Sigma Delta Modulation.

When the input signal is applied at the input of SDM modulator it samples the input signal at many times the Nyquist rate (process referred to as oversampling) and performs very coarse analog-to-digital conversion at the resulting narrow-band sequence pulses. Through the use of feedback, the quantization error introduced by the quantizer is spectrally shaped (process referred to as quantization noise shaping) so that its power resides primarily outside the signal band. The digital decimation filter removes the out-of-band portion of the quantization error and reduces the output rate to the Nyquist rate of the input signal. Oversampling, in conjunction with quantization noise shaping. The main advantage of SDM modulator is the noise shaping property. The noise shaping does not reduce the magnitude of the noise, but alters the spectral density of the quantization error by shifting its energy towards higher frequencies. The low order SDM consists of a difference node, an integrator and one bit quantizer. The output of the modulator is terminated by feedback and subtracted from the input of the modulator. The result is integrated and subsequently quantized. the out put of DSM is shown below in figure (3.9).[19,20]



Figure(3.9) First Order LPSDM Model

3.10 Sigma Delta Modulation Detection.

The signal can be detected simply from SDM modulated signal just by using only low Pass filter which remove the high frequency quantization noise without affecting the input signal characteristics and obtain a very good approximation input signal from the modulated signal. Figure (3-10) shows the block diagram of the SDM demodulator. [22]



Figure (3.10) SDM Demodulator

HIGH PASS SIGMA DELTA MODULATION

4.1 Introduction.

Sigma delta modulator ADCs is widely used in the measurement, telecommunications, digital signal processing based on oversampling and noise shaping. The transformation of a low-pass to high-pass modulator is a low-pass to high-pass transformation for the integrator work. that is the integrator acts as differentiator to block the low frequency instead high frequency. High pass sigma delta modulator(HPSDM) is well suited for the digitization of narrow band signals modulated and perform a high resolution analog to digital conversion at high frequency. [15,16,20]

4.2 Development of LPSDM to HPSPDM.

HPSD modulator operate in much the same manner as conventional LPSD modulator and their advantages of Neyquest-rate converter ,as well as its performance modeling. The transformation of a low-pass to high-pass modulator is a low-pass to high-pass transformation for the integrator work. that is the integrator acts as differentiator to block the low frequency instead high frequency. Then look at the technique of oversampling, which can be used to improve the resolution of classical A/D methods..[14]

The quantization noise can be pushed to low frequency region or suppressed in the high frequency region around one-half of the sampling frequency, $\frac{f_s}{2}$ depending on the type of modulator if it is low pass or high pass.

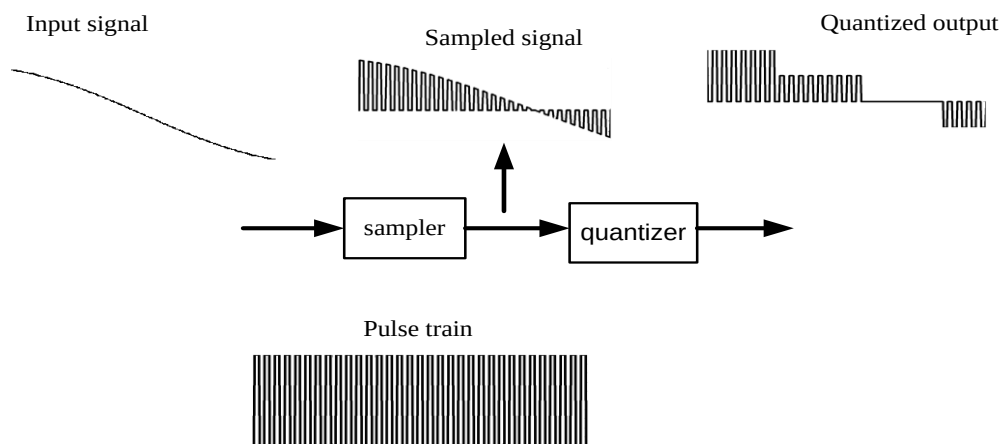
Theory of HPSPDM modulator depending on two principles things such as:

- 1- Quantization.
- 2- Over sampling and noise shaping.

These principles are described in the following subsections.

4.2.1 Quantization.

The sequence numbers which can theoretically still take on any value on a continuous range of values. Quantization is a process to represent each number from such a continuous range to number of quantization levels. Some of error introduced by quantization is referred to as quantization error, the difference between the actual analog value and quantized digital value is called quantization error or quantization distortion. The error signal is sometimes modeled as an additional random signal called quantization noise. If assumes that the quantization step size of the **ADC** is Δ and the quantization error is evenly distributed in the range off $\pm \frac{\Delta}{2}$, we obtain the quantization noise power by using the following formula..[17]



Figure(4.1) Represent Quantization Process

The Quantization noise power is defined by the following equation:

$$e^2_{\text{rms}} = \frac{1}{\Delta} \int_{-\frac{\Delta}{2}}^{\frac{\Delta}{2}} e^2 \, de = \frac{\Delta^2}{12} \quad (4.1)$$

where:

Δ : Quantization step size.

e :Quantization error.

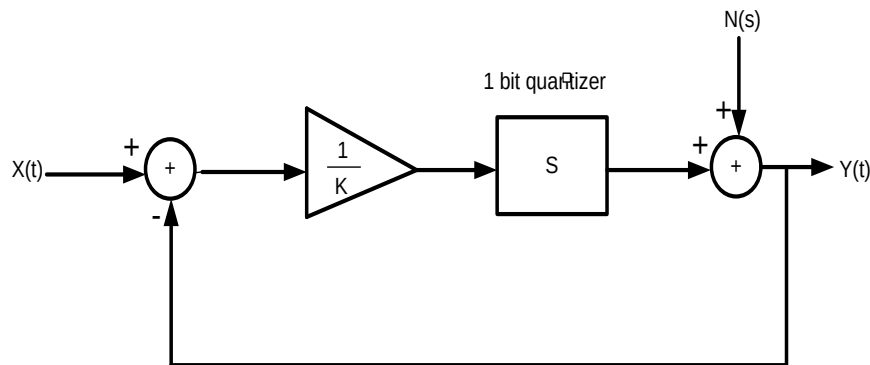
4.2.2 Over Sampling and Noise Shaping.

The operation of delta-sigma modulation is also based on oversampling, noise shaping. High over sampling rate are necessary. For HPSSD modulator, however a lower sampling rate can be used as long as it is much greater than bandwidth of interest in input signal.[18]

Oversampling improves resolution, reduces noise and helps avoid aliasing and phase distortion by relaxing anti-aliasing filter performance requirements. The efficiency of the over sampling can be improved by the sigma delta modulator, in which the feedback signal is used to shape the quantization noise spectrum. The quantization noise can be shaped to low frequency region or suppressed to the high frequency region around one-half of the sampling frequency. The Noise shaping works by putting the quantization error in a feedback loop. Any feedback loop functions as a filter, so by creating a feedback loop for the error itself, the error can be filtered as desired noise-shaping is to reshape the spectrum of quantization noise so that most of the noise is filtered out of the relevant frequency band. [17,20]

4.3 Transfer Function of HPSDM.

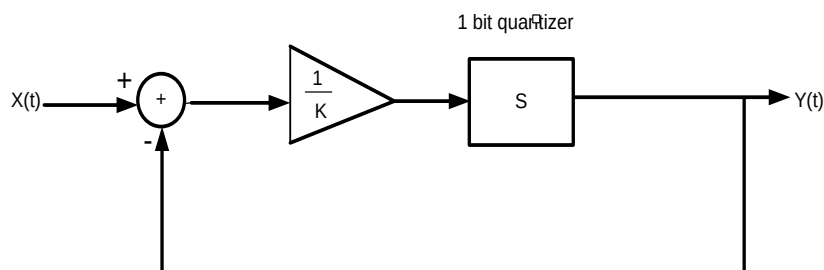
Figure(4.2) shows the general model of first order **HPSDM**. which consist of subtract , filter and quantizer . To achieve signal transfer function and noise transfer function of first order **HPSDM** the quantizer is modeled by addition white noise $N(s)$.



Figure(4.2) Modeling of First Order HPSDM

4.3.1 Signal Transfer Function of(HP SDM).

The signal transfer function of the high-pass **DS** modulator, has a pass band around one-half of the sampling frequency so that the input signal in the frequency band around $\frac{f_s}{2}$ can go through the modulator. To find STF, it must be assumed that the other input equal zero. So that when STF is desired, the noise input is assumed to be zero as shown in figure (4.3).



Figure(4.3) Modeling of First Order HPSDM to obtain STF

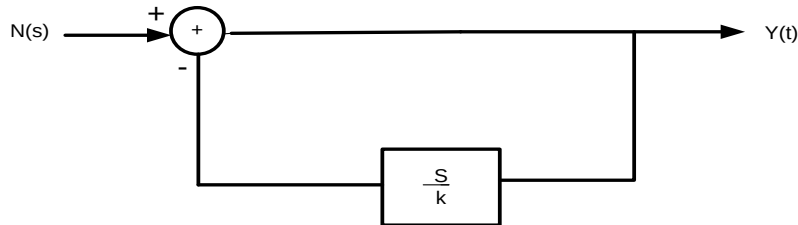
Then STF is given by

$$\text{STF} = \left. \frac{Y(s)}{X(s)} \right|_{N(s)=0} = \frac{\frac{s}{k}}{1 + \frac{s}{k}} = \frac{s}{k + s} \quad (4.2)$$

4.3.2 Noise Transfer Function of HPSDM.

The quantization noise transfer function of the high pass **DS** modulator have a very high attenuation around $f_s/2$ so that a large part of the quantization noise can be distributed out of the signal band around one-half of the sampling frequency. NTF model is shown in figure (4.4). when NTF is required the signal input is assumed zero[23]. Also NTF which is defined by :

$$\text{NTF} = \left. \frac{Y(s)}{N(s)} \right|_{X(s)=0} = \frac{k}{k + s} \quad (4.3)$$



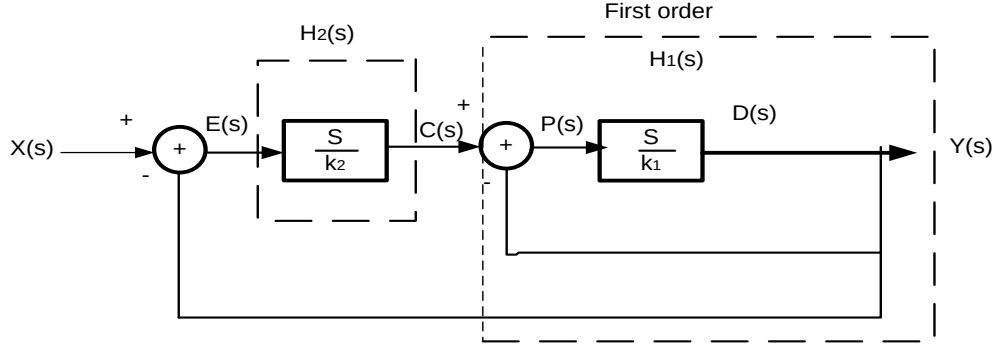
Figure(4.4) Modeling of First Order HPSDM to Obtain NTF

4.3.3 Signal Transfer Function of Second Order.

Signal transfer function of second order HPSDM can be obtained as following:

$$H_1(s) = \frac{s}{k_1} \quad (4.4)$$

$$H_2(s) = \frac{s}{s + k_2} \quad (4.5)$$



Figure(4.5) Modeling of Second Order HPSPDM (STF)

$$G(s) = H_1(s) * H_2(s) \quad (4.6)$$

Substituting equation (4.4) and (4.5) into equation (4.6), it gets

$$G(s) = \frac{s^2}{k_2 s + k_1 k_2} \quad (4.7)$$

$$H(s) = \frac{Y(s)}{X(s)} = \frac{G(s)}{1 + G(s)} \quad (4.8)$$

So, the STF of second order HPSPDM is given by this

$$\text{STF} = \frac{Y(s)}{X(s)} = \frac{\frac{s^2}{k_1 k_2 + k_2 s}}{1 + \frac{s^2}{k_1 k_2 + k_2 s}} = \frac{s^2}{s^2 + k_2 s + k_1 k_2} \quad (4.9)$$

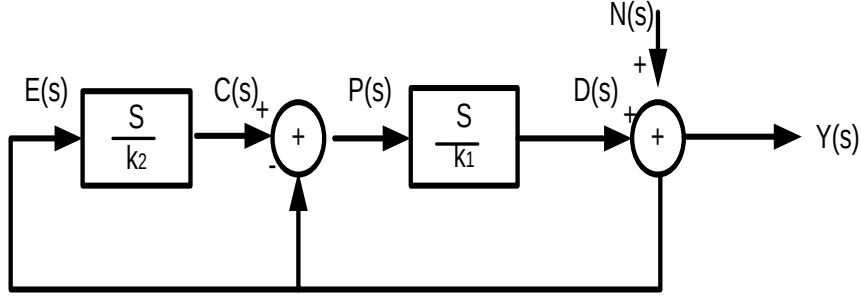
4.3.4 Noise Transfer Function of Second Order HPSPDM.

Also, the NTF of second order HPSPDM can be obtained by using the following steps.

$$\text{NTF} = \left. \frac{Y(s)}{X(s)} \right|_{X(s)=0} \quad (4.10)$$

The output of the subtracted is given by:

$$E(s) = -Y(s) \quad (4.11)$$



Figure(4.6) Modeling of Second Order HPsDM (NTF)

So, the output of the integrator $C(s)$ is given by:

$$C(s) = -E(s) \frac{s}{k_2} \quad (4.12)$$

$$P(s) = C(s) - Y(s) \quad (4.13)$$

Substituting equation (4.12) and (4.12) into (4.13), it gets

$$P(s) = \left[-\frac{s}{k_2} \right] Y(s) - Y(s) \quad (4.14)$$

$$\therefore P(s) = -Y(s) \left[\frac{s}{k_2} + 1 \right] \quad (4.15)$$

So the output of an integrator $D(s)$ is given by:

$$D(s) = P(s) \frac{s}{k_1} \quad (4.16)$$

Substituting equation (4.15) into equation (4.16), it gets

$$D(s) = -Y(s) \left[\frac{s}{k_1} + \frac{s^2}{k_1 k_2} \right] \quad (4.17)$$

The out put $Y(s)$ is

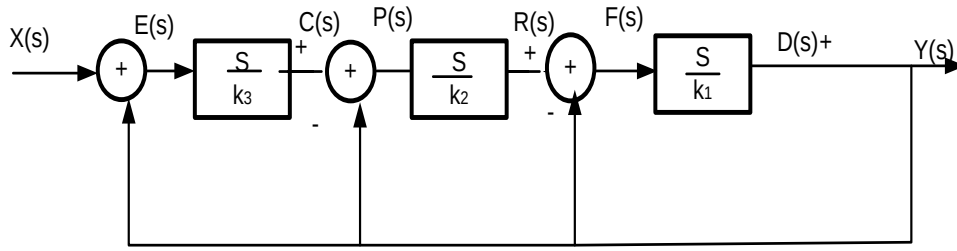
$$Y(s) = D(s) + N(s) \quad (4.18)$$

Substituting equation (4.18) into equation (4.19), we get

$$(4.19) \quad NTF = \frac{Y(s)}{X(s)} = \frac{k_1 k_2}{s^2 + s k_2 + k_1 k_2}$$

4.3.5 Signal Transfer Function of Third Order HPSDM.

Signal transfer function of third order HPSDM can be obtained by referring to equation (4.9) in second order as following:



Figure(4.7) Modeling of Third Order HPSDM(STF)

$$G(s) = H_1 * H_2 \quad \& \quad H_1 = \frac{s}{k_3} \quad \text{And} \quad H_2 = \frac{s^2}{s^2 + k_2 s + k_1 k_2}$$

$$G(s) = \frac{s^3}{k_3 s^2 + k_2 k_3 s + k_1 k_2 k_3} \quad (4.20)$$

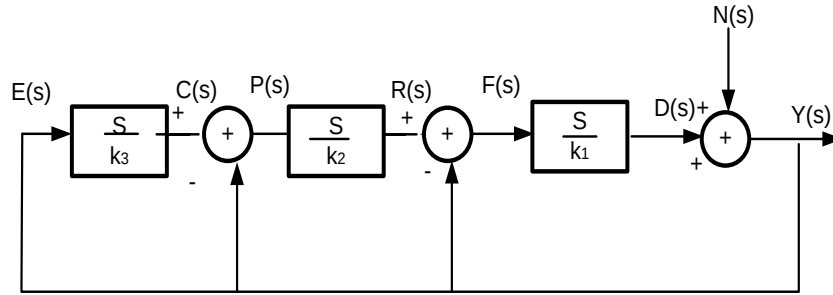
$$H(s) = \frac{Y(s)}{X(s)} \quad (4.21) = \frac{G(s)}{G(s) + 1}$$

Substituting equation (4.21) into (4.22) we get

$$STF = \frac{s^3}{s^3 + k_3 s^2 + k_2 k_3 s + k_1 k_2 k_3} \quad (4.22)$$

4.3.6 Noise Transfer Function of Third Order HPSDM.

The NTF of third order HPSDM can be obtained by using the following steps .



Figure(4.8) Modeling of Third Order HPSDM(NTF)

$$NTF = \frac{Y(S)}{N(S)} \Big|_{X(s)=0}$$

$$E(s) = - Y(s) \quad (4.23)$$

$$C(s) = E(s) \frac{s}{k_3} = -Y(s) \frac{s}{K_3} \quad (4.24)$$

$$P(s) = C(s) - Y(s) \quad (4.25)$$

Substituting equation (4.24) into equation (4.25) ,we get

$$P(s) = -Y(s) \left[\frac{s}{k_3} + 1 \right] \quad (4.26)$$

$$R(s) = P(s) * \frac{s}{k_2} \quad (4.27)$$

Substituting equation (4.26) into equation (4.27) ,we get

$$R(s) = -Y(s) \left[\frac{s^2}{k_2 k_3} + \frac{s}{k_2} \right] \quad (4.28)$$

$$F(s) = R(s) - Y(s) \quad (4.29)$$

Substituting equation (4.28) into equation (4.29) ,we get

$$F(s) = -Y(s) \left(\left[\frac{s^2}{k_3 k_2} + \frac{s}{k_2} \right] + 1 \right) \quad (4.30)$$

$$D(s) = F(s) * \frac{S}{K_1} \quad (4.31)$$

Substituting equation (4.30) into equation (4.31) ,we get

$$D(s) = -Y(s) \left[\frac{s^3}{k_1 k_2 k_3} + \frac{s^2}{k_1 k_2} + \frac{s}{k_1} \right] \quad (4.32)$$

$$Y(s) = D(s) + N(s) \quad (4.33)$$

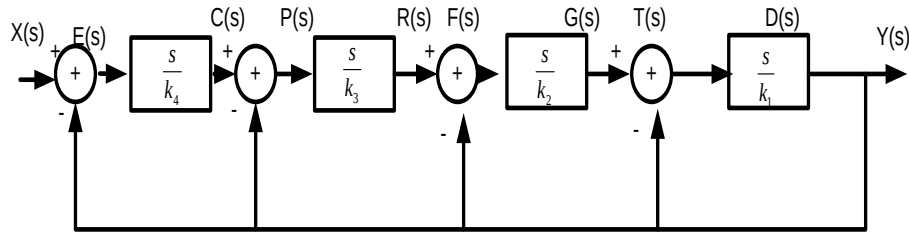
Substituting equation (4.32) into equation (4.33) ,we get

$$Y(s) = -Y(s) \left[\frac{s^3}{k_1 k_2 k_3} + \frac{s^2}{k_2 k_1} + \frac{s}{k_1} \right] + N(s) \quad (4.34)$$

$$NTF = \frac{Y(s)}{N(s)} = \frac{k_1 k_2 k_3}{s^3 + k_3 s^2 + k_2 k_3 s + k_1 k_2 k_3} \quad (4.35)$$

4.3.7 Signal Transfer Function of Fourth Order HPSDM.

Signal transfer function of fourth order HPSDM can be obtained by referring to equation (4.22) in third order as following:



Figure(4.9) Modeling of Fourth Order HPSDM(STF)

$$G(s) = H_1 * H_2 \quad \& \quad H_1 = \frac{s}{k_4} \quad \text{And} \quad H_2 = \frac{s^3}{s^3 + k_3 s^2 + k_2 k_3 s + k_1 k_2 k_3}$$

$$G(s) = \frac{s^4}{s^3 k_4 + s^2 k_3 k_4 + s^2 k_2 k_3 k_4 + k_1 k_2 k_3 k_4} \quad (4.36)$$

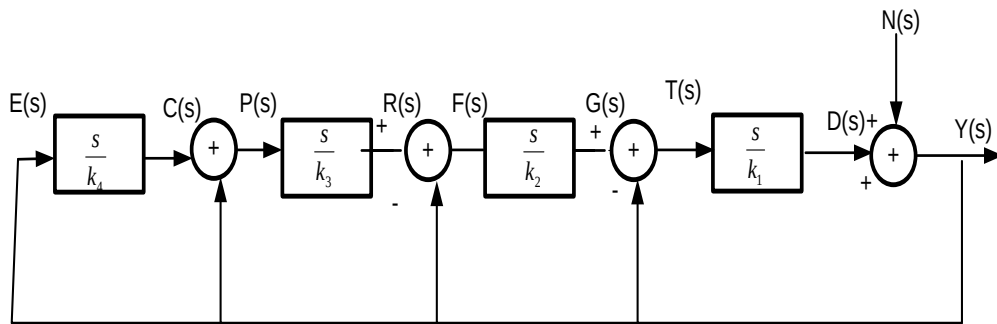
$$H(s) = \frac{Y(s)}{X(s)} \Big|_{N(s)=0} = \frac{G(s)}{1 + G(s)} \quad (4.37)$$

By substitute equation (4.36) into equation (4.37) then we get

$$\text{STF} = \frac{s^4}{s^4 + s^3 k_4 + s^2 k_3 k_4 + s k_2 k_3 k_4 + k_1 k_2 k_3 k_4} \quad (4.38)$$

4.3.8 Noise Transfer Function of Fourth Order HPSDM.

The NTF of fourth order HPSDM can be obtained by using the following steps based on the figure (4.10) instead increasing one integrator called $\frac{s}{k_4}$ as shown in figure (4.7).



Figure(4.10) Modeling of Fourth Order HPSDM(NTF)

$$E(s) = -Y(s) \quad C(s) = E(s) * \frac{s}{k_4}$$

$$C(s) = -Y(s) \frac{s}{k_4} \quad (4.39)$$

$$P(s) = C(s) - Y(s) = -Y(s) \left[\frac{s}{k_4} + 1 \right] \quad (4.40)$$

$$R(s) = P(s) \frac{s}{k_3} = -Y(s) \left[\frac{s}{k_4 k_3} + \frac{s}{k_3} \right] \quad (4.41)$$

$$F(s) = R(s) - Y(s) = -Y(s) \left(\left[\frac{s^2}{k_3 k_4} + \frac{s}{k_3} \right] + 1 \right) \quad (4.42)$$

$$G(s) = F(s) \frac{s}{k_2} = -Y(s) \left[\frac{s^3}{k_2 k_3 k_4} + \frac{s^2}{k_2 k_3} + \frac{s}{k_2} \right] \quad (4.43)$$

$$T(s) = G(s) - Y(s) = -Y(s) \left(\left[\frac{s^3}{k_2 k_3 k_4} + \frac{s^2}{k_2 k_3} + \frac{s}{k_2} \right] + 1 \right) \quad (4.44)$$

$$D(s) = T(s) \frac{s}{k_1} = -Y(s) \left[\frac{s^4}{k_1 k_2 k_3 k_4} + \frac{s^3}{k_1 k_2 k_3} + \frac{s^2}{k_1 k_2} + \frac{s}{k_1} \right] \quad (4.45)$$

$$Y(s) = D(s) + N(s) \quad (4.46)$$

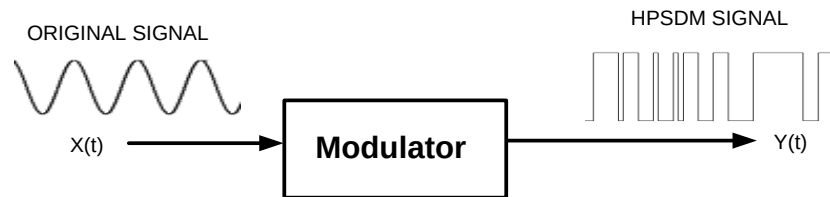
By substitute equation (4.45) into equation (4.46) then gets

$$NTF = \frac{Y(s)}{N(s)} = \frac{k_1 k_2 k_3 k_4}{s^2 + s^3 k_4 + s^2 k_3 k_4 + s k_2 k_3 k_4 + k_1 k_2 k_3 k_4} \quad (4.47)$$

4.7 Generation of First Order of HPSDM.

HPSD modulator shown in figure (4.8) that reduces many of the problems found in the conventional modulators such as in (delta modulation) are represented as quantization noise and slop over load.

When apply an arbitrary signal, to the first order HPSPD modulator. in the previous section the input to integrator is difference between the input signal and quantized out put signal. This difference at integrator input is equal to the quantization error signal, the error signal is passed through an integrator, and then quantized by quantizer to obtain HPSPDM signal in the out put. [19,21]

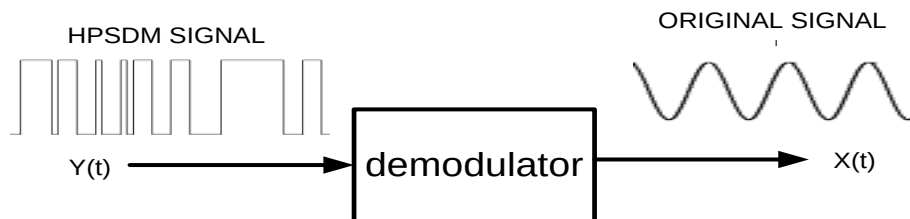


Figure(4.11) First Order HPSPDM Modem

The first order HPSPD modulator consists of a difference node, an HPTF integrator and one bit quantizer. The output of the modulator is terminated by feedback and subtracted from the input of the modulator. The result is integrated and subsequently quantized the out put of DSM[23,20].

4.8 Detection of First Order HPSPDM.

the original arbitrary signal back from the SDM modulated signal is very simple just needs a high pass filter (HPF). Which passed the original signal and reject the unwanted signal and noise as shown in figure (4.9).



Figure(4.12) HPSPD Demodulator.

SIMULATION OF HIGH PASS SIGMA DELTA MODULATION SECOND AND FOURTH ORDER

5.1 Introduction.

simulation, it is very straightforward to represent and then simulate a mathematical model representing a physical system. Models are represented graphically in simulation as block diagrams. A wide array of blocks are available to the user in provided libraries for representing various phenomena and models in a range of formats. Also simulation is a technique used to analyses and observe the behavior of the model representing a physical system in order to verify the behavior of the real system. The main purpose of the simulink is to give the user sense of the fun of modeling and simulation ,the data obtained can be used to write a program in the form of matlab function and commands ,this type of programming written in the file known as M-file where the functions and commands are executed at once. The reason of called it M-file ,due to the extension of the program, which is stored in the directory work, is (file- name .m). The purpose of studying second and fourth order high pass sigma delta modulation systems to achieve different goal without actually construction or representing real process. In this chapter will apply software by matlab to generate 2nd and 4th order HPSDM to achieves the modulated and modulating signals in both time and frequency domains.

5.2. Simulate of Second Order HPSDM

By apply a Matlab program (m. file) (see appendix A) to generate the modulating and modulated signals in both time and frequency domains.

Four testes are carried out is described in the following points:

- **Test I**

The modulating signal is a sinusoidal signal with peak amplitude of 0.45V ,and the modulating frequency is equal to 2kHz applied at the input of the second order high pass sigma delta modulation shown in figure (4.9) . The over sampling ratio is chosen equal to 32 and the center frequency of the modulator is $f_0 = 0.5$, and the maximum gain of the noise transfer function is equal to 1.5, After apply a Matlab software gets the simulation results are shown in figure (5.1).

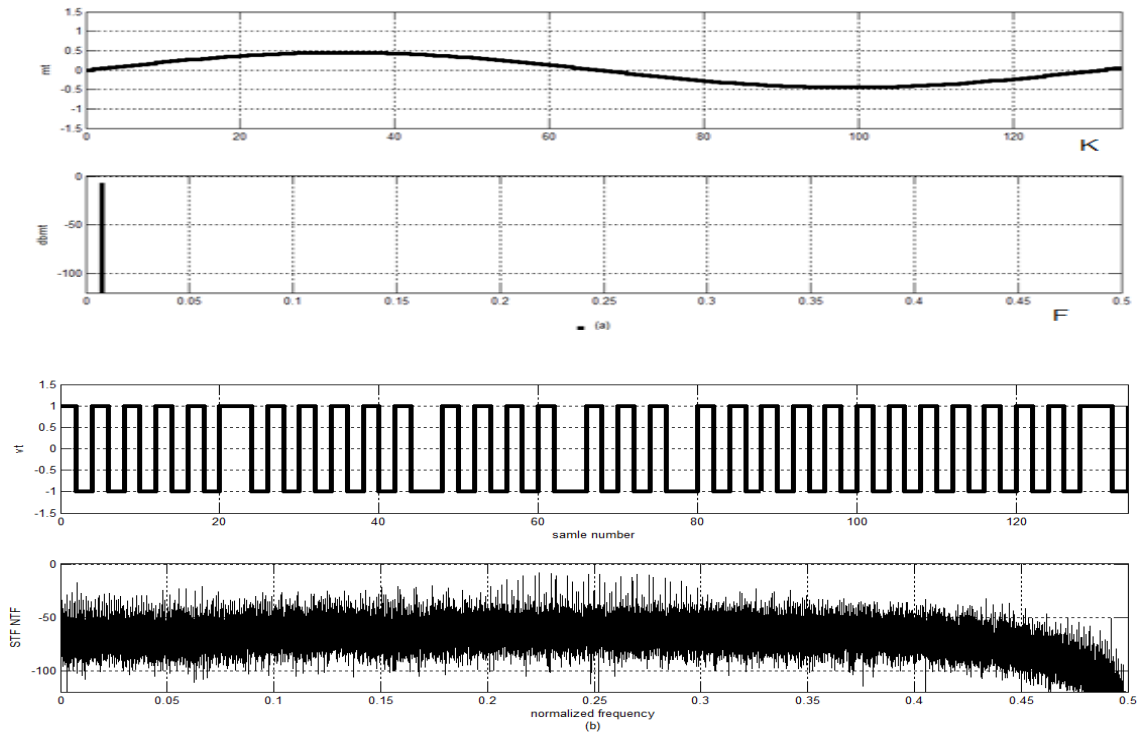


Figure (5.1) Second Order High Pass Sigma Delta Modulator Waveforms In Both Time and Frequency Domain

(a) Modulating Signal

(b) Modulated signal

- **Test II**

Test II same as test I except that the peak amplitude of the signal is doubled i.e. $A_m=0.9V$ and all the other parameters are kept constant also after written a Matlab program achieved simulation results in both time and frequency domains are represented in figure (5.2).

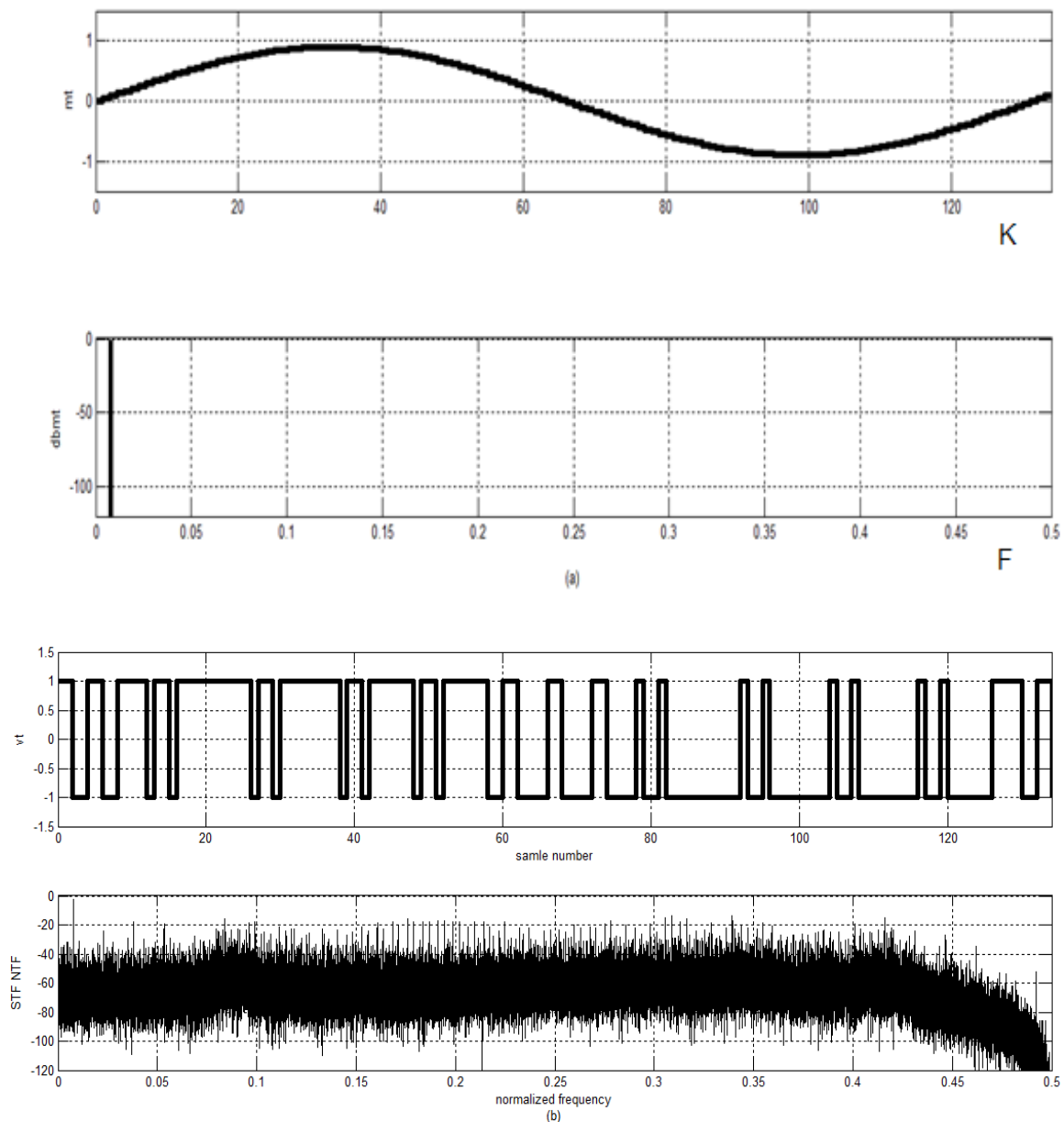


Figure (5.2) Second Order High Pass Sigma Delta Modulator Wave Forms In Both Time and Frequency Domain

(a) Modulating Signal

(b) Modulated signal

- **Test III**

This test same as test I except that is the modulating frequency is doubled i.e. equal 4kHz while all the other parameters of the simulation are kept constant. After written a Matlab program achieved simulation results is represented in figure (5.3).

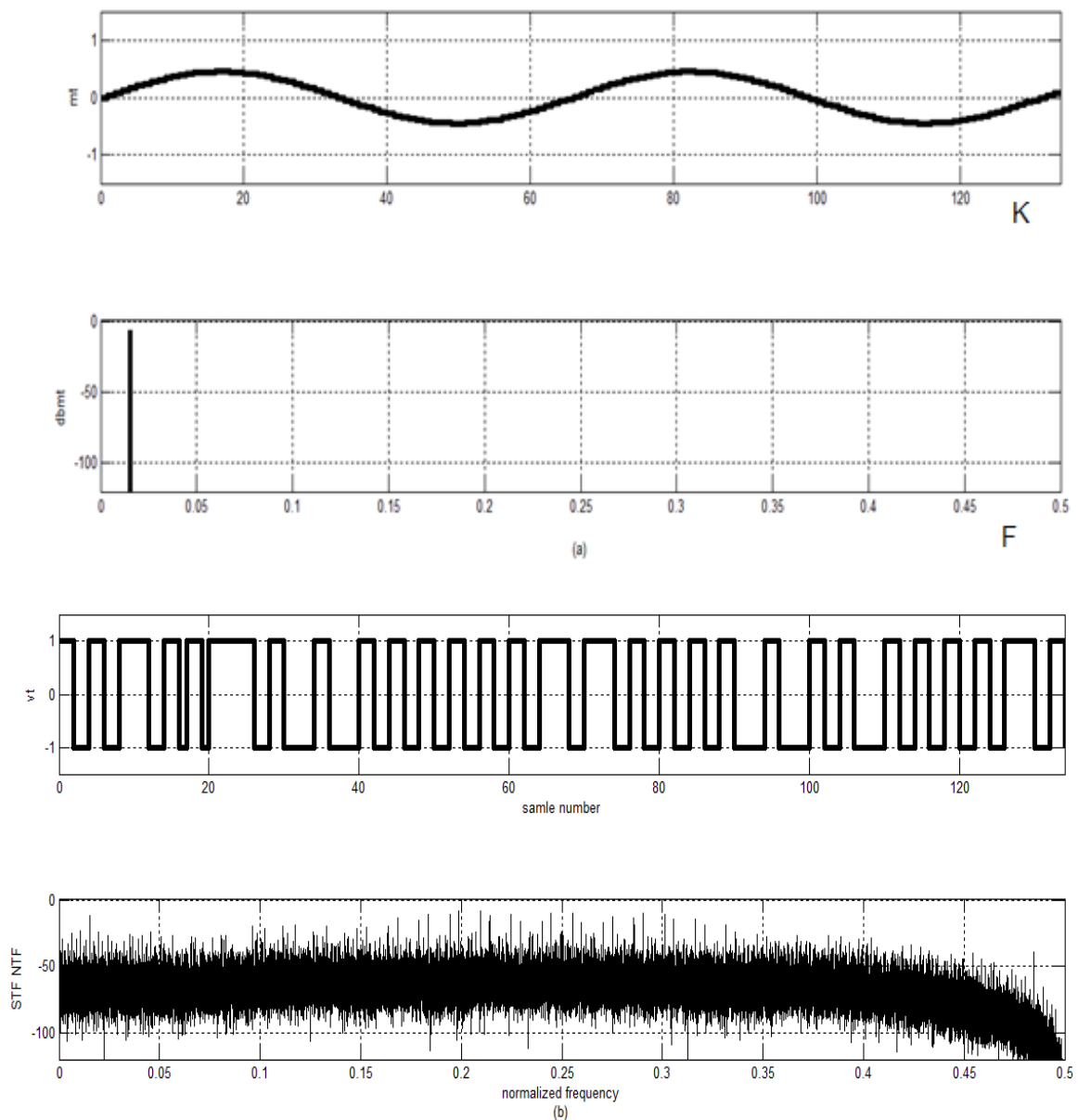


Figure (5.3) Second Order High Pass Sigma Delta Modulator Wave Forms In Both Time and Frequency Domain

(a) Modulating Signal

(b) Modulated signal

- **Test IV**

This test same as test III except that the peak amplitude of modulating signal is doubled i.e. equal 0.9V instead of 0.45V, while all the other parameters of the simulation are kept constant. After writing a Matlab program achieved simulation results is represented in figure (5.4).

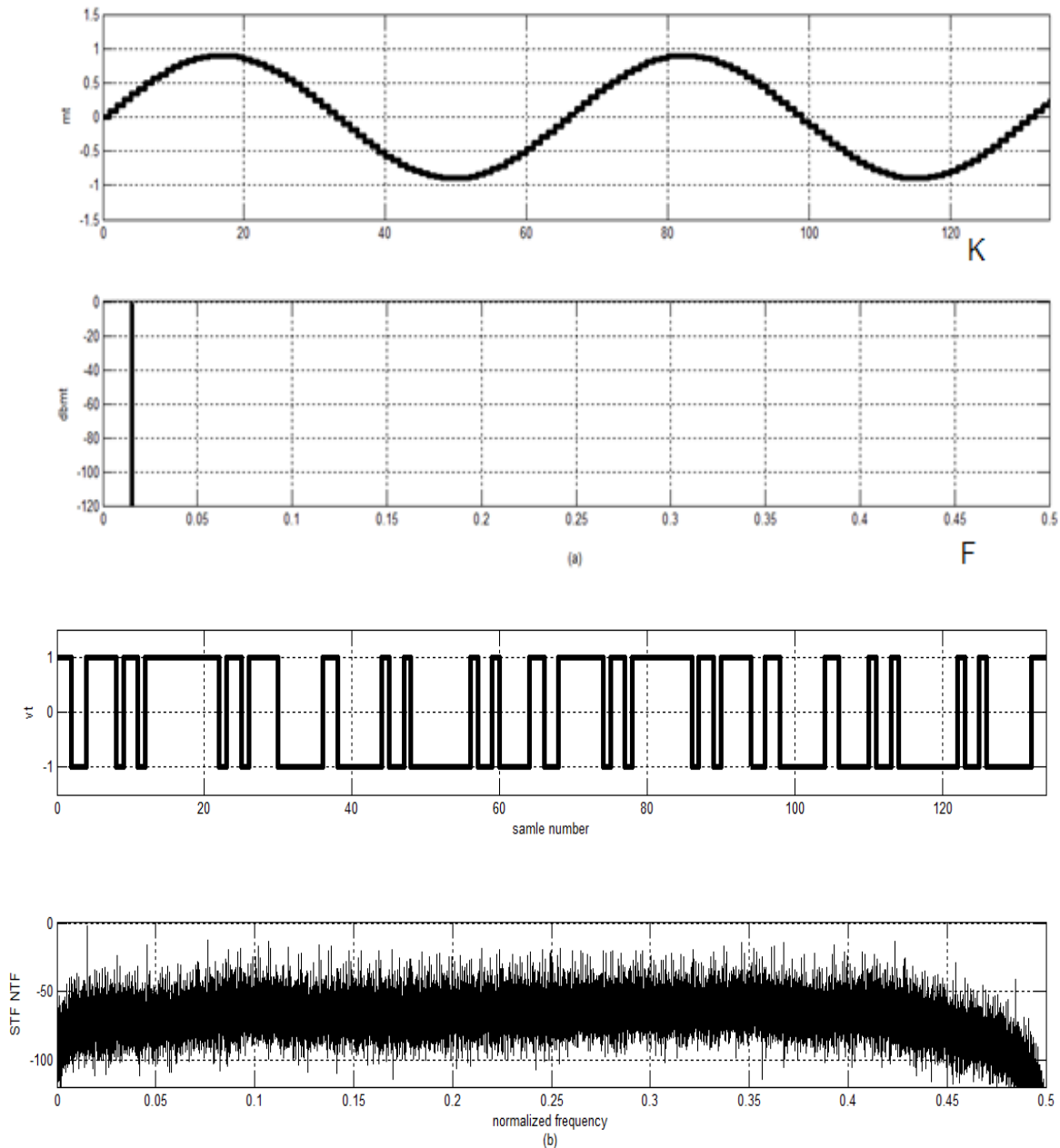


Figure (5.4) Second Order High Pass Sigma Delta Modulator Wave Forms In Both Time and Frequency Domain

(a) Modulating Signal

(b) Modulated signal

5.3 Fourth Order High Pass Sigma Delta Modulation

By written Matlab program (see appendix A) ,for four testes will be carried out in the following points:

- **Test I**

In this test same as in second order testes will do. the modulating signal is a sinusoidal signal with peak amplitude of 0.45V ,and the modulating frequency is equal to 2kHz. as well as the over sampling ratio is equal to 32 and the center frequency of the modulator is $f_0 = 0.5$, and the maximum gain of the noise transfer function is equal to 1.5, After written a matlab program it achieve the simulation results in figure (5.5).

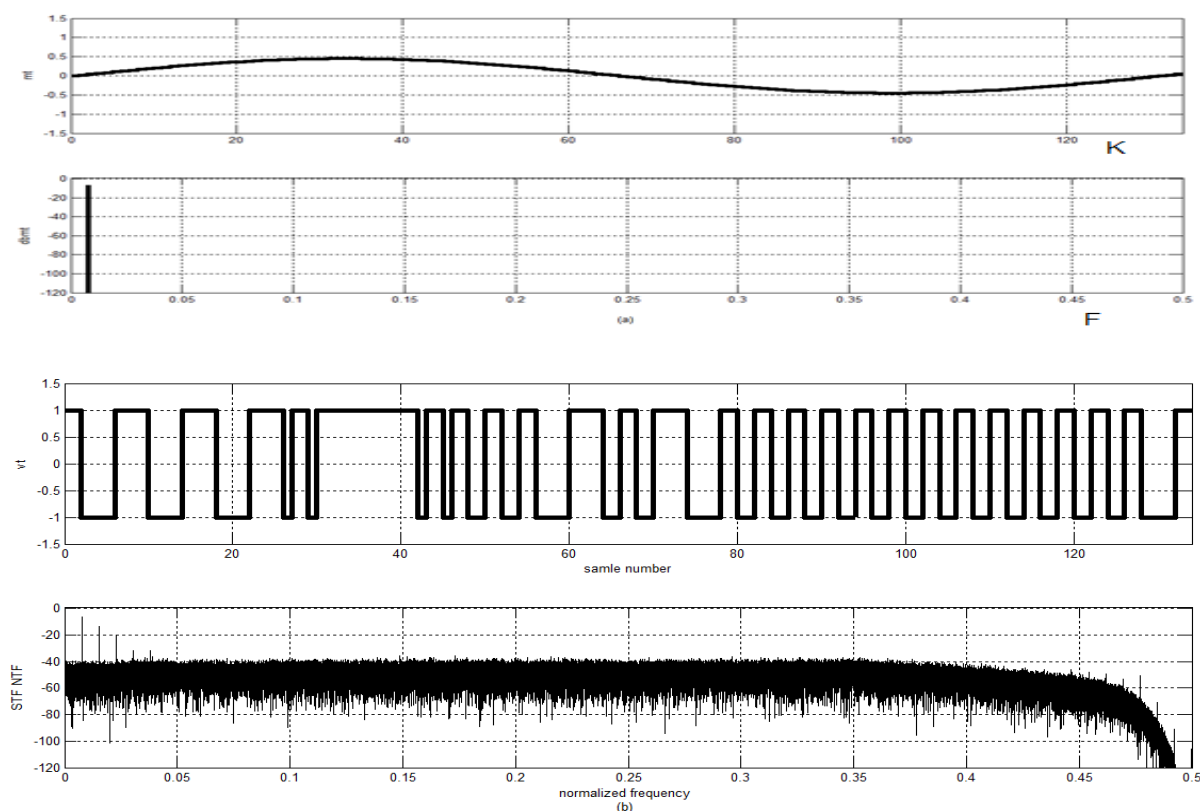


Figure (5.5) Fourth Order High Pass Sigma Delta Modulator Wave Forms In Both Time and Frequency Domain

(a) Modulating Signal

(b) Modulated signal

- **Test II**

Test II same as test I except that the peak amplitude of the signal is doubled i.e. $A_m=0.9V$ and all the other parameters are kept constant also after written a Matlab program achieved simulation results in both time and frequency domains are represented in figure (5.6).

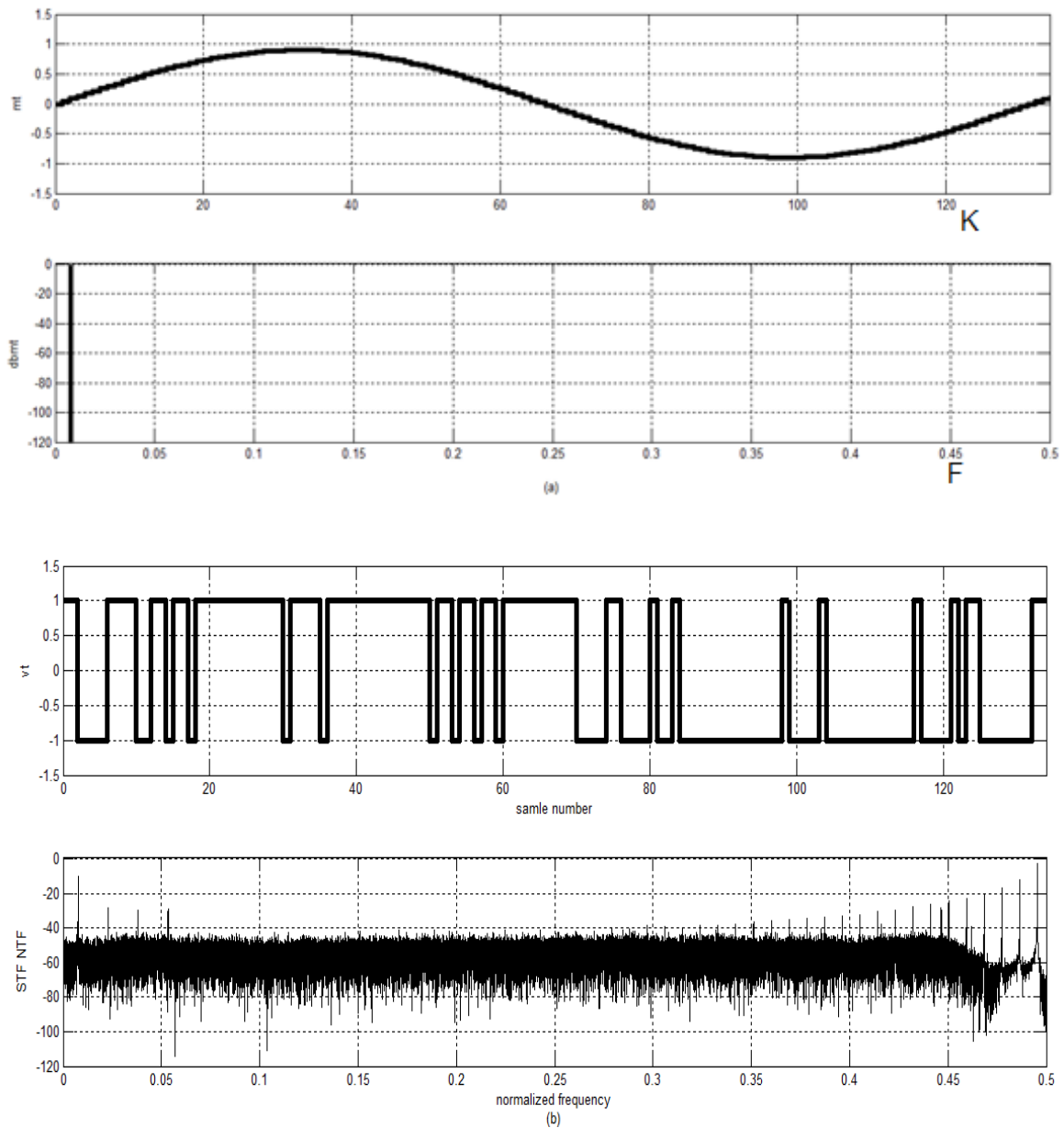


Figure (5.6) Fourth Order High Pass Sigma Delta Modulator Wave Forms In Both Time and Frequency Domain

(a) Modulating Signal

(b) Modulated signal

- **Test III**

This test same as test I except that is the modulating frequency is doubled i.e. equal 4kHz while all the other parameters of the simulation are kept constant. After written a Matlab program achieved simulation results is represented in figure (5.7).

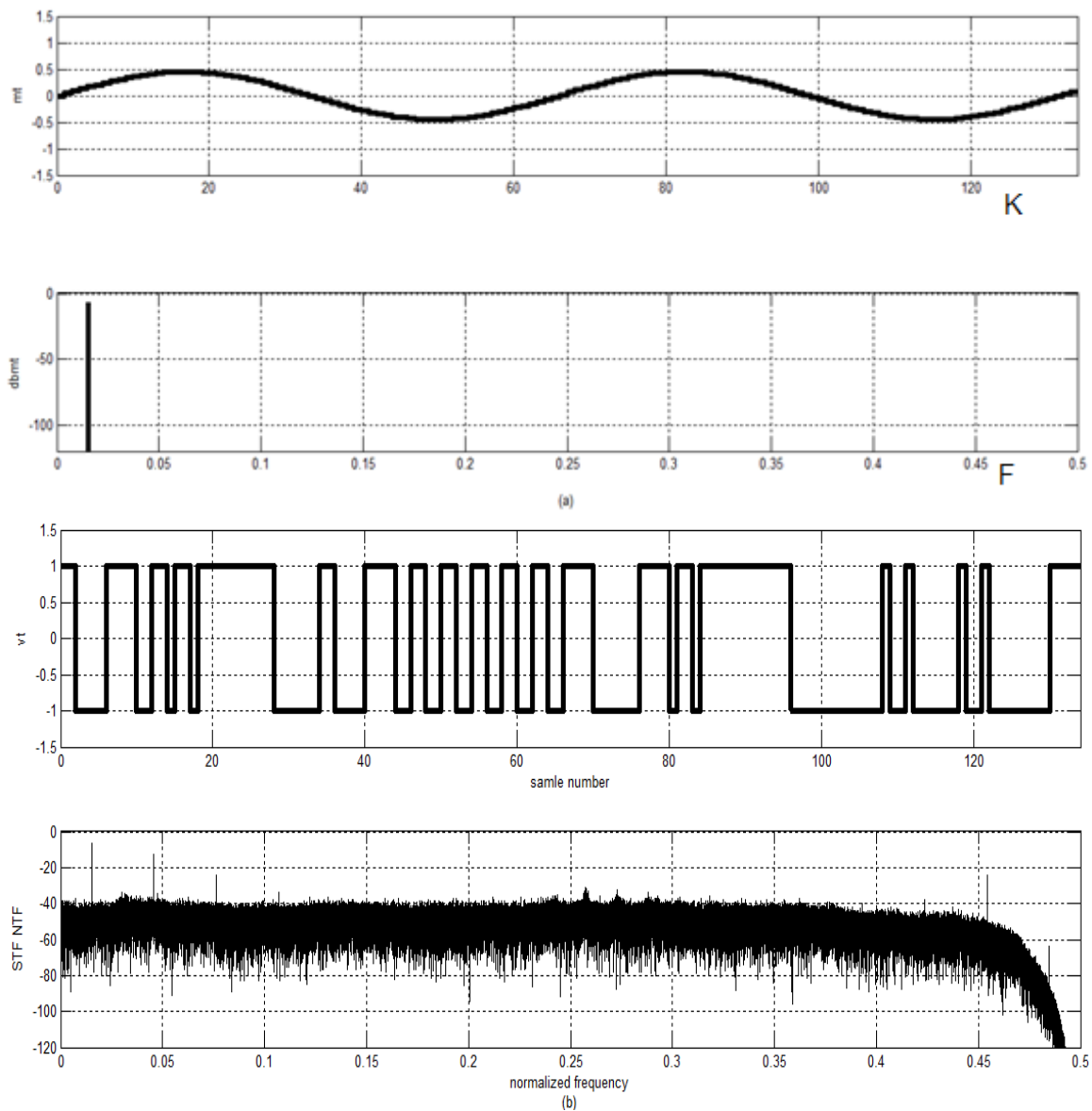


Figure (5.7) Fourth Order High Pass Sigma Delta Modulator Wave Forms In Both Time and Frequency Domain

(a) Modulating Signal

(b) Modulated signal

• Test IV

This test same as test III except that the peak amplitude of modulating signal is doubled i.e. equal 0.9V instead of 0.45V, while all the other parameters of the simulation are kept constant. After written a Matlab program achieved simulation results is represented in figure (5.8).

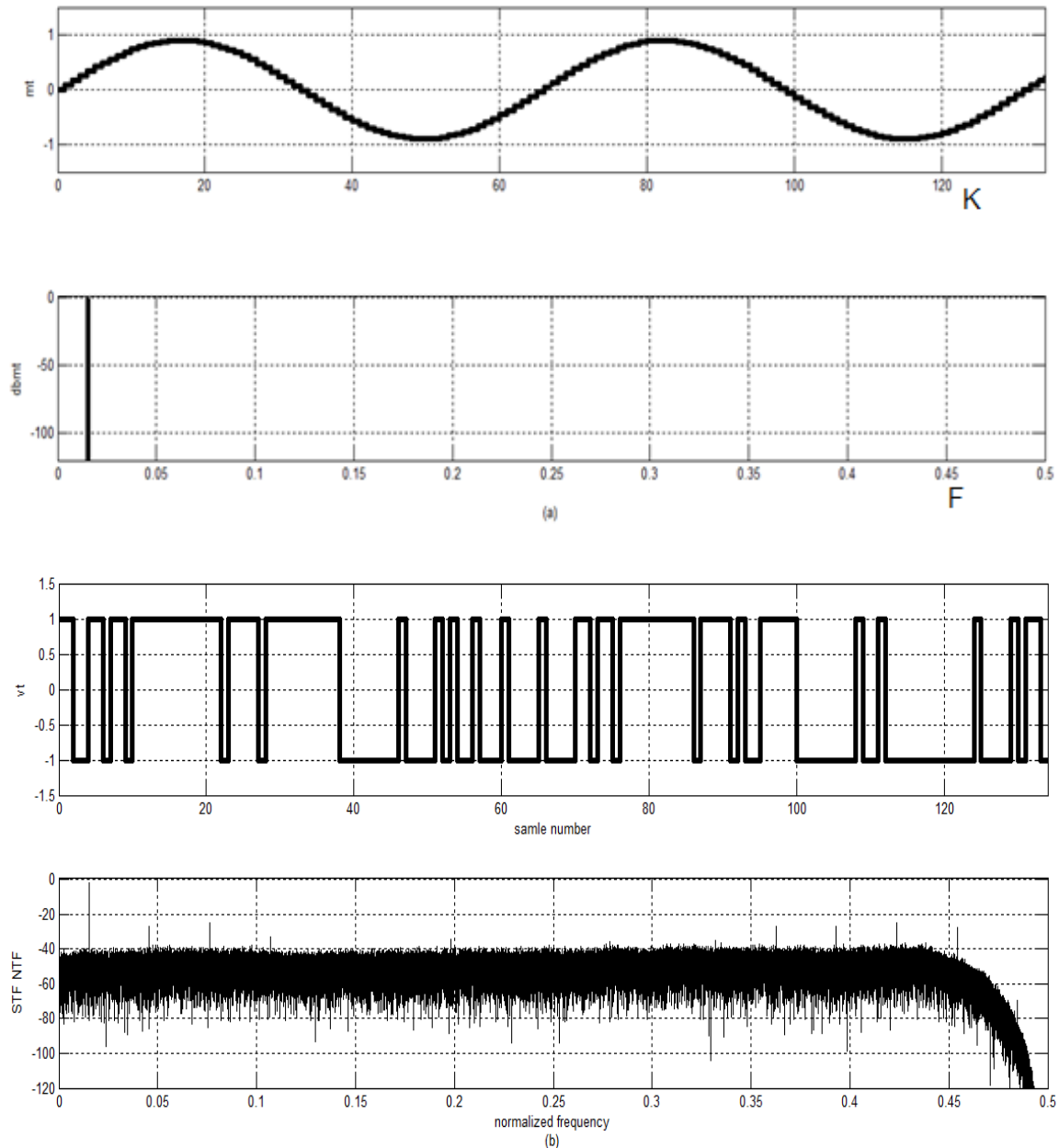


Figure (5.8) Fourth Order High Pass Sigma Delta Modulator Wave Forms In Both Time and Frequency Domain

(a) Modulating Signal

(b) Modulated signal

5.4 Simulation of HPSDM With Additive White Noise.

When a band limited additive noise is added to the modulating signal then after that to study the behavior of the effect of noise on the second and fourth order HPSDM system .

5.4.1 Simulate of Second Order HPSDM With Additive White Noise

In this subsection it generate second order high pass sigma delta modulation with additive white noise by written Matlab program (see appendix B) ,for four testes which will be carried out in the following points:

- **Test I**

The parameters of this simulation as in **test I** of second order and the simulation results are achieved as shown in figure (5.9).

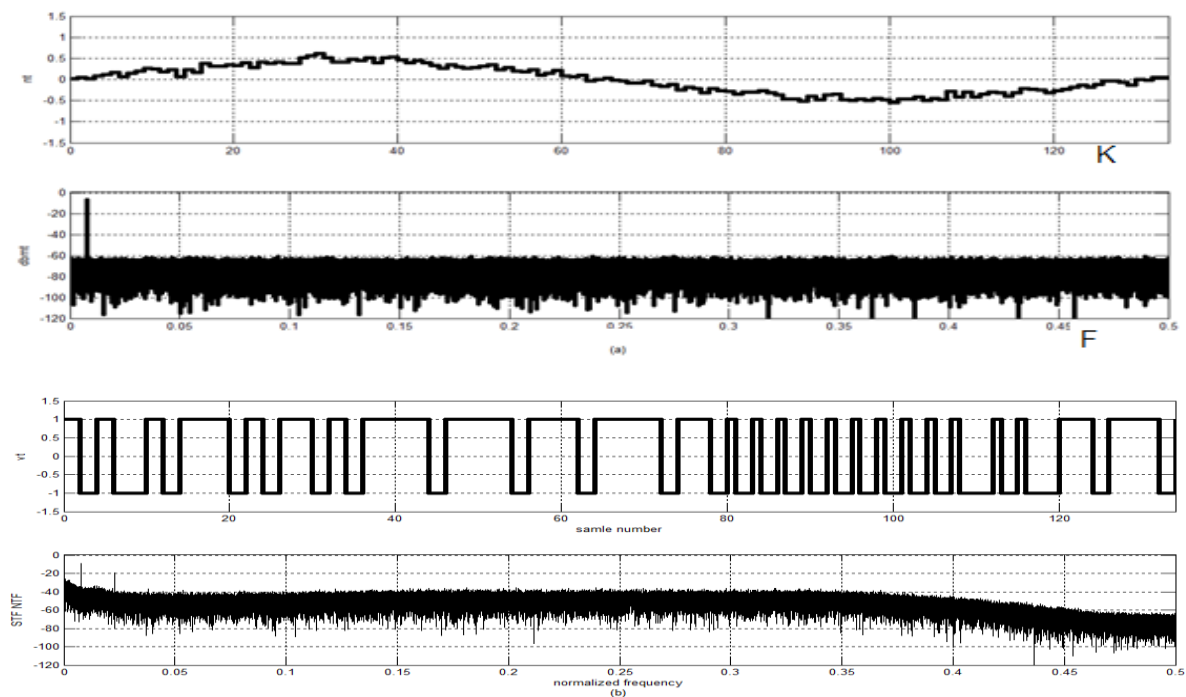


Figure (5.9) Second Order High Pass Sigma Delta Modulator With additive White Noise Wave Forms In Both Time and Frequency Domain

- (a) Modulating Signal
- (b) Modulated Signal

• Test II

Test II same as test I except that the peak amplitude of the signal is doubled i.e. $A_m=0.9V$ and all the other parameters are kept constant also after written a Matlab program achieved simulation results in both time and frequency domains are represented in figure (5.10).

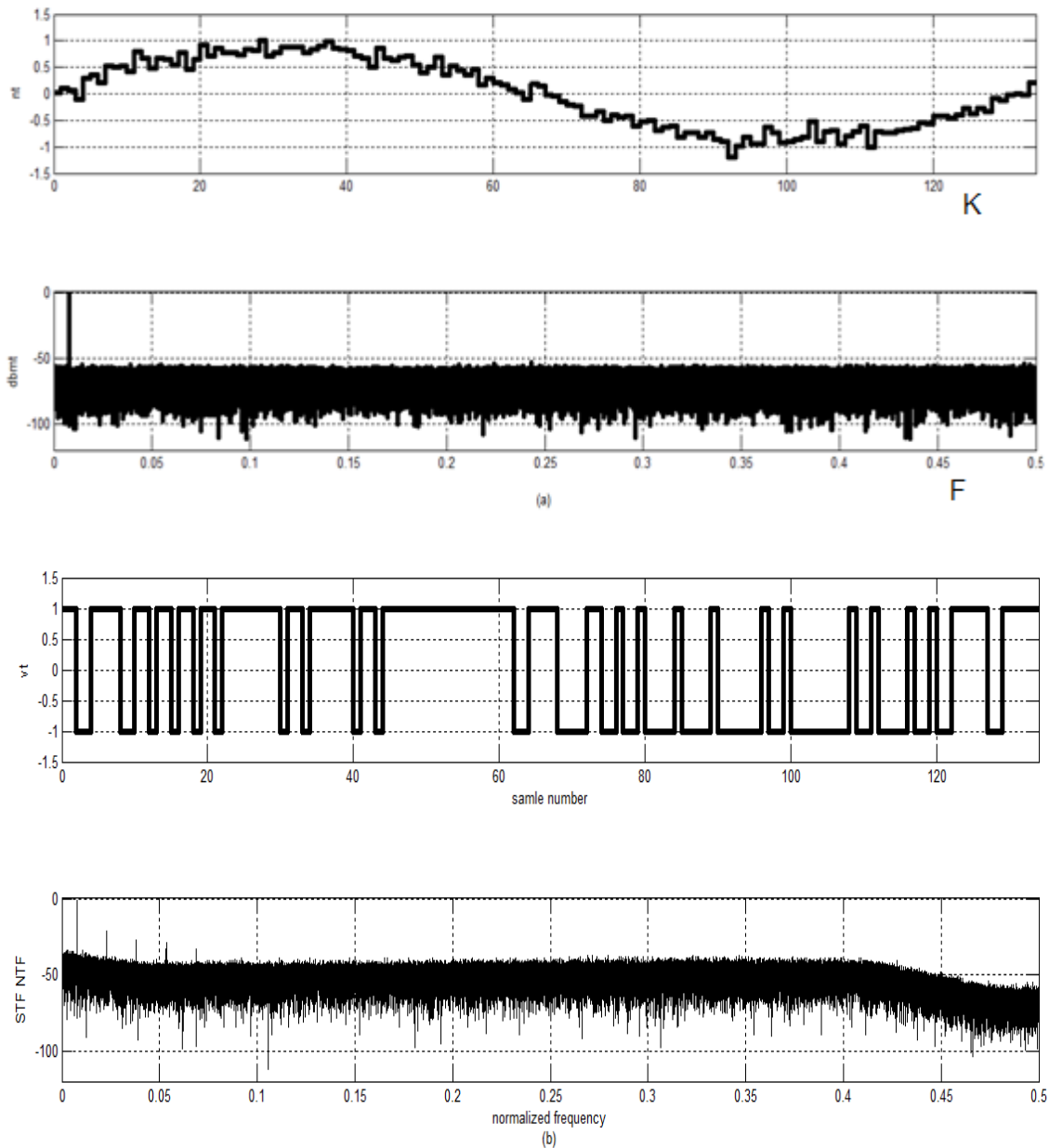


Figure (5.10) Second Order High Pass Sigma Delta Modulator With additive White Noise Wave Forms In Both Time and Frequency Domain

(a) Modulating Signal

(b) Modulated signal

- **Test III**

This test same as test I except that is the modulating frequency is doubled i.e. equal 4kHz while all the other parameters of the simulation are kept constant. After written a Matlab program achieved simulation results is represented in figure (5.11).

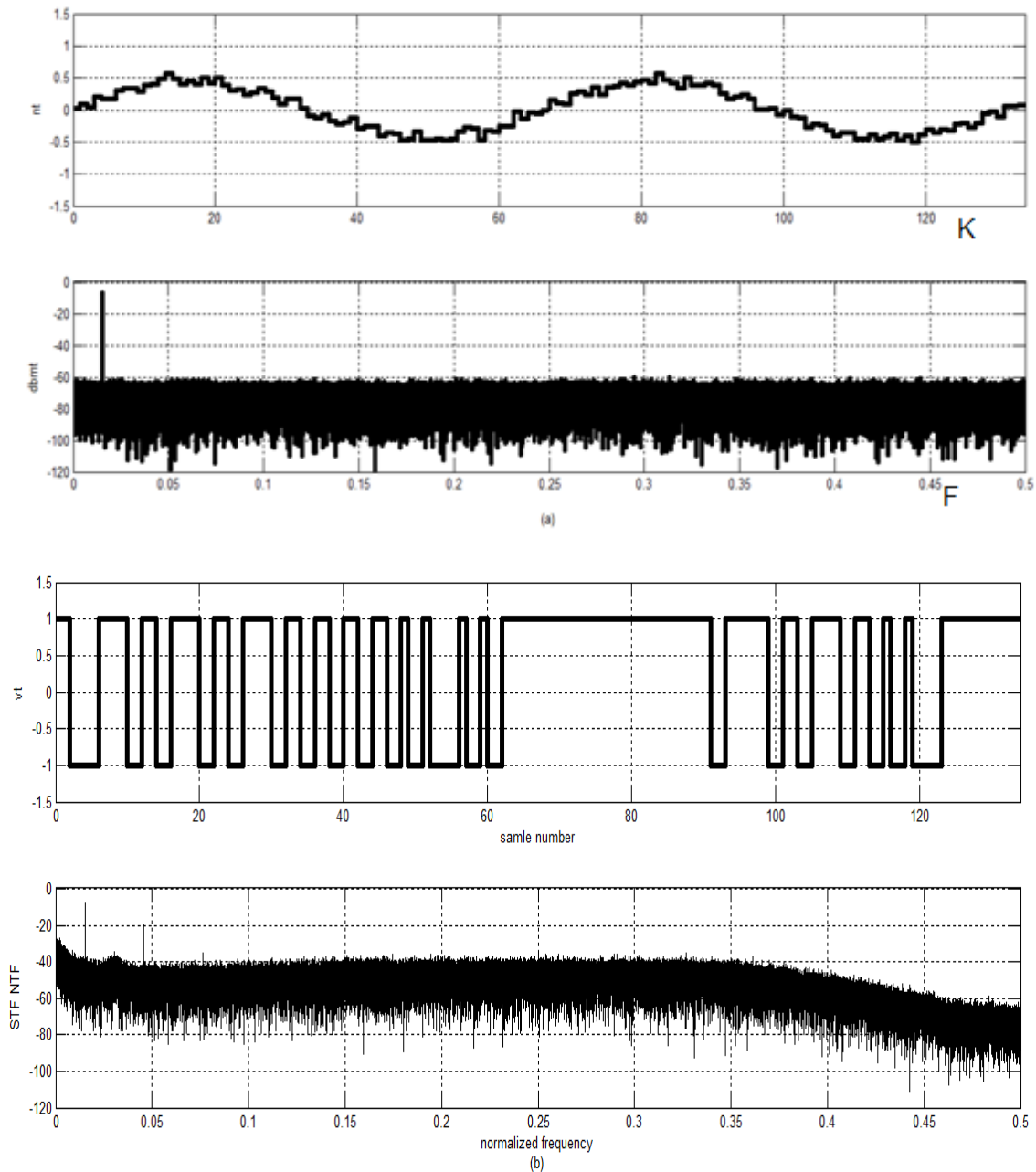


Figure (5.11) Second Order High Pass Sigma Delta Modulator With additive White Noise Wave Forms In Both Time and Frequency Domain

(a) Modulating Signal

(b) Modulated signal

- **Test IV**

This test same as test III except that the peak amplitude of modulating signal is doubled i.e. equal 9V instead of 0.45V, while all the other parameters of the simulation are kept constant. After written a Matlab program achieved simulation results is represented in figure (5.12).

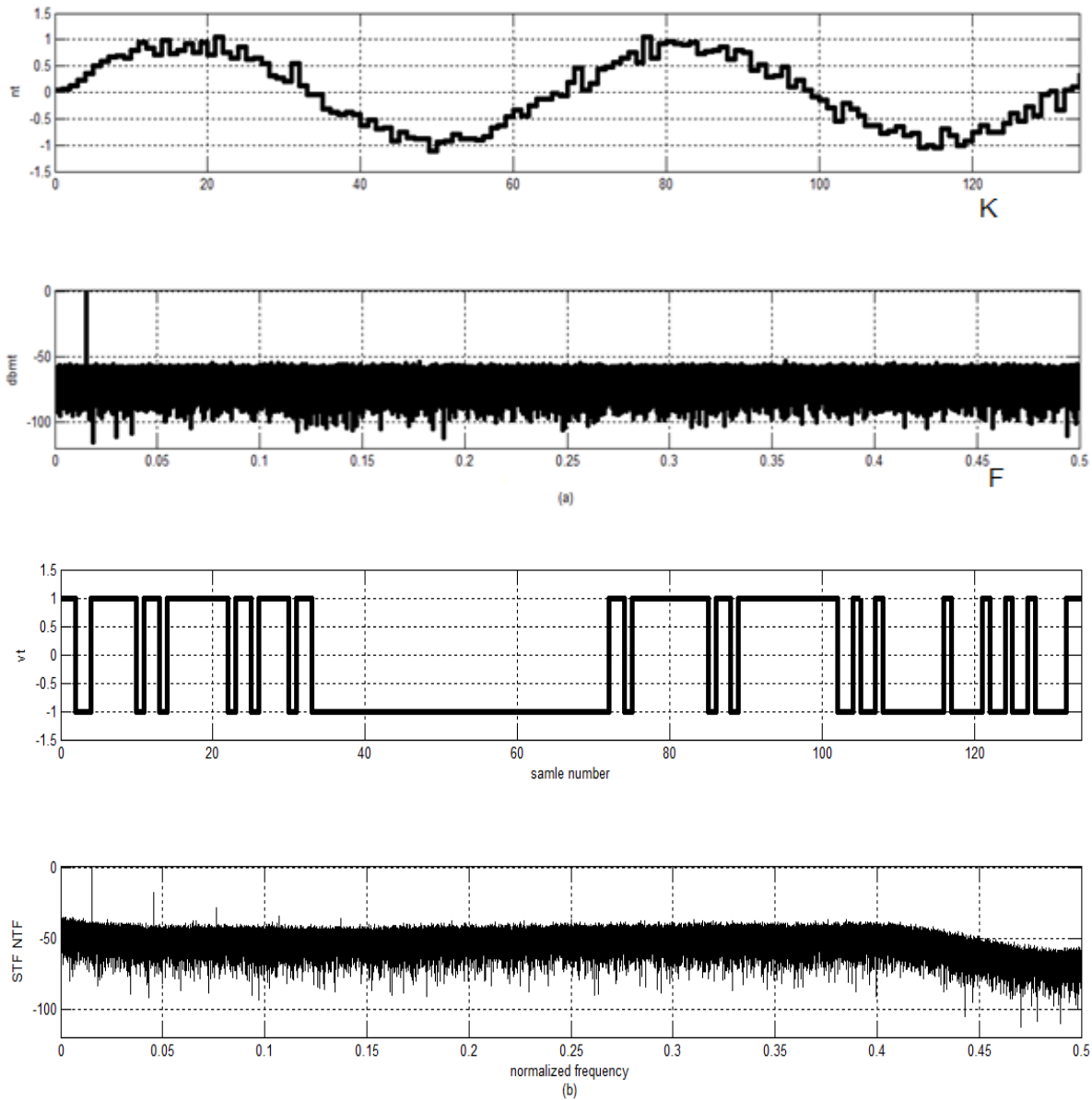


Figure (5.12) Second Order High Pass Sigma Delta Modulator With additive White Noise Wave Forms In Both Time and Frequency Domain

(a) Modulating Signal

(b) Modulated signal

5.4.2 Fourth Order High Pass Sigma Delta Modulation With additive white noise

Same as section(5.3) will generate fourth order high pass sigma delta modulation with additive white noise by written matlab program (see appendix B) ,four testes which will be carried out in the following points:

- **Test I**

This test same as test I in section (5.3) except an adding white noise is added to the modulating signal .

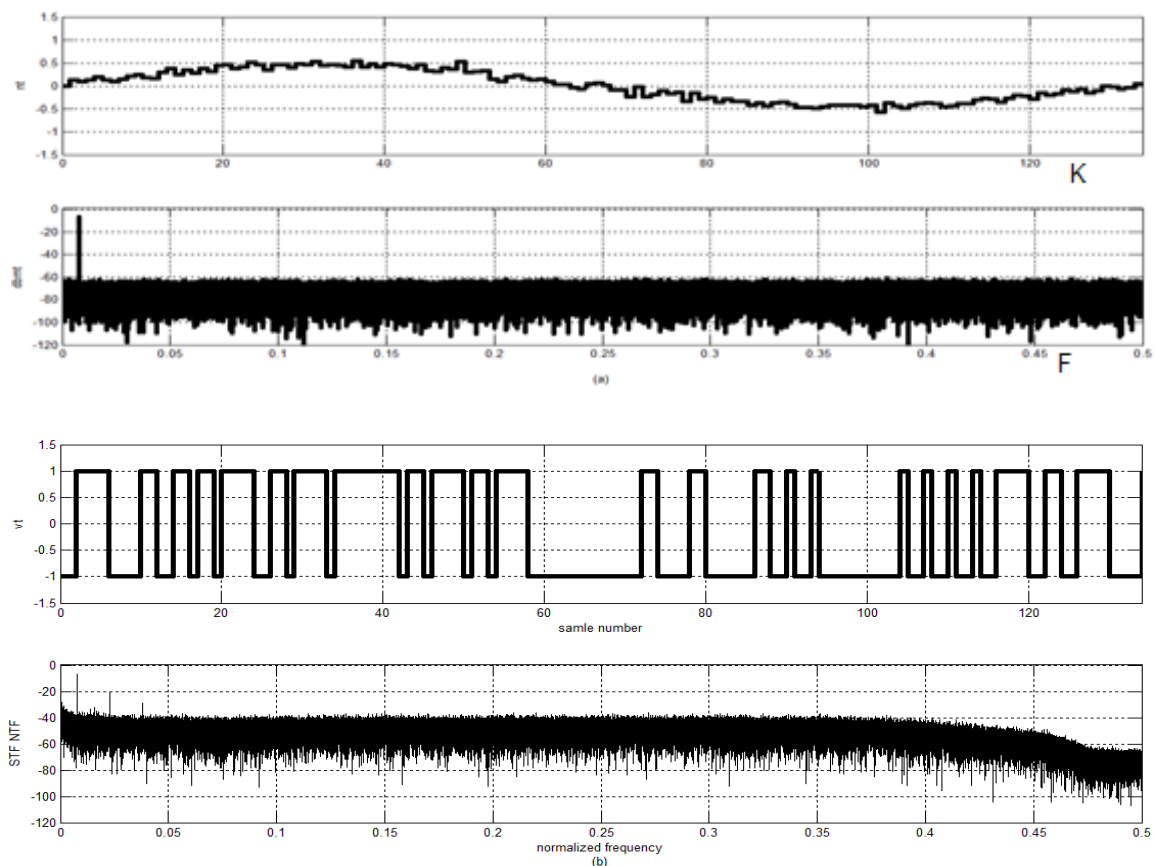


Figure (5.13) Fourth Order High Pass Sigma Delta Modulator With additive White Noise Wave Forms In Both Time and Frequency Domain

(a) Modulating Signal

(b) Modulated signal

- **Test II**

Test II same as test I except that the peak amplitude of the signal is doubled i.e. $A_m=0.9V$ and all the other parameters are kept constant.

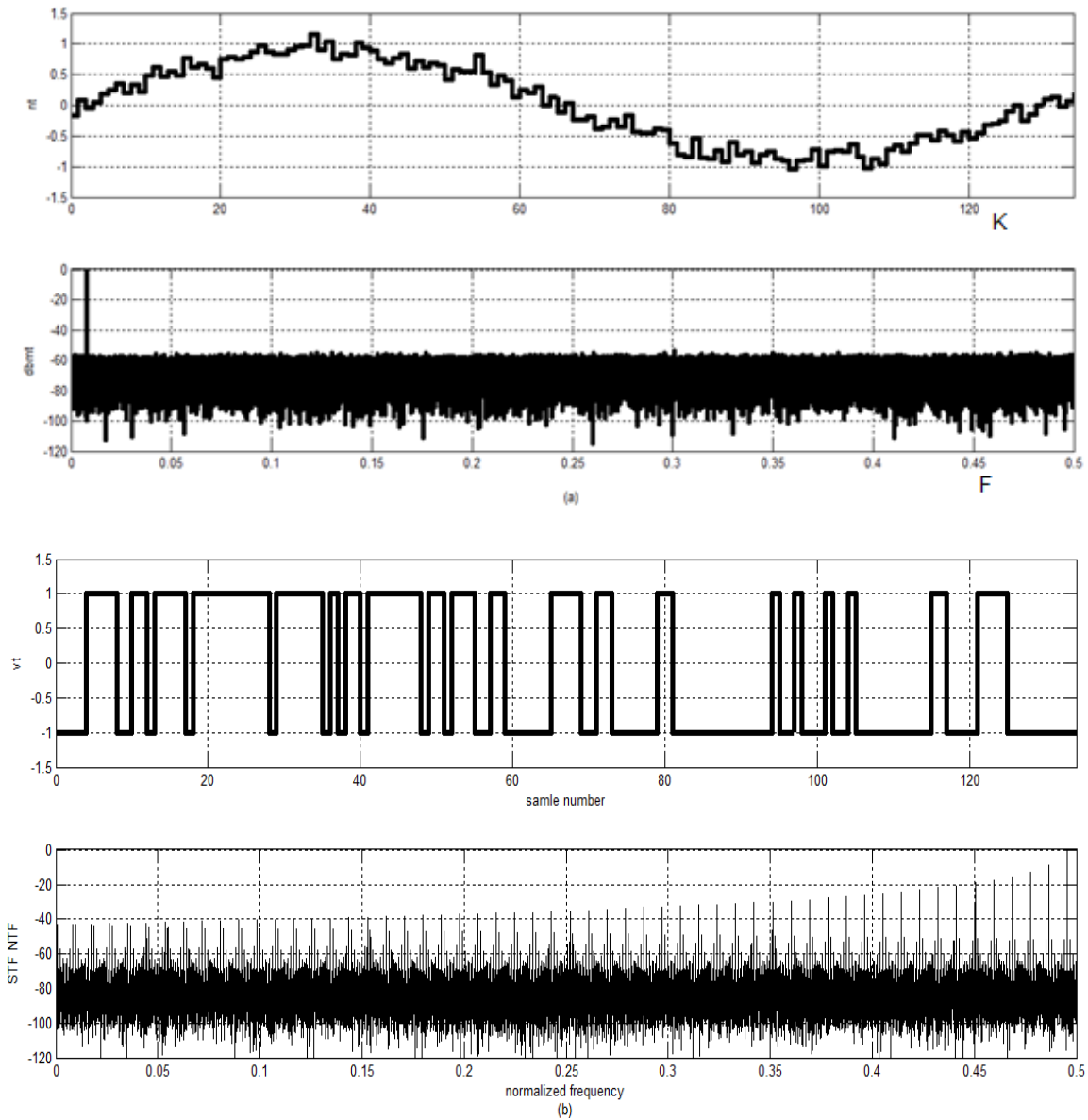


Figure (5.14) Fourth Order High Pass Sigma Delta Modulator With additive White Noise Wave Forms In Both Time and Frequency Domain

(a) Modulating Signal

(b) Modulated signal

• Test III

The same steps carried in test III section (5.3) will repeated here just additive white noise to modulating signal and the modulating frequency is doubled to 4kHz. the achieved simulation results as shown in figure (5.15).

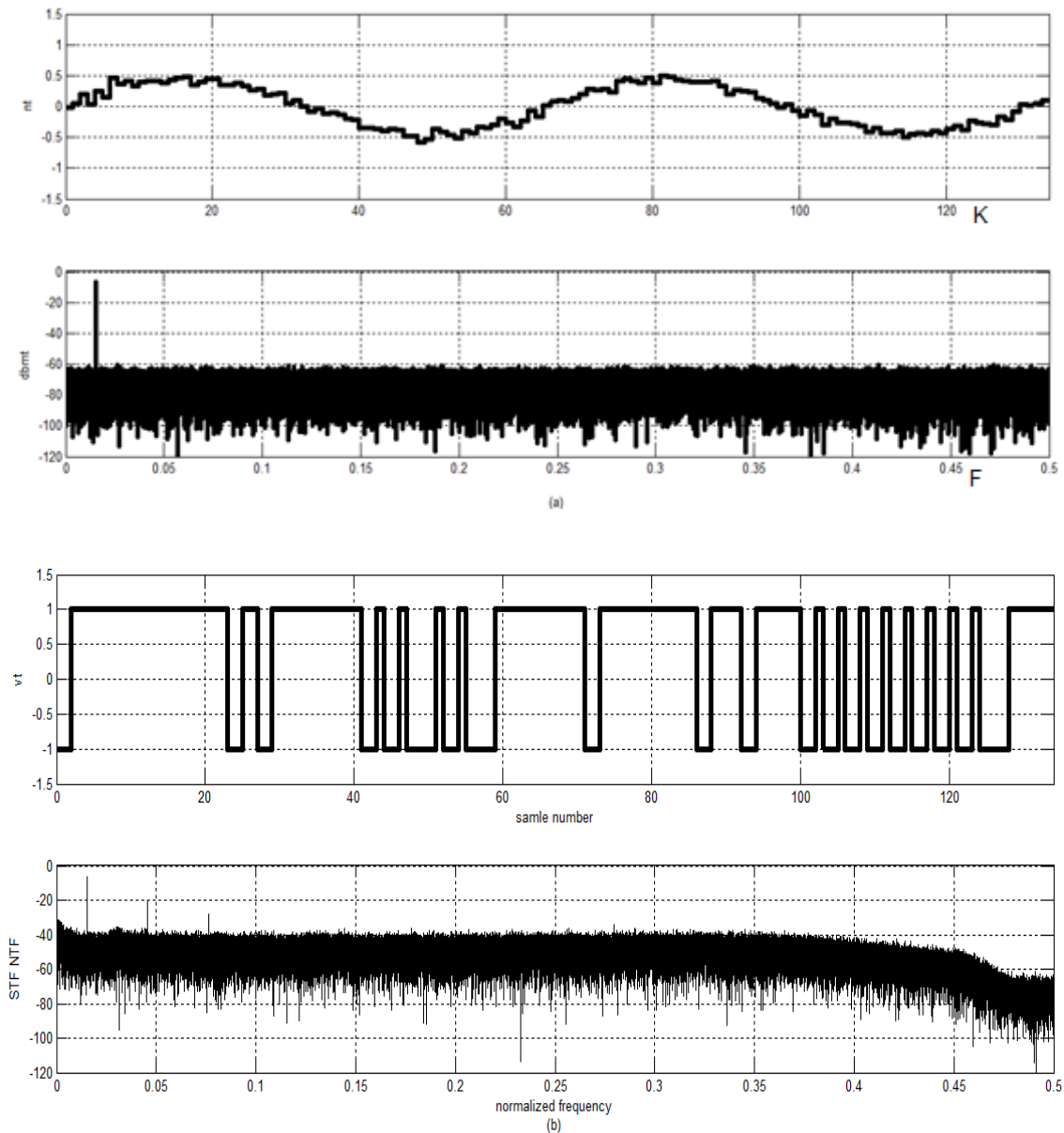


Figure (5.15) Fourth Order High Pass Sigma Delta Modulator With additive White Noise Wave Forms In Both Time and Frequency Domain

(a) Modulating Signal

(b) Modulated signal

• Test IV

Referring to the previous point the amplitude of analog input signal is doubled to 0.9V instead of 0.45V, while all the other parameters of the simulation are kept constant. After writing a matlab program achieved simulation results is represented in figure (5.16).

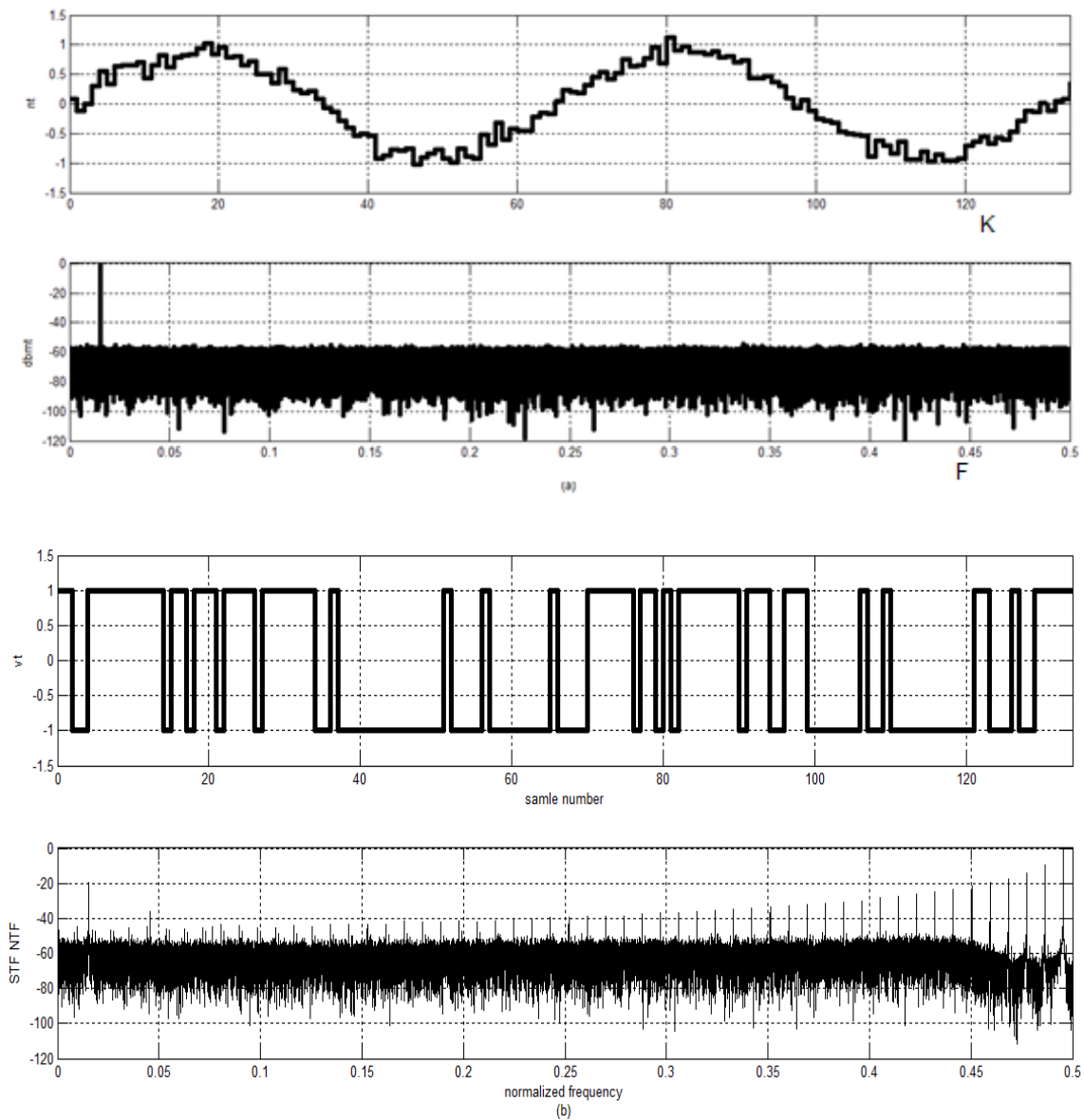


Figure (5.16) Fourth Order High Pass Sigma Delta Modulator With additive White Noise Wave Forms In Both Time and Frequency Domain

(a) Modulating Signal

(b) Modulated signal

5.4.3 Comparison Between 2nd Orders With Additive Noise and Without Noise Signal:

By integrating the second order with additive white noise and second order without noise and assumes the modulating frequency is equal to 2KHz and the amplitude of the modulating signal is equal to 0.45V , will gets the achieved simulation result as shown below in the figure (5.17)

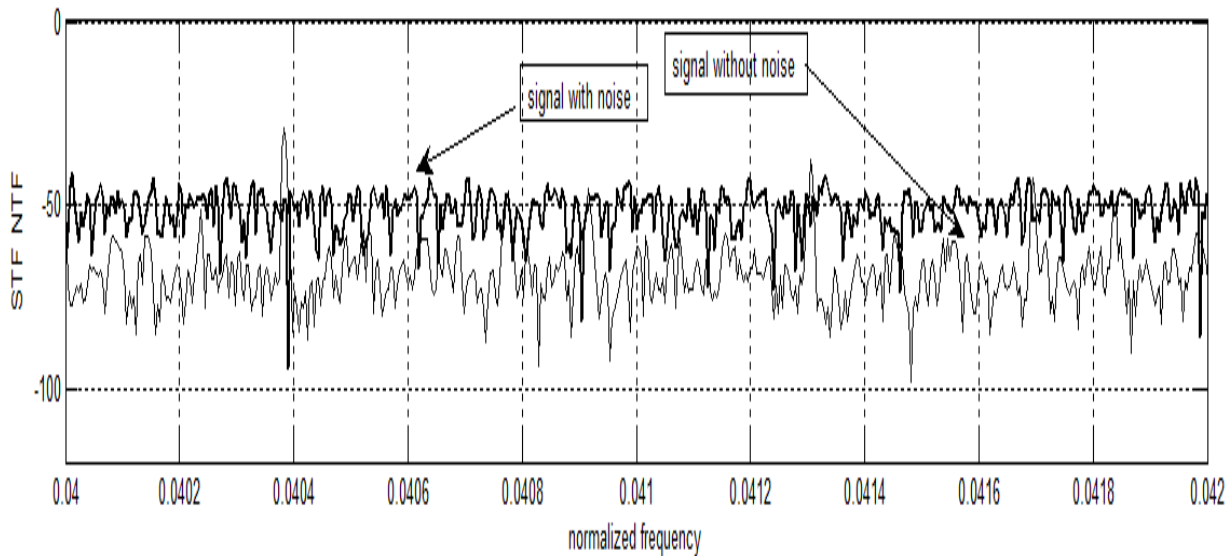


Figure (5.17) Comparison between 2nd orders with additive noise and without noise signal

5.4.4 Comparison Between 4th Orders With Additive Noise and Without Noise Signal:

Same as in subsection (5.4.3) by integrating the forth order with additive white noise and forth order without noise and assumes the modulating frequency is equal to 2KHz and the amplitude of the modulating signal is equal to 0.45V , will gets the achieved simulation result as shown below in the figure (5.18)

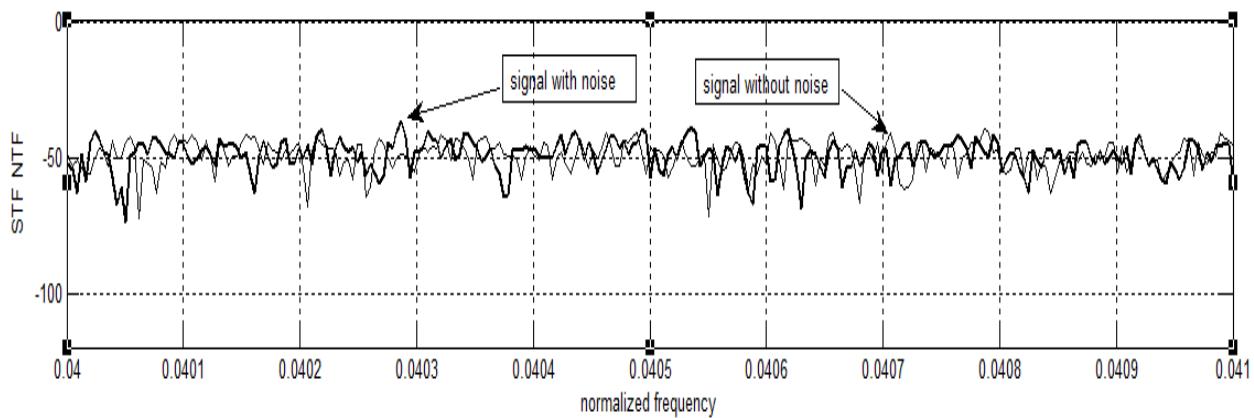
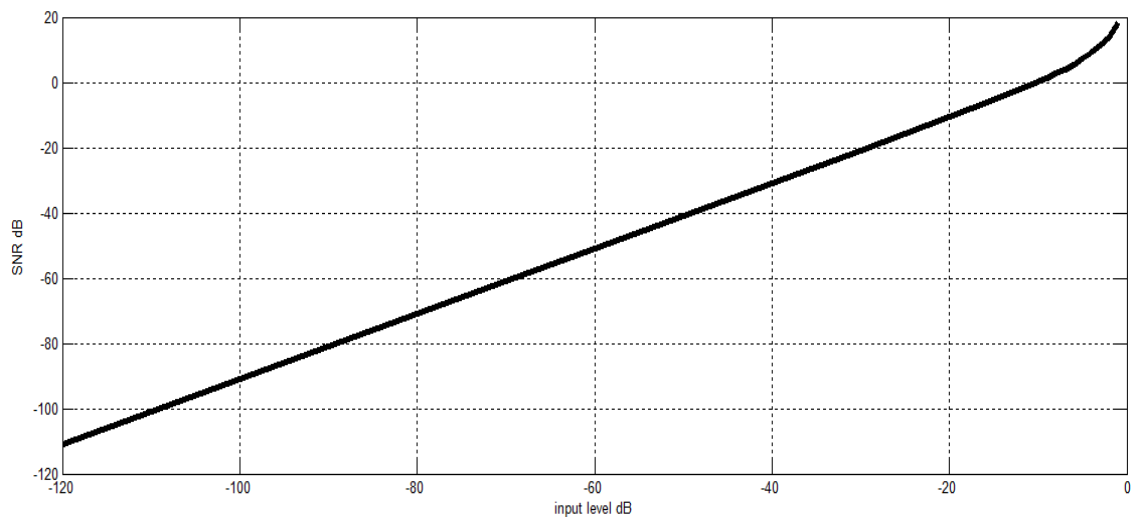


Figure (5.18) Comparison between 4th orders with additive noise and without noise signal

5.4.5 Relationship Between SNR With Various Amplitude of Modulating Signal.

After written a Matlab program to generate the HPSDM modulated signal are measure the SNR at different value of the peak amplitude of the modulating signal . Then the achieved result which represent the peak amplitude of the modulating signal VS the SNR as shown in figure(5.19).



Figure(5.19) Relationship Between SNR with various peak amplitude of modulating signal

CONCLUSION AND FUTURE WORK

6.1 Conclusion

Referring to the main aim of this work which was described in the introductory chapter, we can summarize the conclusion in the following points :

- The output spectrum of the HPSDM consists of the spike which represents the input sinusoid, and some high frequency harmonics.
- The HPSDM shifts the quantization noise to high frequency band away from the baseband signal.
- When the modulating frequency is doubled as mentioned in test III for both 2nd and 4th order HPSDM, we observe that the over sampling ratio is reduced, low resolution occurs and the bandwidth increase. This represent a disadvantage of HPSDM.
- When doubling the amplitude of modulating signal the spike signal in the spectrum will be better than lower amplitudes. This means that the spike signal is more than unwanted components .
- When increasing the order of HPSDM from 2nd to 4th. We see better performance in all results of 4th order when compared with 2nd order. In the same time the dynamic range is increased.
- From figure(5.19) it is seen that the SNR is increased when the peak amplitude of the modulating signal is increased .
- Referring to figure(5.17) illustrate the upper level for 2nd order with additive white noise and the lower level for 2nd order without noise. From the figure we can see that when adding noise at the input with signal, this noise will appear at the output spectrum and distortion occurs, and the spike signal will appear smaller than before noise addition. Before noise addition the signal must be

measured from (-120dB to 0dB) and this is clear in figure (5.9a) in the frequency domain. But after noise addition at the input the signal must be measured from(-60dB). This is because the noise has high frequency components and it passes through the differentiator in the input and appear at the output.

- For 4th order HPSDM figure(5.18) illustrate the upper level for the signal with noise and the lower one for the signal without noise. By adding noise to the signal input this will effect at the output. This is because the noise signal has high frequency components which will pass to the output and appear on the spectrum. Comparing this spectrum with one without noise, it is not smooth and the spike signal is less clear in appearance as before.

6.2 future work.

In this thesis there are some points can be suggested as future work such as:

- Investigation the higher order i.e. 5th,6th order of HPSPDM sigma delta modulation .
- Execution the Program to low pass sigma delta modulator first and second order with different frequency band . and compare with both.