1988

NEAR EAST UNIVERSITY

GRADUATION PROJECT

D P S K MODULATOR & DEMODULATOR

SUPERVISED BY : PROF. DR. MEHMET FAHRETTIN.

PREPARED BY : MOHAMMED TUFAIL HAIDRI.

STUDENT NO : 93- 1176

DEPARTMENT : ELECTRICAL & ELECTRONIC.

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DEDICATION

I dedicate my this project to my parents who have brought their all efforts to me without knowing about return and to those people who devote themselves to others such as my brother AHMER RAZA JAFFRI .

ACKNOWLEDGEMENT

I would like my heartily gratitude to my respectable teacher Prof.Dr Mehmet Fahrettin for his guidance and encouragement at the every stage of this project. I am also very grateful to my friends Saud ,Ahmer, Naveed,Ayaz and Sohail Zikria for their co-operation and love to achieve this goal.

CONTENT OVERVIEW

INTRODUCTION

ACTUALITY OF THE PROJECT: In the modern data communication system the use of Differential phase shift keying has become wide because of its highly advantageous features . Such as high speed of data transmission ,low probity of error , simple realisation of DPSK modems in M- array system (Bell 208A,208B,209A and etc.)and very low signal to noise ratio.These features make DPSKmodulation technique worth use in digital data transmission systems.But on the other hand the main shortcoming of this technique is very difficult detection of the desired signal at the receiver because it required extra methods of investigation and circuitry.

ORIGINALITY OF THE PROJECT : DPSK is the modification of the BPSK which has the merit that eliminates the ambiguity about whether the demodulation of the data is or not inverted .In addition DPSK gives the probability to avoid the coherent reception .In other word to DPSK modulation technique eliminates the use of local oscillator at the receiver . The main scope of this project is the very simple designing of DPSK modulator and demodulator proposed by the supervisor of this project .Even the designing of the demodulator is little bit complicated but in general it is simple to realise .

PRACTICAL VALUE OF PROJECT: This project can be used by the student to study the subject of telecommunication and by the those engineers specialising in the field of data communication system designing.

The outline of each chapter is as fallow: Chapter one profiles the brief history of communication, fundamental concepts of digital modulation methods and timing diagrams of ASK , FSK, PSK,DPSK.

The second chapter treats the detail and realisation of DPSK modulator and demodulator by means of simple circuitry implementation.

Chapter three is devoted to the explanation of error detection and correction technique in data transmission .A detail procedure of error detection and correction system by means of Hamming codes is considered in this chapter.

fourth chapter contains the summary and conclusion of the complete FRAM of work.

A BRIEF HISTORY OF COMMUNICATION

Every day, in our work and in our leisure time , we come in contact with and we use a variety of modern communication systems and communication media , the most common being the telephone ,radio , and television . Through these media we are able to communication (nearly) instantaneously with people on different continents, transact our daily business , and receiver information about various developments and events of note that occur all around the world . Electronic mail and facsimile transmission have made it possible to rapidly communicate written messages across great distances.

One of the earliest invention of major significance to communication was the invention of the electric battery by Alessandro electric telegraphy, which he demonstrated in 1837. The first telegraphy line linked Washington with Baltimore and become operational in May 1844 .

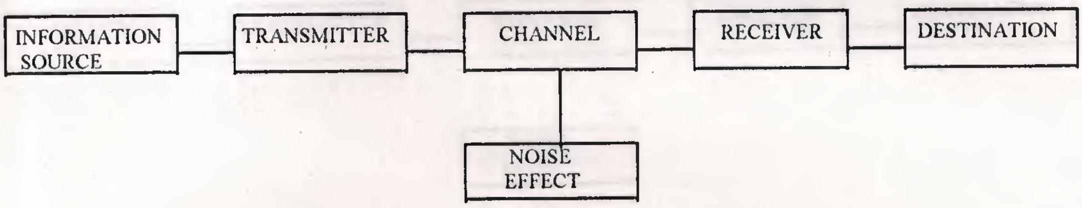
Nearly forty years later , in 1875 , Emile baudot developed a code for telegraphy in which each latter was encoded into a fixed length binary code words of length 5. An important milestone in telegraphy was the installation of the first transatlantic cable in 1858 that links the united states and Europe .Telephony came into being with the invention of the telephone in the 1870's Alexander Graham Bell patented his invention of the telephone in 1876's and in 1877 established Bell Telephone company .Automatic switching was another important advance in the development of telephony . The first automatic switch developed by Strowger 1897 , was an electromechanical step-by-step switch.

The growth in communication services over the past fifty years has been phenomenal .The invention of transistor in 1947 by walter brattain ,john bardeen, and william shockley; the integrated circuit in 1958 by Jack Kilby and Robert Noyce ; and the laser by towns and schawlow in 1958 , have possible development of small size ,low power low wight and high speed electronic circuits which are used in the construction of satellite communication systems ,wideband microwave radio system ,and lightwave communication systems using fiber optic cables.

Currently ,most of the wireless communication system are being replaced by the fiber optic cables which provide extremely high bandwidth and make possible the transmission of a wide variety of information source .

(1.2) WHAT THE COMMUNICATION SYSTEM IS.

A typical model of communication system is shown below



In the above mention communication system the terminologies information, transmitter, channel, receiver, destinationare define as

INFORMATION

The source originates the messages may be human voice, a television picture, a teletype message or data. I f the desired transmit data is not in electrical form than it is changed into electrical wave form by an input transducer that is a device which change physical message into electrical message . The converted signal is then said to be message signal or base signal

TRANSMITTER

It is a device that takes an input signal and process it(modulat or convert to sound etc) or in other word the transmitter modifies the base band signal for an efficient transmission. A transmitter consist of one or more of sub systems ,a preemphasizer, sampler,aquantizer a coder and modulator. Eventually, a transmitter modulates informations into the message carrier waves that is superimposed on high frequency sine wave.The modulation technique may vary from one system to another system.Modulation may be high level or low level and a system may be itself amplitude modulator(A.M) or frequency modulator

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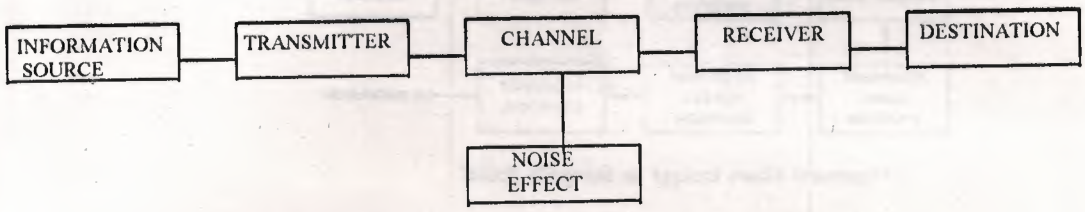

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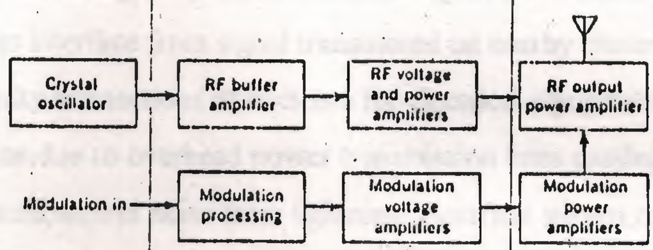
(F.M),puls modulator(P,M) or combination of these modulations techniques depending on the requirement. The given block diagram is an example of high model of A.M modulation broadcast transmitter used in A.M radio transmission system.



Block diagram of typical radio transmitter.

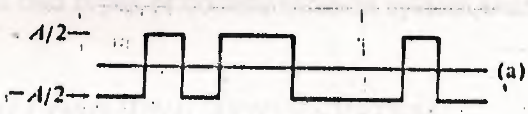
CHANNEL

A specified frequency band or a particular pathnused in communication for the reception or transmission of electrical signal. A channel may be a wire, coaxial cable,a wave guide,an optical fibber or a radio link through which the transmitter output is sent.. It should be noted that the term is often used to a particular service or transmission such as T.V channels .

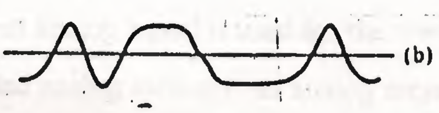
N OIS E

Another term related to the channel is noise effect. Because the theory of communication is inherently concerned with uncertainty because of the masking of the desired signal by unwanted signal that is called NOISE whose time behavior is only statically predictable. If an unwanted signal were exactly predictable it could be subtracted from the total signal in order to recover the desired signal.A noise signal not only destroys the data signals but also contaminates the whole path of transmission .In real life a noise signal is a transmitted signal that overlaps the another transmitted signal and destroy its original form .Basically there are two types of noise signals ,an external noise signal and internal noisesignal.The external noise includes interface from signal transmitted on nearby channels,men made noise is due to the faulty connections of switches for electrical equipments,due to the auto mobil ignition radiation,due to overhead power transmission lines passing over the communication channels,naturel noise from lightning ,electrical stroms,mysterious signals from glaxaies.These external noise problems can be minimized or even eliminated by proper care.The other source of noise is due to the thermal motion of electrons,conductor random emission and diffusion or recombination of charge carriers in electronic devices.These types of noise can also be minimized but can never be eliminated.The pictorial representation of how the noise destroy the transmitted signal is shown below.

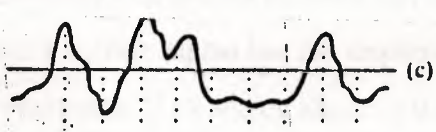
(A)



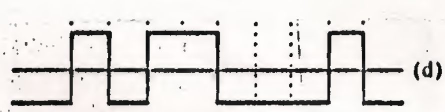
(B)



(C)



(D)



In the above figure (A) is atrasnmtted signal ,(B) is an analog form of signal,(C) is an effected signal due to the noise effect while the signal(D) is received signal which shows time delay due the noise effect.

RECEIVER

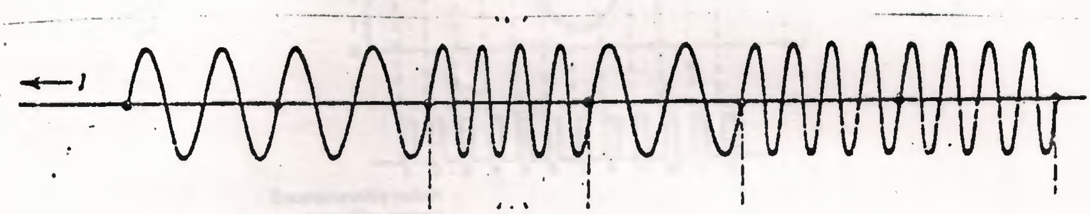
It is another essential part of communication system which converts the transmitted signals wave into desired form of output.The range of frequencies over which a receiver operates with a selected performance , that is known as sensitivity ,is the band width of the process of the receiver.,The receiver output is fed to the out put transducer which converts the electrical signal into its original form that is a message .The basic function of receiver is to demodulate the transmitted signal.

(1.3) ANALOG AND DIGITAL COMMUNICATION SYSTEM.

There are two types of communication system,analog and digital

ANALOG COMMUNICATION SYSTEM

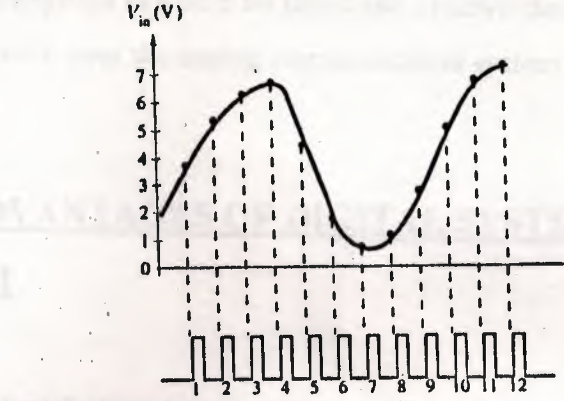
In this system an analog signal is used for·the transmission of message. So that this message is called analog message.An analog message is message whose value varies with continuous time fashion or in other word analog messages are characterized by data whose value changes over a continuos range.For example ,the temperature or the atmospheric pressure of a certain location can vary over a continuous rang and can assume infinite possible a speech signal has the amplitude that vary over the continuous rang .The pictorial representation of an analog signal is shown below



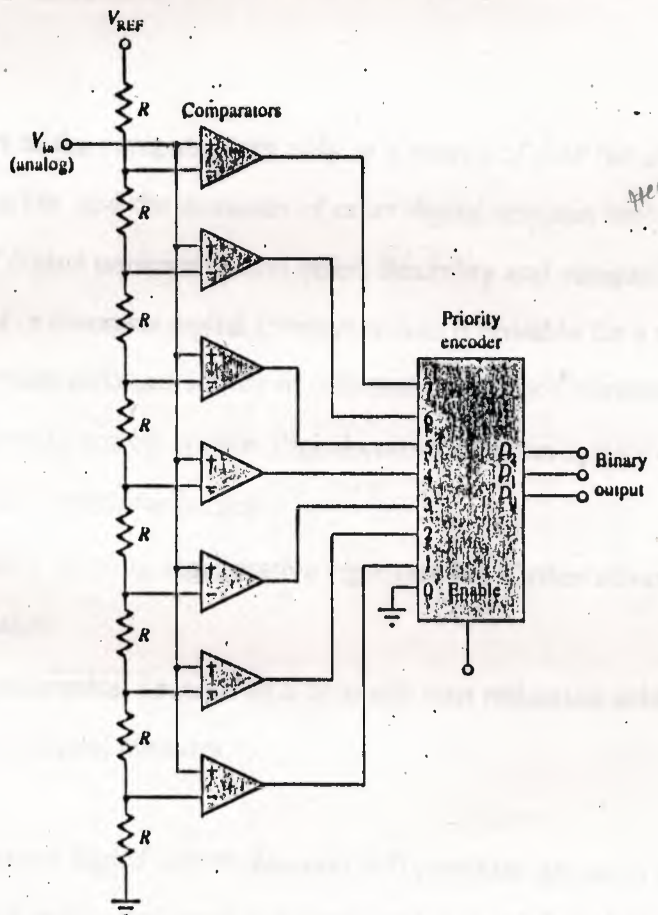
The block diagram of analog communication system is shown on page(2)

(1.4) DIGITAL COMMUNICATION SYSTEM

By using digital technology we can communicate through the discontinuous signals instead of continuous signals ,i.e. signals which appear in discrete steps rather than having continuous variation characteristic of analog signals the value of digital techniques derives from the ability to construct unique codes to represent different items of information.These code are the language of computers(binary number system)and the other types of digital electronic equipment which have revolutionized modern society. So that we can say that in a digital communication system, the message produced by the source are converted into a sequence of binary digits .Ideally ,we should like to represent the source output (message) by as few binary digits as possible .In other word we seek an efficient representation of the source output that result in little or no redundancy. The pictorial representation of a digital message is shown below .



Encoder enable pulses
(sampling pulses)



Here; $n=3$;
# of Resistors
$2^3 = 8$
# of comparators
$2^n-1 = 8-1 = 7$

Because our main frame of work is related to the digital communication system so that we will, discuss this system in a little bit detail and observe the advantages of digital communication system over the analog communication system. .

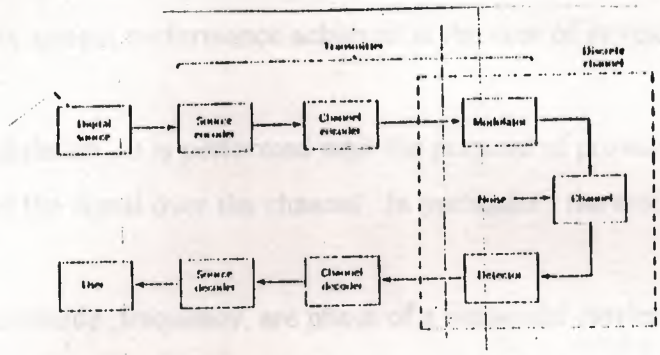
(1.5) THE ADVANTAGES OF DIGITAL SYSTEM OVER ANALOG SYSTEM

The impact of the digital communication has experienced a phenomenal growth in both scoop and application .The growth of digital communication is largely due to the following factors.

* The impact of the computer ,not only as a source of data but also as a tool for communication ,and the demands of other digital services such as telex.
* The use of digital communication offers flexibility and compatibility in that the adoption of a common digital formays makes it possible for a tranmission system to sustain many different source of information in flexible manner.
* In contrast with analog system ,digital communication system can transmit messages with greater accuracy
* The possibility of using regenerative repeaters is a further advantage in digital communication
* It is more economical because of a dramatic cost reduction achieved in the fabrication of digital circuitry.

Indeed the trend toward digital communication will continue ,so much so that the half of the twentieth century will be recorded in history as the era of digital communications

(1.6) BLOCK DIAGRAM OF DIGITAL COMMUNICATION SYSTEM.



(1.7) BASIC SIGNAL PROCESSING OPERATIONS IN DIGITAL COMMUNICATIONS

The above mentioned block diagram shows the three basic signal processing operation source coding ,channel coding, and modulation. It is assumed that the source of information is digital by nature or converted into it by design.

In source coding ,the encoder maps the digital signal generated at the source output into another signal in digital form .The mapping is one to one ,and the objective is to eliminate or reduce redundancy so as to provide an efficient representation of the source output .The primary benefit thus gained from the application of source coding is reduce bandwidth requirement .

In channel coding , the objective is for the encoder to map the incoming digital signal into a channel input and for the decoder to map the channel output into an out put digital signal in such a way that the effect of channel noise is minimized ,that is the combined role of the channel encoder and decoder is to proved for reliable communication over a noisy channel. This provision is satisfied by introducing redundancy in a prescribed fashion in the channel encoder and exploiting it in the decoder to reconstruct the original encoder input as accurately as possible . Thus , in source coding , we remove redundancy whereas in channel in channel coding , we introduce controlled redundancy .

Clearly we may source coding alone ,channel coding alone or the two together .In the latter case ,naturally ,the source coding is performed first followed by channel encoding in the transmitter .In the receiver , we produce in the reveres order ; channel decoding is performed first , followed by source decoding .Whichever combination is used resulting improvement in system performance achieved at the cost of increased circuit comlexcity .

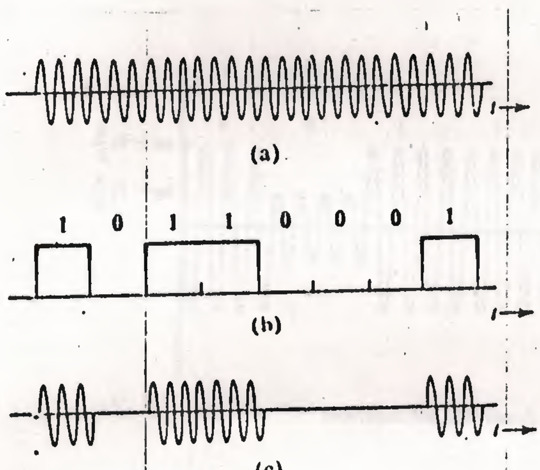
As for the modulation , it is performed whit the purpose of providing for the efficient transmission of the signal over the channel . In particular , the modulator operates by keying
shifts in the amplitude ,frequency, are phase of a sinusoidal carrier waves to the channel encoder output .The digital modulation techniques for the doing Keying is referred tom modulation as amplitude -shift keying ,frequency-shift keying or phase -shift keying respectively .
The detector performs demodulation(inverse of modulation), thereby producing a signal follows the time variations in the channel encoder output (except for the effects of noise).

The combination of modulator , channel ,and detector ,enclosed inside the dashed rectangle shown in fig ,is called discreet channel . It is so called since both its input and output signals are in discreet form.

Traditionally ,coding and modulation are performed as separate operations, and the introduction of redundant symbols by the channel encoder appears to imply increasdsed transmission bandwidth .In some applications ,however ,these two operations are performed as one function in such a way that the transmission bandwidth need not be increased .In situations of this kind , we define the joint function of the channel encoder and modulator as the impositions of distinct patterns on the transmitted signal , which are discernible by the combined actions of the channel decoder detector in the receiver .
In the coming section we will observe some digital modulation techniques such as A.S.K F.S.K and P.S.K

(1.8) AMPLITUDE SHIFT KEYING (A.S.K)

In amplitude modulation ,the carrier amplitude is varied in proportion to the modulating signal (i.e. the baseband signal). The A.S.K modulation scheme is shown below .



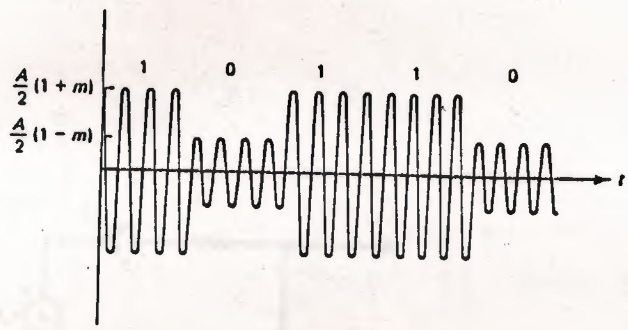
(a) The carrier cos $\omega_c t$. (b) The modulating signal $m(t)$. (c) ASK: the modulated signal $m(t)$ cos $\omega_c t$.

The oboe mentioned scheme shows that if the message signal is binary , the sinusoid signal has one of only two possible amplitudes during each bit period . We can therefor analyze the binary case where the signal is picewise constant by using the following equation for the transmitted signal segment .

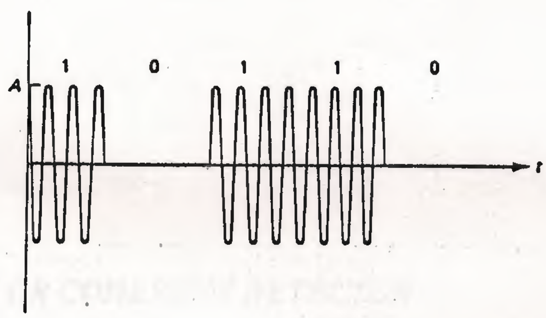
$$Si(t) = A/2\ [1+Mdi(t)]\ \cos(2*3.14159Fct)$$

$2\pi f$

Here i=0 and i=1 to send a binary 0 and 1 respectively . The value of D(t) is +1 or -1 to make the data bipoler . M is the index of modulation . Suppose M=0 so this represents that the transmitted signal is sinusoid and if the value of M=1/2 then the transmitted signal would be a sinusoidal burst of amplitude A/4 for binary 0 and 3A/4 for binary 1 The common case used in A.S.K is M=1 which creates an off and on condition . So that we transmit the data only at the on condition (i.e is binary 1)or off condition (i.e. binary 0) and this type of data transmission is called on_off keying (OOK). The pictorial representation of A.S.K and OOK is shown below .



Representative ASK waveform.

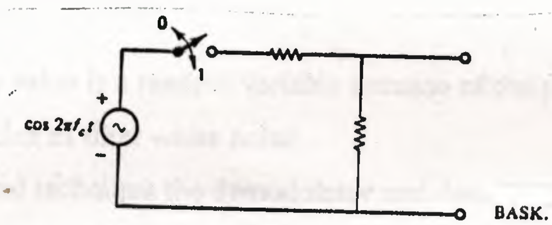


On–off keying.

A. S K MODULATORS

There are two approaches to generating the .A.S.K waveform . One technique starts with the baseband signal and uses this to amplitude modulate a sinusoidal carrier . Since the baseband signal consists of distinct waveform segment ,the A.M also consist of distinct modulation segments.

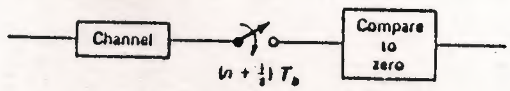
Another approach is to generate the A.M wave directly without first forming the baseband signal. In the binary case ,the generator would only have to be capable of formulating one of the two distinct AM wave segments For on - off keying we need simply switch an oscillator on and off The pictorial representation of OOK modulator is shown below



Modulator for OOK

DEMODULATOR OR COHERENT DETECTOR

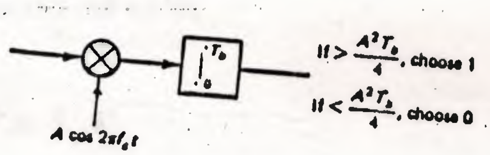
Such as we have there are two types of A.S.K modulators so that in a same way there are two types of A.S.K demodulators . In the first demodulation technique we recover baseband signal by demodulating A.M wave form than we make the sample of baseband signal at the mid point of the each interval . After making samples we compare the values of each sample with other one and if the value of sample is positive than it is assigned binary 1 onthe other hand if the value is found negative than it is assigned binary 0. This operation at the receiver is shown below .



The sample value is a random variable because of the presence of noise .The additive noise can be molded as filter white noise .

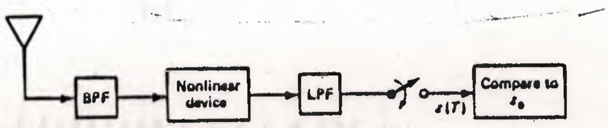
In the second technique the demodulator and decoder uses together in a single operation Since the communication is digital ,the received AM. wave form consists of discrete signal segment . the receiver need simply recognize which of the possible signal segment is being received during each sampling period . We already know that the optimum receiver for this purpose is matched filter detector .Because the energies of the two signals are not equal(one of them is zero) ,so that threshold is not zero against the compared output .

Because in order to use matched filter it is necessary that we reconstruct the carrier at the receiver and it can be done by using band pass filter or phase lock loop . The pictorial representation of matched filter detector is shown below .



INCOHERENT DETECTOR

It is very simple to design and does require the carrier to be reconstructed but it has higher rate of bit error than that of coherent and this make it very limited



(Envelop detector for OOK)

ADVANTAGES AND DIS ADVANTAGES OF A.S.K.

A.S.K has one main advantage that it is very simple technique in hard ware realization . On the other hand it has two main disadvantages.

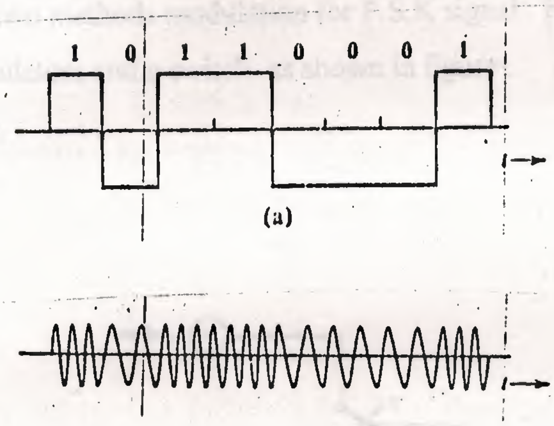
* It is very susceptible to noise interface .
* It uses too much band width.

RATED SPEED OF A. S. .K.

The theoretical rated speed of A. S .K (In telephone lines) is 1500b/s.

(1.9) FREQUENCY SHIFT KEYING (F.S.K)

When the data is transmitted by varying frequency , we have the case of frequency shift keying (F.S.K).The pictorial representation of this scheme is shown below .

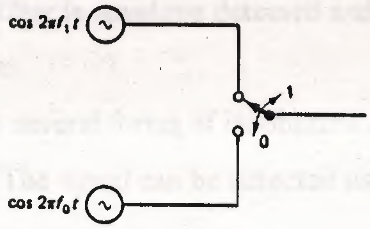


In this scheme the binary 0 is transmitted by pulse frequency Wc0 and the binary 1 is transmitted by the pulse frequency Wc1. All the data of information resides in the carrier frequency . The F.S.K signal may be considered as a sum of two A.S.K signals ,one is of the frequency Wc0 and other Wc1 . The proper use of frequencies eliminates the existence of the discrete components . An other important consideration is that the band width of this modulation is higher than that of A.S.K or P.S.K .(In analog system the F.M signal usually be used in place of A.M , because of the better surviving performance in presence of the noise)

MODULATION AND DEMODULATION

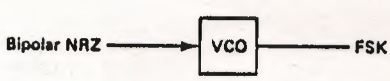
MODULATION

There are two methods modulation for F.S.K signal . First using simple modulator consists of two oscillators and a switch as shown in figure .



Modulator for FSK.

The other method is using F.M modulator (VCO) with bipolar baseband signal as in put signal



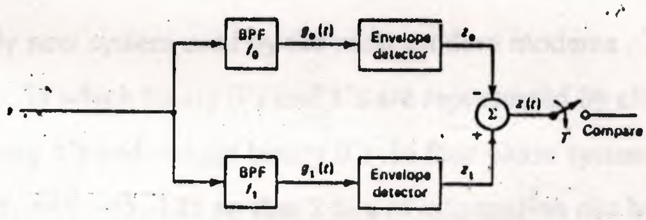
Use of FM modulator to generate FSK.

In the above mentioned scheme all the consideration has been taken as if the frequency is changing instantaneously and the baseband is composed of perfect pulses but in the real life the frequency transition is smooth . So that it is necessary to implement this in proper real device.

DEMODULATOR

Such as modulator there two types of an F.S.K demodulator .One is coherent detector that requires the reconstruction of two separate carriers at the receiver . But using an incoherent detector eliminates this condition . This is possible using two band-pass filter at the input ,one tuned to each of the two frequencies used to communicate 0's and 1's .The output of the filter is envelope detected and then baseband detected using integrate and dump operation

In general there are several forms of incoherent . One is used discriminator followed by envelope detector. The signal can be detected using any of the baseband techniques.



Incoherent detector for FSK.

(433 roden)

ADVANTAGES AND DISADVANTAGES OF F. S. K

It has two main advantages.

* Simple implementation in hardware
* It has much less susceptibility to noise than A.M .

On the other hand it has the following disadvantages

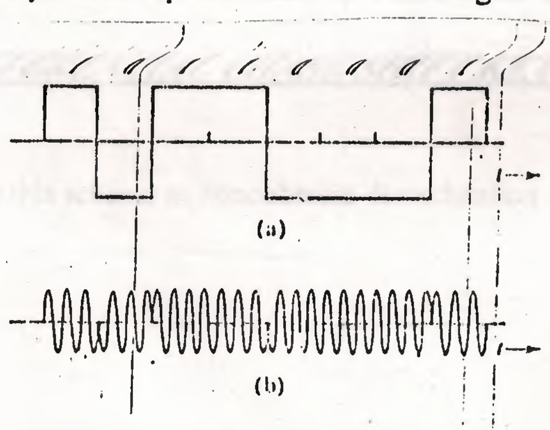
* It uses the band width as much as A.M does
* The speed of the F.S.K is much lower then A.M.

THE RATED SPEED

The rated speed of F.S.K (Theoretically) is 1500b/s while the practical rated speed of the F.S.K modem is 300b/s for the fullduplex,for the half duplex it is 1200b/s.

(1.10) PHASE SHIFT KEYING (P.S.K).

It is a relatively new system used by the most modern modems . This modulation can be quite complex , in which binary 0's and 1's are represented by changing the phase of pulse by +90 for binary 1's and -90 for binary 0's .In four phase system this shifting can be of +135, +45 ,-45,-135 so that 2 bits of information can be indicated instead of one as in two phase system. In this system the information resides in the carrier phase of the signal .The pictorial representation of P.S.K signal is shown below.



The P.S.K is a case of polar signaling .The binary P.S.K(B.P.S.K) transmitted signal is a is a sinusoid signal of fixed amplitude. The mathematical representation of P.S.K is shown below .

U(t)= b(t)*A*Cos(wt+Q)

b(t) = {1 if "1"
{-1 if "0"

The P.S.K signal can not be demodulate noncoherently because the envelope is the same for both 1's and 0's. In proceeding chapter we will observe modulation and demodulation techniques of P.S.K in detail.

ADVANTAGES AND DISADVANTAGES OF P.S.K

The main advantages of P.S.K over other modulation techniques are

* It has relatively simple modulator relaxation.
* It has much less susceptibility then A.S.K and F.S.K.
* It requires less bandwidth then A.S.K and F.S.K.

On the other hand the main disadvantages of P.S.K

* This scheme is relatively more complex for demodulation then other .

(1.11) DIFFERENTIAL PHASE SHIFT KEYING (D.P.S K)

We can compare this scheme as noncoherent demodulation of P,.S.K .because in this

This modified version of P.S.K provide the stability against uncertain change in phase of transmitted signal, i.e. the cause of error in P.S.K modulation .An other main advantage this modulation scheme that it eliminates the requirement of coheren modulation.

(1.12) ARGUMENTATION TO PROSPECTIVE DIGITAL MODULATION SCHEME.

Now in this section we will observe the advantages and disadvantages among the above mentioned modulation schemes then we will observe the most comprehensive scheme in detail as a main frame work of our activity .As a whole each of the above techniques its own advantages and disadvantages or in the word we can say that each scheme offers system trad-offs of its own.The final choice made by the designer is determined by the way in which the available primary communication resources, transmitted power and channel bandwidth, are the best exploited . In particular , the choice is made in favor of the scheme that attains as many as of the following design goals as possible:

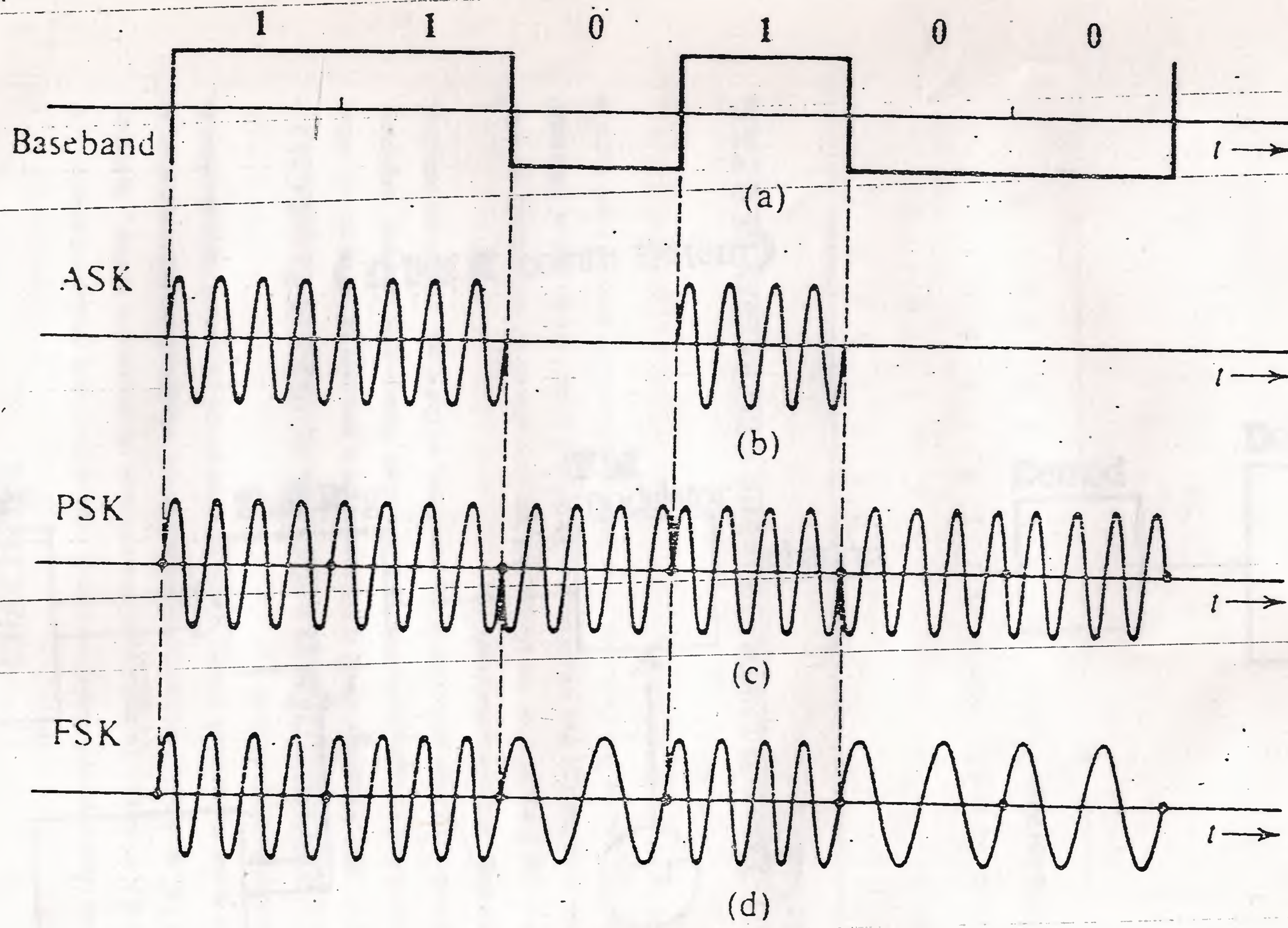
(1) Maximum data rate .
(2) Minimum probability of symbol error .
(3) Minimum transmitted power .
(4) Minimum channel width.
(5) Maximum resistance to interfering signals.
(6) Minimum circuit complexity.

We will take in account all these futures during our compression of above scheme.

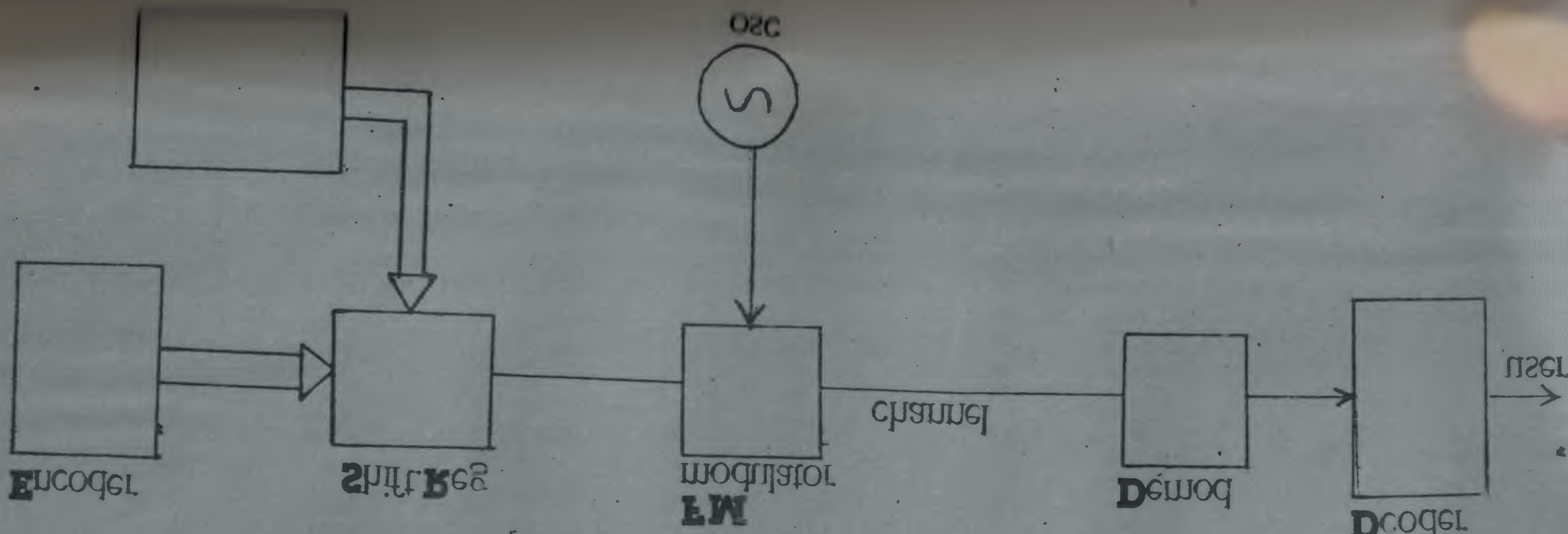
Starting from A.S.K we observe the rated speed of A.S.K ,i.e ,1500b/s which is relatively good speed then F.S.K which proved the rated speed of 1200b/s but on the other hand P.S.K has much higher speed then both of them ,i.e. approximately 2500b/s.Compereing the noise susceptibility of the three schemes we observe that P.S.K is more stable to the noise effect then other both schemes which minimize the probability symbol error during the data transmission .Realizing the economical concerns that is required during the data transmission we select the P.S.K which requires the minimum transmitted power then that of other two schemes.An other important consideration we must take in account during the designing of communication system is channel width which should be minimum as much as possible so that P.S.K provide this future that requires minimum band width as compare to A.S.K and F.S.K. Minimum circuit complexity also pays an important character during the data transmission operation because "The simpler the circuitry the lesser the error rate " .The realization of modulator and demodulator of P.S.K is much simpler than that of A.S.K and F.S.K .But on the other hand most advantageous P.S.K scheme has one main disadvantage ,i.e. , uncertain change of phase during data transmission but we can overcome this problem by using modified version of P.S.K that is D.P.S.K which covers all the future of P.S.K and eliminates the uncertain change of phase during data transmission .

After the above arguments we concludes that D.P.S.K provides .a comprehensive mode of operation during data transmission by every aspect .So that in our proceeding chapter our main frame of work will be D.P.S.K modulation and demodulation operations

... " END OF CHAPTER ONE"



1
1
0
1
0
0
Baseband
t
(a)
ASK
t
(b)
PSK
t
(c)
FSK
t
(d)



(DPSK comm system)

CHAPTER #2

In this chapter we will discuss our sleeted most advantageous modulation scheme ,I, e, .D.P.S.K in detail . To under stand the operation of transmission system , based on D.P.S.K. modulation scheme better we will briefly discuss all the major components used in the transmission system. These components are mentioned blow in order as shown in the block diagram

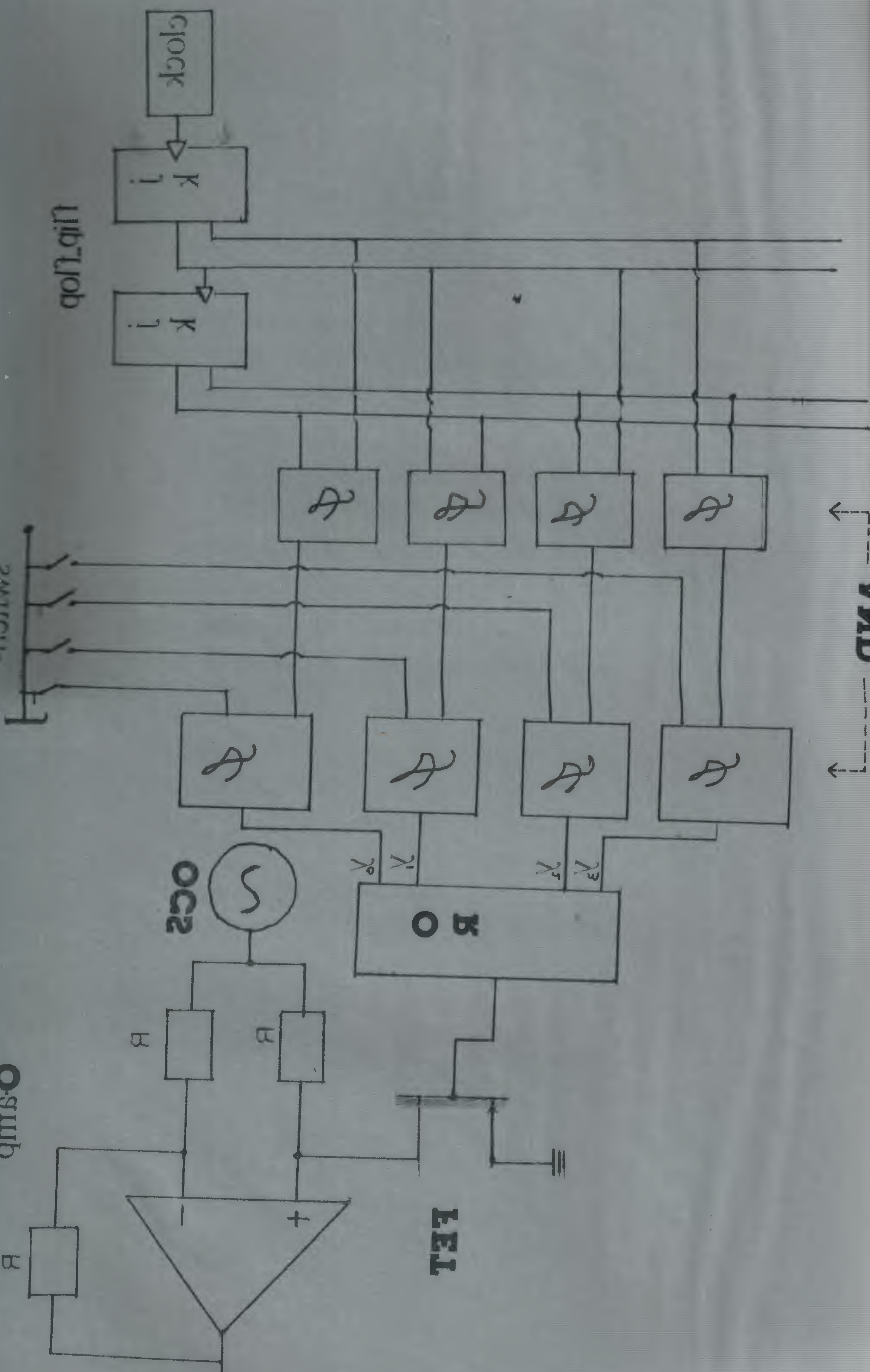
(2.1) DIFFERENTIAL PULSE CODE MODULATOR (D.P.CM)

In P.C. M. A massage signal is represented by a sequence of coded pulses, which accomplished by representing the signal in discrete from in both time and amplitude. The basic operation performed in the transmitter of D.P.C.M. system are sampling, quantizing , and differential encoding .

But in our fram of work we assume that we have already an differentially encoded signal as an input so that we no need to go through in the back ground of differential encoding system, But illustration an example of differentially encoded input signal is given below.

(2.2) TABLE ILLUSTRATING THE GENERATION OF D.P.S.K. SIGNAL

{ bk }		1	0	0	1	0	0	1	1
{ dk- 1}		1	1	0	1	1	0	1	1
Differentially encoded Sequence {dk}	1	1	0	1	1	0	1	1	1
Transmitted Phase Cradious	0	0	pi	0	0	pi	0	0	0



(2.3)

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(2.3) *AN INSIDE VIEW OF D P S K MODULATOR*

Because we have assumed that our input signal is already DPC modulated so that we need not go in detail of DPC modulated system..Starting from the structure of D P S K modulator we will observe the all parameters of used in modem and three function . Basically we may split out the D P S K modulator into three of operations .The first two parts are logical circuits that produced logic "1" or "0" at the out put line according to the input signal
while the other parts produce the phase shifted version of the input clock signal .Normally the three parts of D P S K modulator are

* Shift register
* Multiplexer
* FM modulator

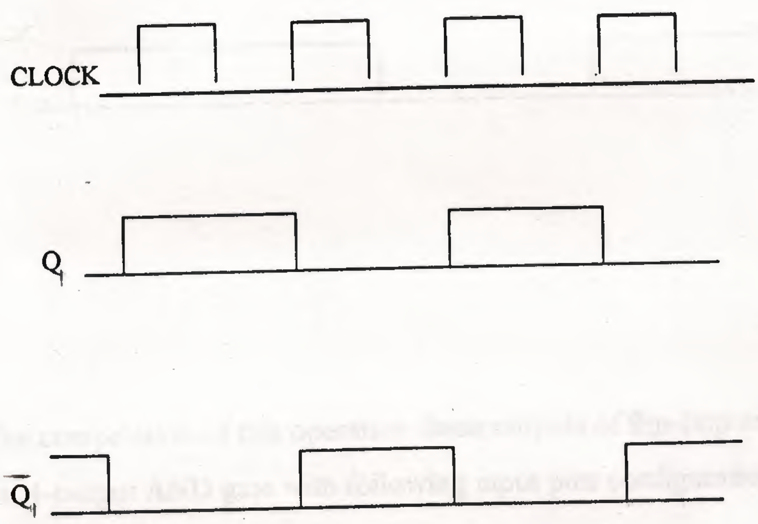
Now in the coming sections we will observe the function of all the parameters of D P S K modulator .

(2.31) SHIFT REGISTER.

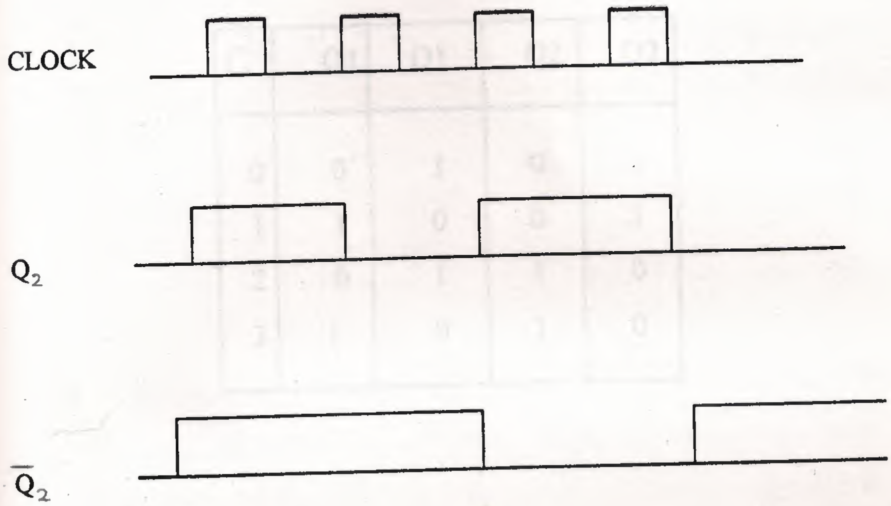
A shift register is sequential logic module constructed from flip-flop that manipulates that bit position of binary data by shifting the data bits to the left or right or this is a temporary memory register that shifts its contents to right or left .
In this case we are establishing very simple shift register according to our goal .This shift register requires only two negative-edge flip-flops parallely attached to each other in such a manner that the output "Q" of first flip-flop becomes the input of the second flip-flop.It is important to note that in our case we are using synchronised flip-flop that

requires only one clock signal as an input .Such type of flip-flop is also called trigger edge flip-flop. The basic operation of edge-tigger flip-flop is as fallows that output appears "Q" by toggling each time when a negative going plus comes along .On the other hand the complimentary out put appears at "Q".

The pictorial representation of the behaviour of negative -edge- flip flop on clock input is shown below .



After this processing the output of "Q" becomes the clock input of "Q" and here the same function of flip-flop starts again as shown in the figure below.



After the compelation of this operation these outputs of flip-flop are connected to the 8-input ,4-output AND gate with following input pins configuration.

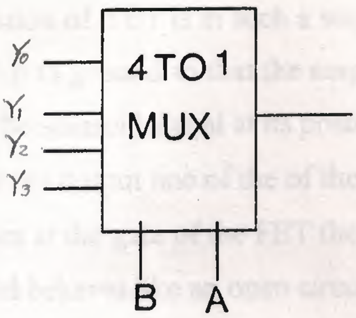
$$Y0 = \overline{Q}_1 \cdot Q_2$$
$$Y1 = Q_1 \cdot \overline{Q}_2$$
$$Y2 = \overline{Q}_1 \cdot \overline{Q}_2$$
$$Y3 = Q_1 \cdot Q_2$$

Here the Y0 ,Y1,Y2 and Y3 are the output of the AND gates while Q1 ,Q1 ,Q2 and Q2 are the input of the gates. The truth table of the flip-flop are shown below .

C	Q1	Q1	Q2	Q2
0	0	1	0	1
1	1	0	0	1
2	0	1	1	0
3	1	0	1	0

(2.32) MULTIPLEXER.

After the appearance of outputs on Y0,Y1,Y2 and Y3 the next task is to select only the one output at the same time and this can be done by using data selector or multliplexer.In general a multiplexer is modular device that selects one of the many input lines to appear on a single output line .the functional diagram and truth table of mux is shown below .

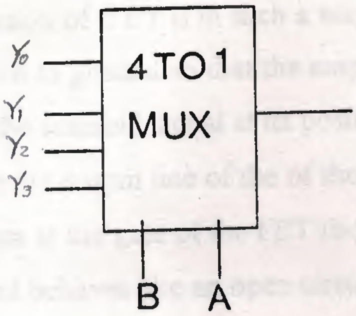


B	A	Y
0	0	Y0
0	1	Y1
1	0	Y2
1	1	Y3

After the appearance of single output at the mux output line the data is sent to the gate of a F E T transistor where the FET works as switching operator .

(2.33) F.M MODULATOR

The FET As we have mentioned above that in D P S K modulator FET works as switching operator for the amplifier, attached with it, and this is called analoge switching operation.It received the input from the output of multiplexer at the gate and performs the switching operation according to the gate voltage level.The reason to chose the FET is that it has very high input impedance ,from1 megaohm to several megaohms,an other advantage of FET is that it has very low susceptibility to noise that is the very important consideration in any communication system this make it superior to BJT .It is important to note that a B J T is basically an anaolge operating divce that works on negative and positive response of the sinusoidal signal.



B	A	Y
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0	1	Y1
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1	1	Y3

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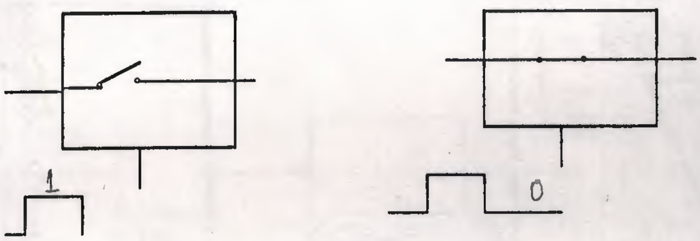
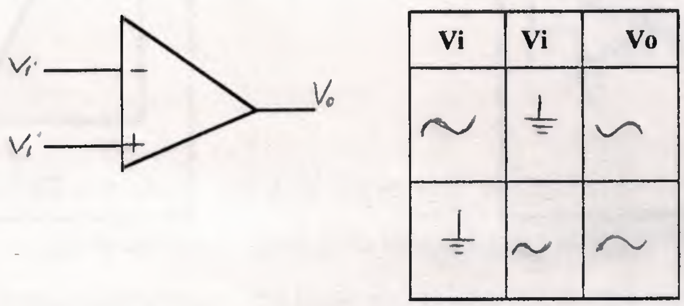
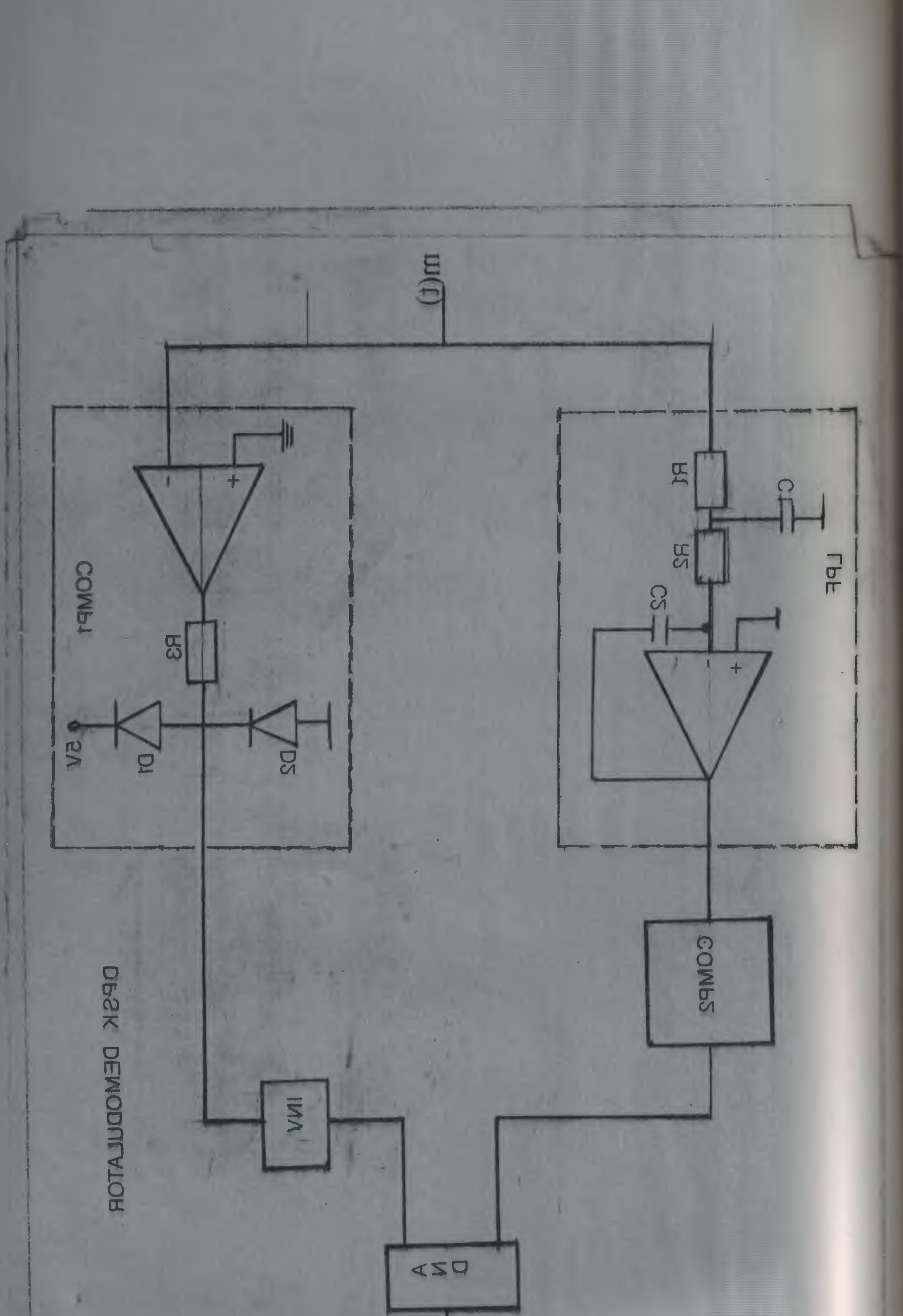
The operation of FET is in such a way that when it gets logic "1" from mux become short circuit to ground so that the amplifier ,which is connected to the an oscillator, provides the sinusoid signal at its positive terminal so that a noninverting output appears at the output line of the of the amplifier .On the other hand whenever the logic "0" appears at the gate of the FET the internal resistance of the transistor become infinite and behaves like an open circuit in this case the signal of amplifier passes through the negative terminal of amplifier and provied an inverting out put .
The figur below shows the complete operation of the amplifier.



(F E T)



(Amplifier)


m(t)
COMP1
R3
D1
D2
5V
LPF
R1
R2
C1
C2
COMP2
INV
AND
DPSK DEMODULATOR

(2.

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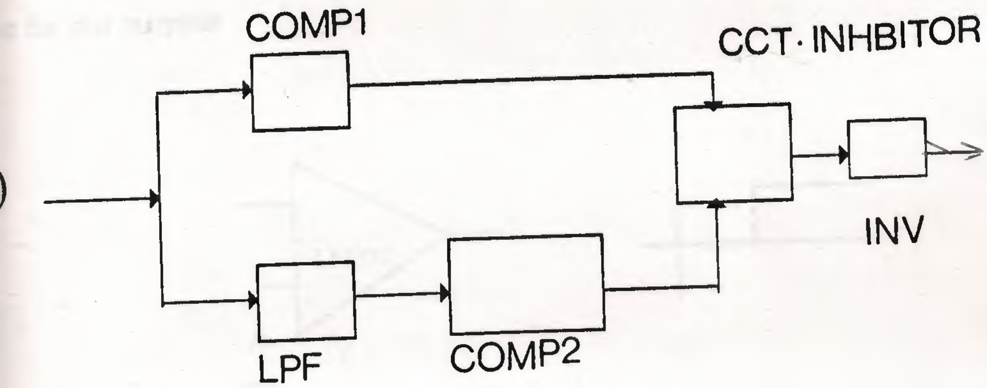
m(

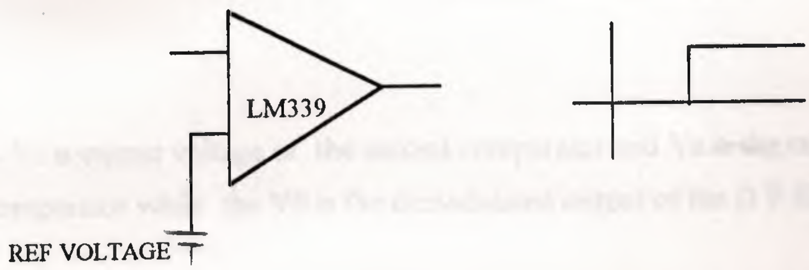
D P S K DEMODULATOR.

r transmitting the D P S K modulated signal the next task at the reciever is to odulate the received signal and this this proceses is little bit complicated then that nodulation of the signal and required more complicated circuitry then DPSK lulator such as ,an amplitude limmiter ,low pass filter ,comparator and circuit ıbitor .The block diagram of D P S K demodulator is shown below .



'he received DPSK modulated signal at the reciver passes to two branches of the odulator .In the first branch the signal goes into the amplitude limiter or comperator 'here the the comperator change the input signal into the digital signal by rejecting ıe negative part of the message signal .
ı the second branch the message signal passes to another comperator via low pass lter where the low pass filter stable the signal for certain value of amplitude .

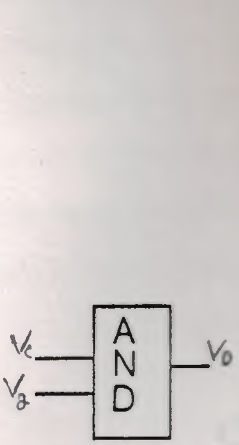
A comparator circuit is one to which a linear input voltage is compeard to another reference voltage ,the output being a digital condition representing the input voltage exceeded the reference voltage or in other we can say that a comparator accepts the liner voltage from the input and provides a digital output when one input is less than or greater than the second .The output values of comparator can be change to the require maximum and minimum level of signal according to the need of operation and this can be achieved by choosing the suitable value of comparators parameters ,such as the input resisteance ,and reference voltage .The figure of LM339 comparator that provied maximum "5"volt and minimum "0 "as an output and it is also a suitable choice for our purpose .



The value of reference voltage can be calculated by using the formula shown below .

$$Rf = R2 / R1+R2 * Vcc$$

Here Vcc is the input inverting terminal voltage of comparator.Now we have two output
signal one from the amplitude limiter while the other from coparator now the both signal procceds to the circuit inhbitor .This circuit consists of an inverter connected to the second output of comparator and one AND gate connected to the outputs of the both comperator . Finally the output of circuit inhbitor again inverted . The truth table of this operation is shown below .



Vc	Va	Vo
0	0	0
0	1	1
1	0	0
1	1	0

Here the Vc is output voltage of the second comparator and Va is the output voltage of first comperator while the V0 is the demodulated output of the D P S K signal.

(END OF CHAPTER NO 2)

CHAPTER # 3

(3) ERROR DETECTION AND CORRECTION SYSTEM USING HAMMING CODES

Transmission error in digital communication depends on the signal -to- noise ratio and this called bit error .If a particular system has a fixed value of the S/N and then the error rate is unacceptable high then some other means of improving reliability must be sought . Error control coding often provides the best solution..

Error control coding involves systematic addition of extra digits to the transmitted message.These extra check digits convey no information's by themselves ,but they make it possible to detect or correct errors in the regenerated message digits .In this chapter we will observe the simplest method for error -control coding and correction that is called Hamming codes but before starting our main fram of work we will make some concepts related to the our goal.

(3.1) PARITY.

Parity is the simplest and oldest form method of error detection although it is not very effective in data transmission ,it is still widely used due to its simplicity .A parity is a single bit added to a group of bits representing the sign of number of 1-bits set in the character .Parity can be odd or even .If the number of 1's in the data stream is even then the parity is "0" while if the number of 1's are odd then the parity is set to "1".The calculation of parity bit is based on the sum of data bits by module '2' as shown in the example given below .

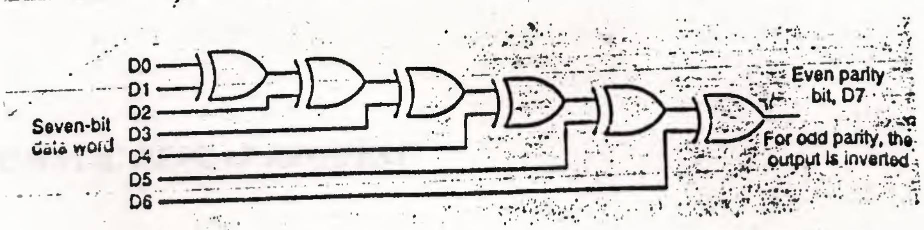
1 0 1 1 1 0

# of 1's = 4

sum by modular '2' = 1 + 1 + 1 + 1 = 0

parity = 0

The logic modulator of parity is shown below .



the data at the transmitter has even parity and if it is changed to odd parity during the ansmission so that at the receiving end wrong parity will reached this will indicate the ror in received data .

.2) FORWARD ERROR CORRECTION(F.E.C).

E.C is one of the basic technique used by the communication system to ensure the liable transmission of the data .F.E.C system is based on simplex communication system in application where it is impractical or impossible to request the retransmission of the orrupted data block ,an example might be the telemetry signals transmitted to an earth ation from the satellite on a deep space mission.A garbled message e can take several inutes or even hours to travel the distance between the two stations .

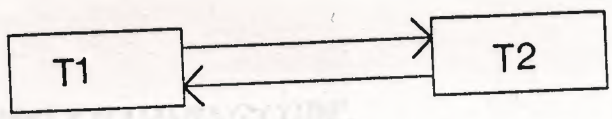
SIMPLEX

.3) AUTOMATIC REPEAT REQUEST

This the more popular error correction techniques .When a data block is received without error

a positive acknowledgement is sent back to the transmitter via the reverse channel . The pictorial

representation of ARQ is shown below



FULL DUPLEX

So this technique is used in two way system, one for receiving the data and one for sending the data.

(3.4) *HAMMING CODE.*

In F.E.C ,a return path is not used for requesting the retransmission of a message block in error, hence the name forward error correction .Several codes have been developed to suit application requiring F.E.C . Those must commonly recognised have been based on the research of mathemation Richard W.Hamming .These codes are referred as Hamming codes . Hamming codes employ the use of redundant bits that are inserted into the massage stream for error correction. The positions of these bits are established and known by the transmitter and receiver beforehand. If the receiver detects an error in the message block, the Hamming bits are used to identify the position of the error. This position, known as the syndrome, is the underlying principle of the Hamming code.

(3.5) *DEVELOPING A HAMMING CODE.*

We will now develop a Hamming code for single- bit FEC. For simplicity, 10 data bits will be used. The number of Hamming bits depends on the number of data bits m0, m1....mn,transmitted in the message stream, including the Hamming bits. If n is equal to the total number of bits transmitted in a message stream and m is equal to the number of Hamming bits , then m is the smallest number governed by the equation

$$2^{m} >= n+1$$

for a message of 10 data bits m is equal to 4 and is equal to 14 bits (10+4):

$$2 >= (10+4)+1$$

If the syndrome is to indicate the position of the bit error , check bits , or Hamming bits. $C_0 C_1 C_2$ & C_3 serving as parity, can be inserted into the message stream to perform a parity check based on the binary on the binary representation of each bit position . How is this possible? Note in Table that binary representation of each bit position forms an lternating bit pattern in the vertical direction . Each column proceeding from the LSB to he MSB alternates at one-half the rate of the

Bit position in message	Binary representation	Check bit	Position set
1	0001	C0	1,3,5,7,9,11,13
2	0010	C1	2,3,6,7,10,11,14,
3	0011	C2	4,5,6,7,12,13,14
4	0100	C3	8,9,10,11,12,13,14
5	0101		
6	0110		
7	0111		
8	1000		
9	1001		
10	1010		
11	1011		
12	1100		
13	1101		
14	1110		

vious column . The LSB alternates with every position .The next bit alternates every o bit positions, and so forth.

o illustrate how the check bits are encoded, the 10- bit message 1101001110 is labelled through m0, as illustrated in Figure 12-13 By inserting the check bits into the massage eam as shown, the total message length n is extended to 14 bits . For simplicity ,bit sitions 1,2 ,4 and 8 will be used for the check bits. Even or odd parity

neration can be performed on the bit positions associated with each check bit . We will e even parity here. As discussed earlier, even parity can be performed by exclusive-ing individual bits in a group of bits . For even parity, PEO through PE3 can serve as eighted parity check over the bit positions listed in Table 12-2 Exclusive- Oring these bit sitions together with the data corresponding to the 14 bit message stream shown in gure 12-13, we have the following

13 11 9 7 5 3 1 --- bit positions

$$PE0 = 0 = m8 + m6 + m4 + m3 + m1 + m0 + C0$$

14 11 10 7 6 3 2 ---bit positions

$$PE1 = 0 = m9 + m6 + m5 + m3 + m2 + m0 + C1$$

14 13 12 7 6 5 4 ---bit positions

$$PE2 = 0 = m9 + m8 + m7 + m3 + m2 + m1 + C2$$

14 13 12 11 10 9 8---bit positions

$$PE3 = 0 = m9 + m8 + m7 + m6 + m5 + m4 + C3$$

determine the value of the check bits C0 through C3 ,the equation above can be arranged as follows:

m9	m8	m7	m6	m5	m4	m3	m2	m1	m0
1	1	0	1	0	0	1	1	1	0

m9	m8	m7	m6	m5	m4	C3	m3	m2	m1	C3	m0	C1	C0
1	1	0	1	0	0	1	1	1	1	1	0	0	0

C0 = m8 + m6 + m4 + m3 + m1 + m0

= 1 + 1 + 0 + 1 + 1 + 0 = 0

C1 = m9 + m6 + m5 + m3 + m2 + m0

= 1 + 1 + 0 + 1 + 1 + 0 = 0

C2 = m9 + m8 + m7 + m3 + m2 + m1 (even parity)

= 1 + 1 + 0 + 1 + 1 + 1 = 1

C3 = m9 + m8 + m7 + m6 + m5 + m4

= 1 + 1 + 0 + 1 + 0 + 0 = 1

he check bits inserted into the message stream in positions 8,4,2 and 1are

C3 = 1

C2 = 1

C1 = 0

C0 = 0

s know look at how a bit error can be identified and corrected by the weighted parity s .Suppose that an error has been detected in the transmitted message stream .Bit on 7 has been lost in the transmutation .

11010011111000 --- transmitted bit stream

(lost in transmutations)

11010010111000--- received bit stream

(error in the bit position 7)

eceiver performs an even parity check over the same bit position as discussed above parity should result for each parity check if there are no errors .Since a bit error has red ,how ever the syndrome (location of the error) will be identified by binary er produced by the parity checks ,PE0 through PE3 ,as follows :

14	13	12	11	10	9	8	7	6	5	4	3	2	1	---bit position
1	1	0	1	0	0	1	0	1	1	1	0	0	0	--- received bit

n

C heck 0 : 13 11 9 7 5 3 1 -- bit position

PE0 = 1 + 1 + 0 + 0 + 1 + 0 + 0 =1 (even parity failure)1

Check1 : 14 1 1 10 7 6 3 2 --- bit postion

PE1 = 1 + 1 + 0 + 0 + 1 + 0 + 0 =1 (

ome)

Check2: 14 13 12 7 6 5 2 --- bit postion (=0111)

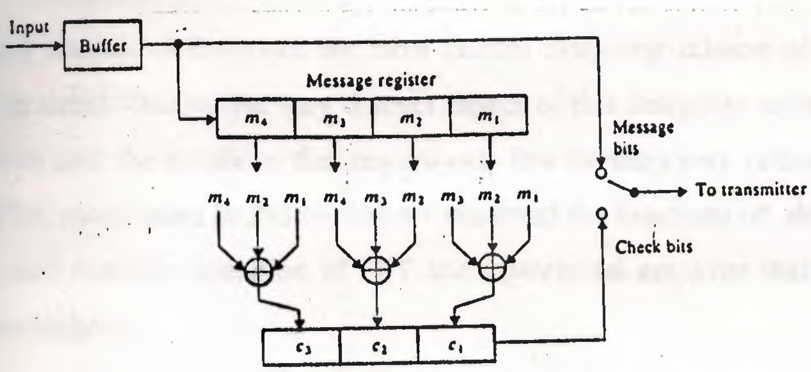
PE2 = 1 + 1 + 0 + 0 + 1 + 0 + 0 =1 (=7)

Check 3 : 14 13 12 11 10 9 8 ---bit position

PE3 = 1 + 1 + 0 + 1 + 0 + 0 + 1 = 0 (correct)

esulting syndrome is 0111, or bit postion 7 simply inverted and the four parity checks esult in 0000 (correct). The check bits are removed from positions 1,2,4 and 8, by resulting in the original message . One nice feature of this Hamming code is that the message is encoded there is no difference between the check bits and the original ge bits, that is the syndrome can just as well identify a check bit in error.

ENCODER FOR HAMMING CODES



END

CHAPTER # 4

JMMARY OF THE TOPIC

the previous chapters we have considered procedure , advantages and disadvantages the different digital modulation techniques such as ASK, FSK , PSK and DPSK .In SK modulation technique the amplitude of the carrier signal is changed with respect the binary '1' and '0' and in the frequency shift keying(FSK) the frequency of the rrier signal is changed with respect to binary '1' and '0' while in PSK the phase of carrier signal be changed according to the binary change in the message signal . we ve observed an other version of PSK ,i.e. is called DPSK in which the phase remain changed for the binary '1' and changed by 180 (In 2 M-array system) degree.

en we selected the Differentially phase shifted version of the digital modulation hniques as a most comprehensive modulation technique among the others due to its h rate of data transmission speed , low error probability and simple designing of dulator even the demodulation process is rather complicated and sensitive and this also the main disadvantage of this sechme.These aspect of the DPSK make it most quently used in digital data transmission system then other modulations techniques.

the second section of the topic we have discuss designing scheme of the DPSK dulator in detail. One of the very distinct aspect of this designing scheme is mind de approach and the simplicity that require only few circuitry part rather then other ailable DPSK modulators.In this section we observed the functions of shift register , ltiplexer, and combine operation of FET and operational amplifier that is basically an F.M modulator .

n section two we also have considered the demodulation technique which is much omplicated then modulation because the detection of the transmitted signal at the eception of the DPSK demodulator is very difficult due to the frequently change of hase in the received signal . The demodulation scheme requires low pass filter , mplitude limiter or comparator and one simple logic circuit called circuit inhabited at contains two input and gate with one inverted input.

ecause it is very important that the transmitted data must not be corrupted by the oise effect during the transmission and if it is then it is essential to detect the corrupted it and then make it correct. For this purpose we observed the complete procedure for rror detection and correction by means of Hamming Codes .The reasons for hoosing this technique is that it is very simple method for error detection and orrection . Although by using Hamming codes we can not detect or correct the nultiple or complex error in data transmission this technique is most frequently using n the digital data transmission system for error detection and correction. In this section e have analysed through one example the procedure to develop the Hamming codes

(END)

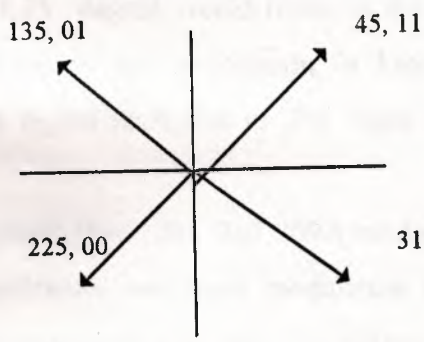
APPENDENDIX A

ELL 201B/C MODEM:

ıe Bell 201 family of modems is desigined to operate at a fixed data transfer rate of 2400 s over the basic ubnconditioned ,3002-type line or two -or- four wire private line. The ell 201A

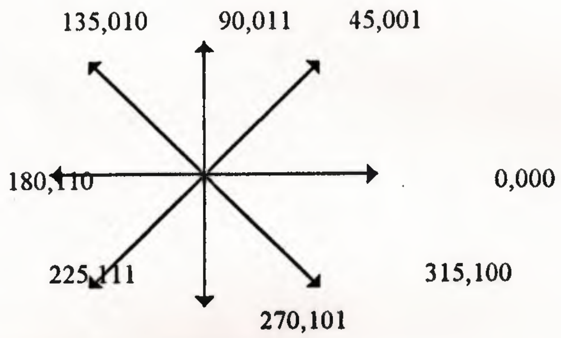
; an obsolelte 2000 bps modem , the bell 201B is for private or leased line application nd the Bell 201C is for switched or leased line application . Each modem is desinged for alf-duplex operation over the switched line or full -duplex operation over the four -wire rivate line .

our-phase DPSK is the modulation technique empolyed to achive 2400 bps .The phasor agram for the Bell 201B/c is shown in figure below.



BELL 208A AND 208B MODEMS:

Bell 208A and 208B modems are desigined for synchronous transmission and reception of the data at 4800 bps over four -wire private leased lines and switched lines , respectively . Eight -phase DPSK is employed on a 1800 -hz carrier frequency .Each consecutive three bits ,called tribits ,of the binary serial input data are encoded into a single phase change of the carrier frequncy .The encoding of three bits into tribit allows the representation eight e phase change of the carrier frequency . A transmission rate of three times the ate of 1600 is achived (4800= 1600*3). The phasor diagram for the Bell 208A n is shown below.



209A MODEM

ransfer rates can be increased further by encoding additional bits into a greater r of phase change .Four bits ,or a quadbit ,for example ,can be encoded into 16 e changes (M-ary = 4) .The phase differential betwwen adjacent phasors would t to 22.5 degrees (360/16 = 22.5 degrees). the problem here ,however is that any jitter in excess of 11.25 degree would result in the detection of errorneous data amount of phase jitter is not uncommon in long -haul network that utilize rative repeaters and digital multiplexers .For thgis resons , 16 -phase phase is lly not used .

oid the problem of phase jitter , the Bell 209A modems employs a combination of and PSK called quadrature amplitued modulation (QAM). QAM ,pronounced m" is a modultuon technique that uses 12 different phase and three different udes to represent 16 possible carrier states .Susceptility to phase jitter is effectively d to reduced by increasing

peration between adjacent phasor .By employing QAM 9600 -bps full duplex onous communication is achived using baud rate of 2400(9600=4*2400) over the four wire lines with D1 condition .The bell 209A also has provision for lexing multiples of 2400 bps into 9600 bps .

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