

DEPARTMENT OF ECE

WEST GODAVARI INSTITUTE OF SCIENCE & ENGINEERING

PRAKASARAOPALEM, AVAPADU, TADEPALLIGUDEM, W.G.DIST, A.P., INDIA



DIGITAL COMMUNICATION

LAB MANUAL

Prepared

by

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Assistant professor

(R-20)

III-B.Tech :I-Sem

ELECTRONICS & COMMUNICATION ENGINEERING

WEST GODAVARI INSTITUTE OF SCIENCE & ENGINEERING

Approved by AICTE & Affiliated to JNTU Kakinada.

Prakasaraopalem, Avapadu, Tadepalligudem, W.G.Dist, A.P., INDIA



Department:ECE

**WEST GODAVARI INSTITUTE OF SCIENCE & ENGINEERING
Prakasaraopalem, Avapadu, Tadepalligudem, W.G.Dist, A.P., INDIA**

INSTITUTE VISION AND MISSION

Vision of the Institution:

Promote academic excellence, research, innovation, and entrepreneurial skills to produce graduates with human values and leadership qualities to serve the nation

Mission of the Institution:

Provide student-centric education and training on cutting-edge technologies to make the students globally competitive and socially responsible citizens. Create an environment to strengthen the research, innovation and entrepreneurship to solve societal problems.

DIGITAL COMMUNICATIONS LAB
SYLLABUS

1. Time division multiplexing.
2. Pulse code modulation.
3. Differential pulse code modulation.
4. Delta modulation.
5. Frequency shift keying.
6. Phase shift keying.
7. Differential phase shift keying.
8. Companding
9. Source Encoder and Decoder
10. Linear Block Code-Encoder and Decoder
11. Binary Cyclic Code - Encoder and Decoder
12. Convolution Code - Encoder and Decoder

Equipment required for Laboratories:

1. RPS - 0 – 30 V
2. CRO - 0 – 20 M Hz.
3. Function Generators - 0 – 1 M Hz
4. RF Generators - 0 – 1000 M Hz /0 – 100 M Hz.
5. Multimeters
6. Lab Experimental kits for Digital Communication
7. Components

Radio Receiver/TV Receiver Demo kits or Trainees

Do's

1. Be regular to the Lab.
2. Students should come prepare theory of the concern experiment to the lab.
3. Read carefully the power ratings of the equipment before it is switched ON.
4. Observe the proper power sockets to avoid mechanical damage.
5. After the completion of the experiments, SWITCH OFF the power supply and handover the apparatus.
6. Strictly follow the instructions given by the Lab Instructor.
7. Avoid unnecessary talking while doing the experiment.
8. Keep your work area clean.
9. Proper dress code has to be maintained while entering into the lab.
10. Students should maintain the observation book and Record up to date.
11. Students Should be confined to the concern work bench.
12. Students should follow the intend procedure to receive and deposit the equipment.
13. Maintain strict academic discipline in the laboratory.

Don'ts

1. Avoid loose connections and short circuits.
2. Do not exceed the specified ratings of the equipment.
3. Do not interchange the components while doing the experiment.
4. Do not throw the connecting wires on the floor.
5. Do not operate the trainer kits unnecessarily.
6. Don't switch on to the circuit until it gets verified by the concern Lab In-charge.
7. Never do experiments on your own without proper guidance.
8. Don't mishandle the Equipment.
9. Don't come late to the laboratory.
10. Do not operate equipment without reading the Instructions.
11. Don't connect or remove the components when the power is ON.

INDEX

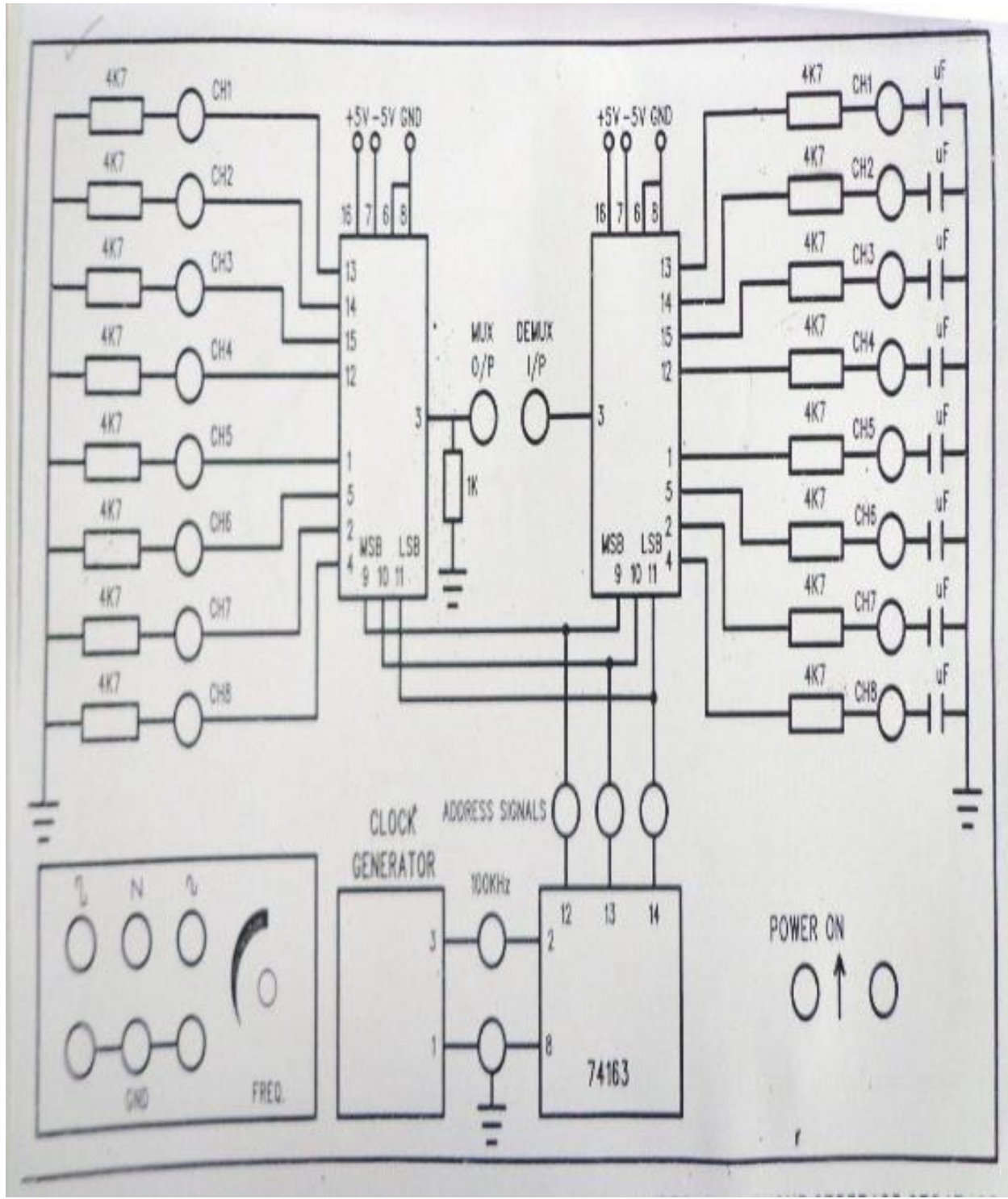
S.No	DATE	NAME OF THE EXPERIMENT	SIGN. OF STAFF	RE MARKS
1		Time division multiplexing.		
2		Pulse code modulation		
3		Differential pulse code modulation		
4		Delta modulation.		
5		Frequency shift keying.		
6		Phase shift keying.		
7		Differential phase shift keying.		
8		Companding		
9		Source Encoder and Decoder		
10		Linear Block Code-Encoder and Decoder		
11		Binary Cyclic Code-Encoder and Decoder		
12		Convolution Code-Encoder and Decoder		

No. of Experiments completed:

Average marks awarded for day to day work:

Signature of Staff Member/Date

BLOCK DIAGRAM OF TIME DIVISION MULTIPLEXING AND DEMULTIPLEXING



EXPT. NO:

DATE:

TIME DIVISION MULTIPLEXING AND DEMULTIPLEXING

AIM:

To transmit a multiplexed output of different message signals through a single channel using TDM system and recover back the original message signals through a demultiplexer at receiver end.

APPARATUS:

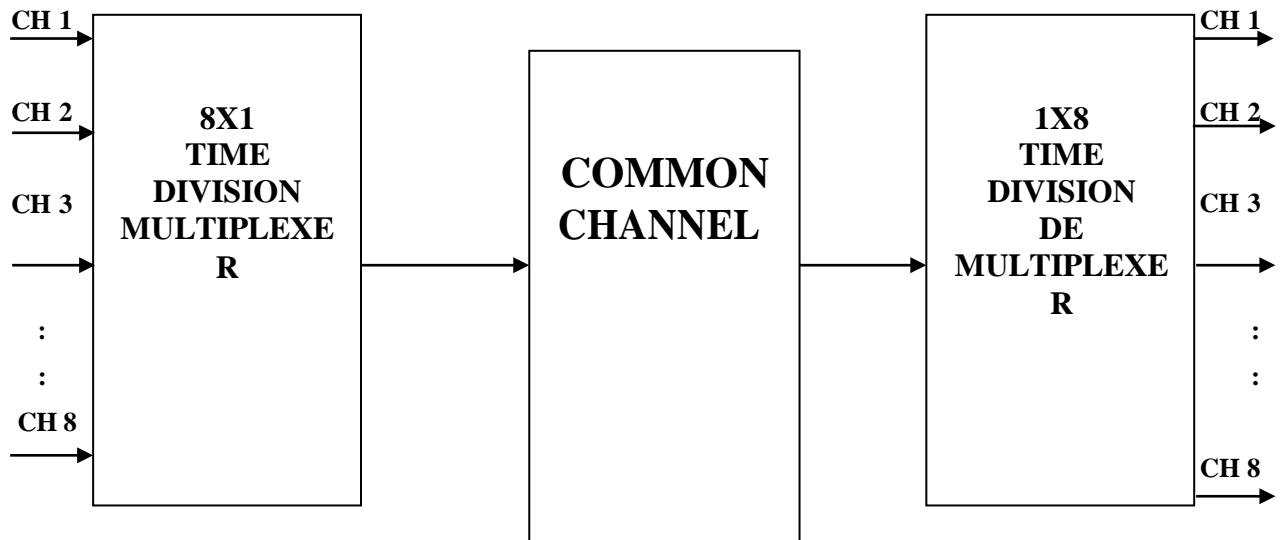
1. Time Division Multiplexing & Demultiplexing Trainer Kit
2. CRO
3. CRO Probes
4. Connecting wires (or) Patch cards

THEORY:

The Sampling Theorem provides the basis for transmitting the information contained in a band limited message signal $m(t)$ as a sequence of samples of $m(t)$ taken uniformly at a rate that is usually slightly higher than the Nyquist rate. An important feature of the sampling process is a conservation of time. That is, the transmission of the message samples engages the communication channel s for only a fraction of the sampling interval on a periodic basis, and in this way some of the time interval between adjacent samples is cleared for use by other independent message sources on a time shared basis. We thereby obtain a time division multiplexing (TDM) system, which enables the joint utilization of a common communication channel by a plurality of independent message sources without mutual interference among them.

The TDM system is highly sensitive to dispersion in the common channel, that is, to variations of amplitude with frequency or lack of proportionality of phase with frequency. Unlike FDM, TDM is immune to non-linearities in the channel as a source of cross talk. The reason for this is, the different message signals are not simultaneously applied to the channel. The primary advantage of TDM is that several channels of information can be transmitted simultaneously over a single cable.

BLOCK DIAGRAM:



OBSERAVATION TABLE:

AT TRANSMITTER SIDE (MUX):

S.NO	CHANNEL	TYPE OF WAVE	AMPLITUDE	TIME PERIOD	FREQUENCY
1					
2					
3					

AT RECEIVER SIDE (DEMUX):

S.NO	CHANNEL	TYPE OF WAVE	AMPLITUDE	TIME PERIOD	FREQUENCY
1					
2					
3					

In the circuit diagram the 555 timer is used as a clock generator. This timer is a highly stable device for generating accurate time delays. In this circuit this timer generates clock signal, which is of 100 KHz frequency (approximately). This clock signal is connected to the 74163 IC; it is synchronous presentable binary counter. It divides the clock signal frequency into three parts and those are used as selection lines for multiplexer and de-multiplexer. Inbuilt signal generator is provided with sine, square and triangle outputs with variable frequency. These three signals can be used as inputs to the multiplexer. IC 4051 is an 8 to 1 Analog Multiplexer. Again IC 4051 is wired as one to eight de-multiplexers. Demux input receives the data source and transmits the data signals on different channels.

PROCEDURE:

1. Switch on Time division multiplexing and demultiplexing trainer kit.
2. Connect the sine wave to channel 1, square wave to channel 2 and triangle wave form to Channel 3 terminals of n to 1 multiplexer.
3. Observe the Multiplexer output on channel 1 of a CRO.
4. Connect Mux output to Demux input.
5. Observe corresponding signal outputs at channel 2 of CRO.

PRECAUTIONS:

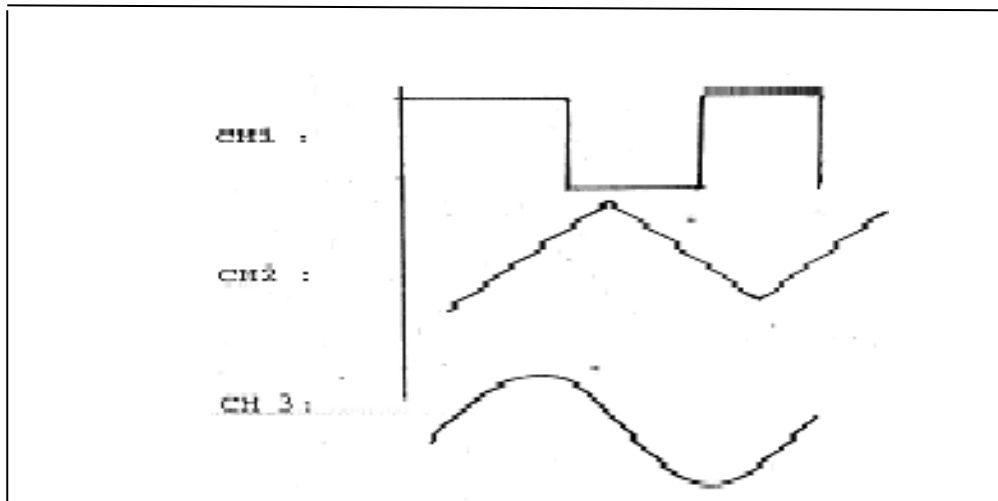
- 1 .Avoid loose Connections.
2. Switch off the power supply during connections
3. Waveforms must be noted carefully.

RESULT:

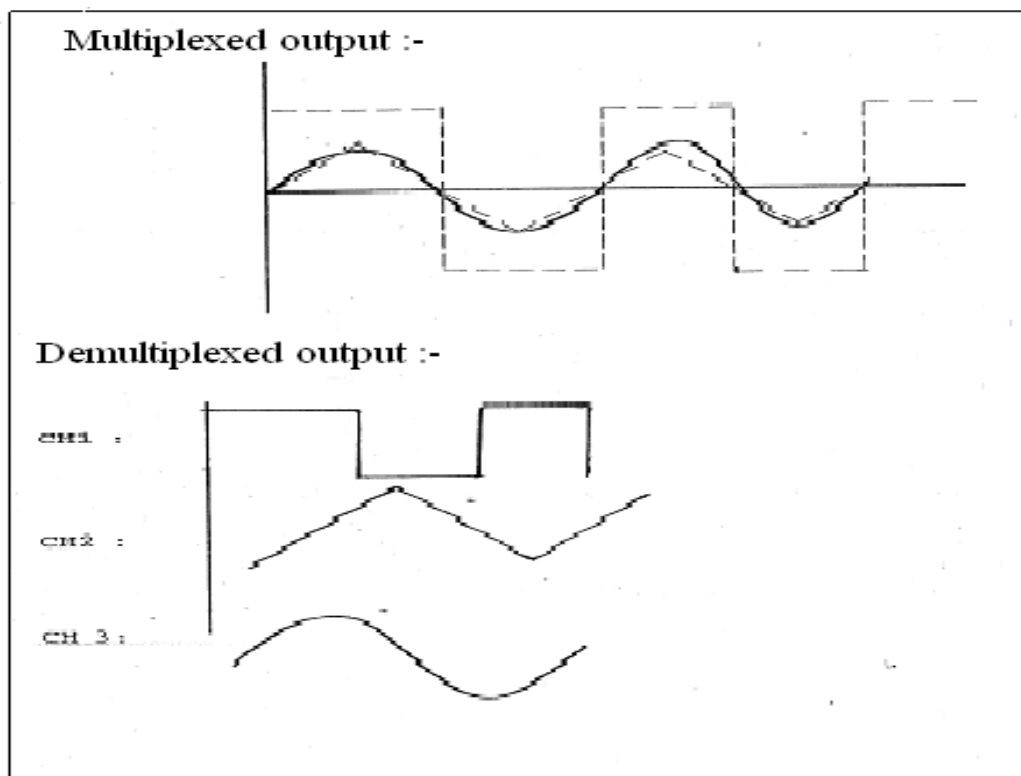
Three different message signals (square, sine & triangular waves) are transmitted at a time through a single communication channel using TDM system and hence the TDM waveform is observed and the amplitude and time periods of the three message signals are measured. The Demultiplexed message signals at the receiver are observed and hence the time period and amplitude of the waveforms are measured. From the measurements we can observe that the amplitude and frequency values are the same at the receiver also.

MODEL WAVEFORMS:

1. INPUT WAVEFORMS:-



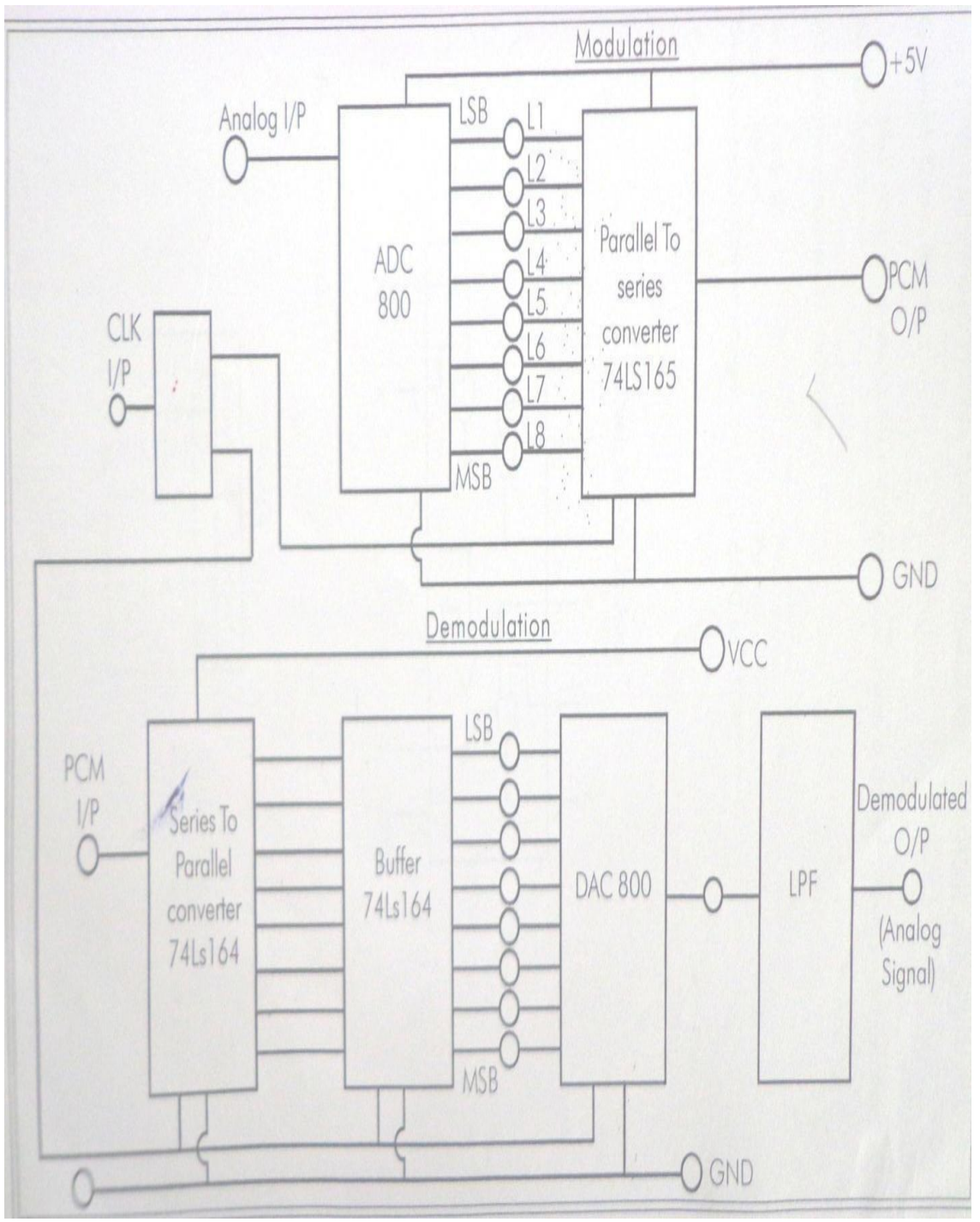
2. OUTPUT WAVEFORMS:-



VIVA QUESTIONS:

1. What is meant by multiplexing technique and what are the different types of Multiplexers?
2. Briefly explain about TDM&FDM?
3. What is the transmission band width of a PAM/TDM signal?
4. Define crosstalk effect in PAM/TDM system?
5. What are the advantages of TDM system?
6. What are major differences between TDM&FDM?
7. Give the value of T_s in TDM system?
8. What are the applications of TDM system?
9. What is meant by signal overlapping?
10. Which type of modulation technique will be used in TDM?

CIRCUIT DIAGRAM OF PULSE CODE MODULATION AND DEMODULATION



EXPT. NO:

DATE:

PULSE CODE MODULATION & DEMODULATION

AIM:

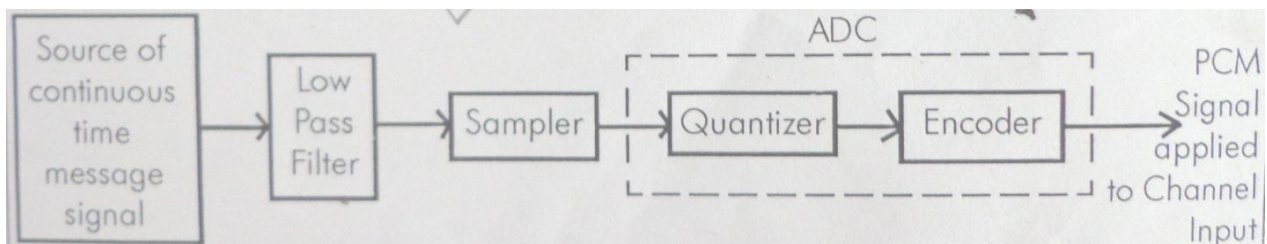
To convert an analog signal into a pulse digital signal using PCM system and to convert the digital signal into analog signal using PCM demodulation system.

APPARATUS:

1. Pulse Code Modulation & Demodulation trainer kit
2. CRO
3. CRO Probes
4. Connecting wires (or) Patch cards

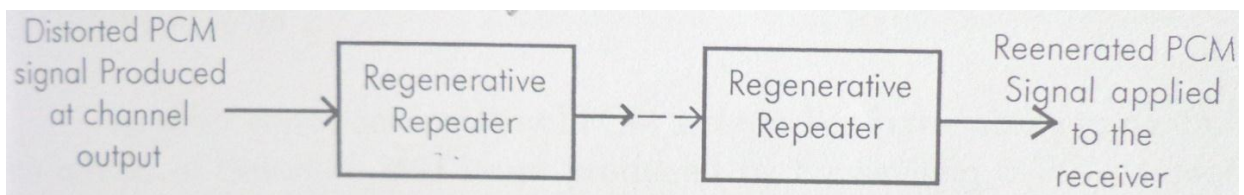
THEORY:

In pulse code modulation (PCM) a message signal is represented by a sequence of coded pulses, which is accomplished by representing the signal in discrete form in both time and amplitude. The basic elements of a PCM system are shown below.



(a)

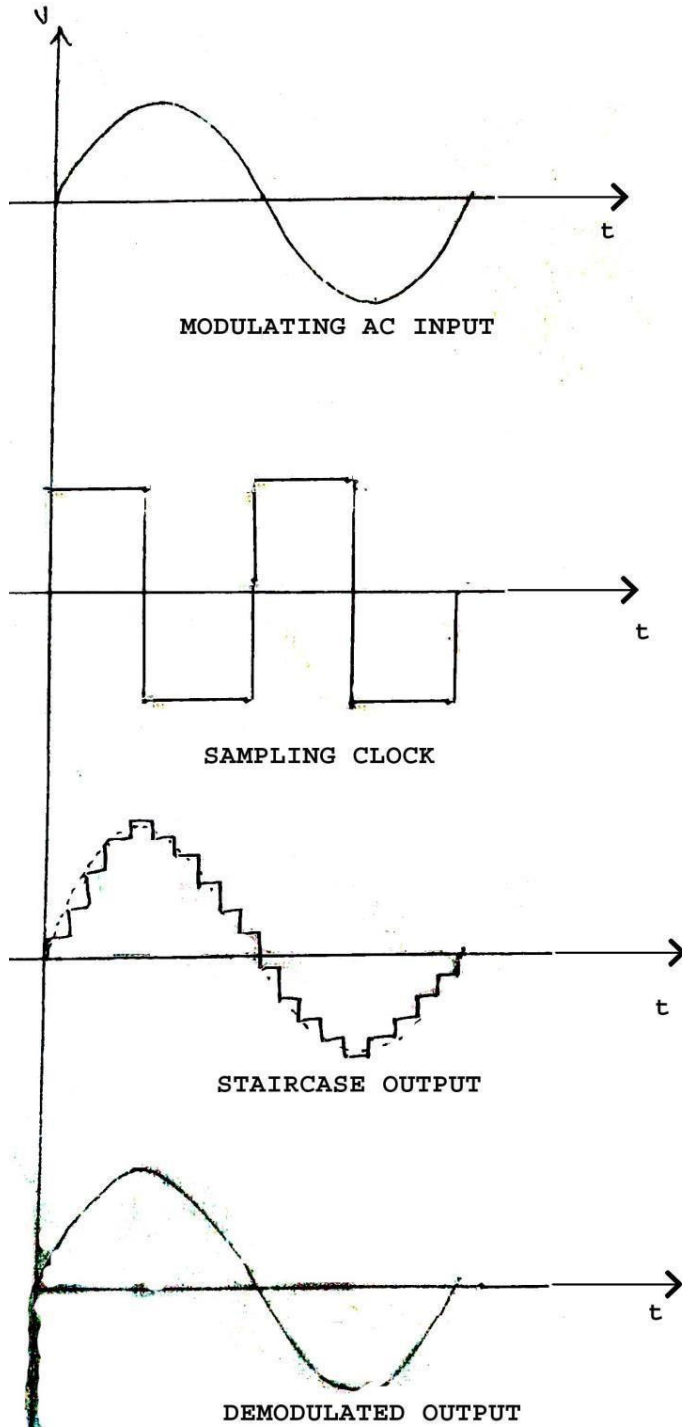
Transmitter (MODULATION)

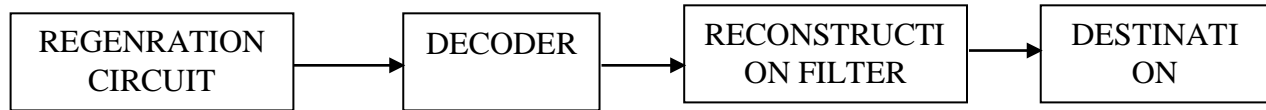


(b) Transmission Path (CHANNEL)

MODEL WAVEFORMS:

FOR SINE INPUT





(c) **Receiver (DEMODULATION)**

The basic operations performed in the transmitter of a PCM system are sampling, quantizing and encoding. The low pass filter prior to sampling is included to prevent aliasing of the message signal. The incoming message signal is sampled with a train of narrow rectangular pulses so as to closely approximate the instantaneous sampling process. To ensure perfect reconstruction of the message signal at the receiver, the sampling rate must be greater than twice the highest frequency component W of the message signal in accordance with the sampling theorem.

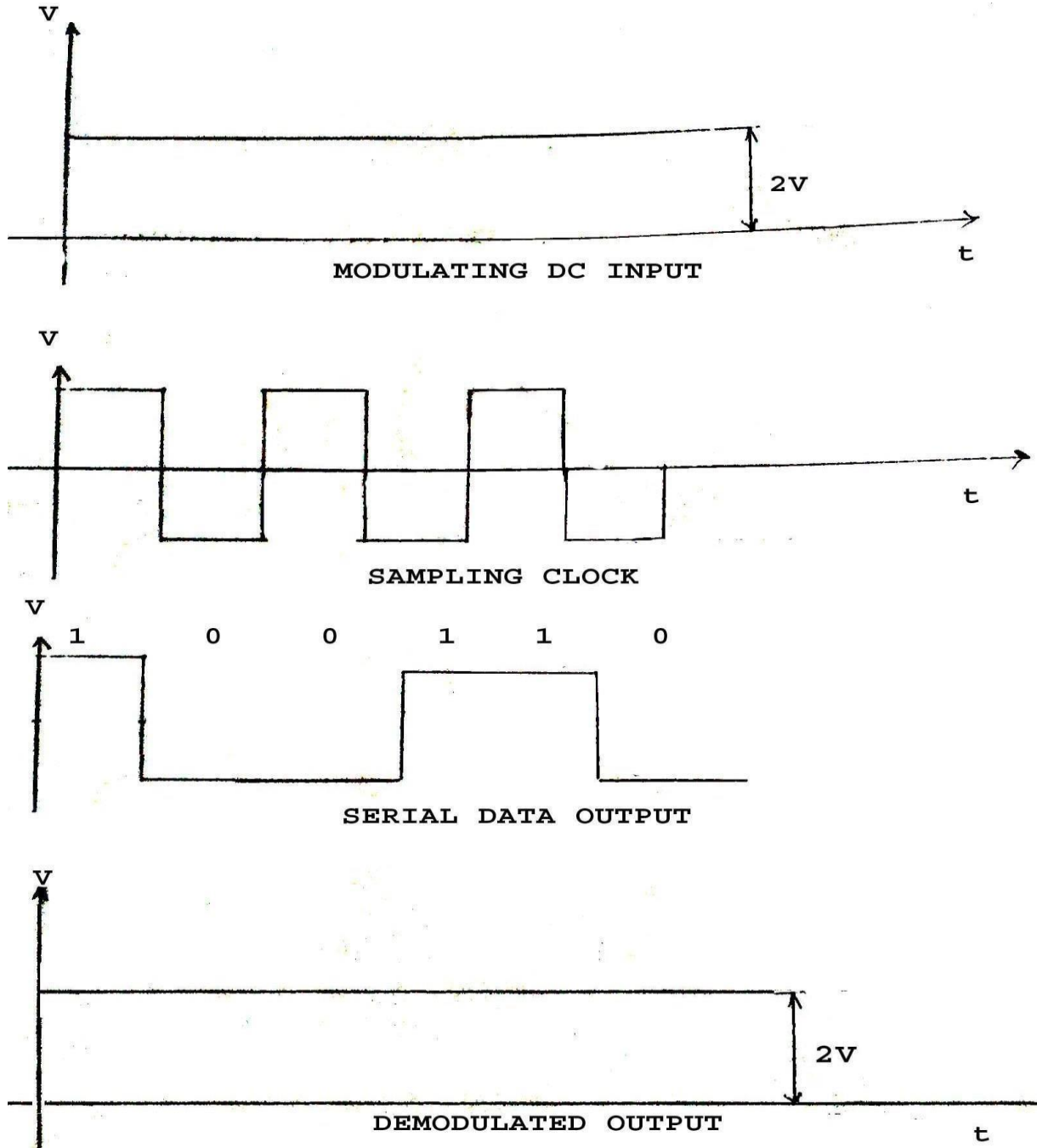
The quantizing and encoding operations are usually performed in the same circuit, which is called an analog-to-digital converter. The same circuit, which is called an analog-to-digital converter. The sampled version of the message signal is then quantized, thereby providing a new representation of the signal that is discrete in both time and amplitude. In combining the process of sampling and quantization, the specification of a continuous message (baseband) signal becomes limited to a discrete set of values, but not in the form best suited to transmission. To exploit the advantages of sampling and quantizing for the purpose of making the transmitted signal more robust to noise, interference and other channel impairments, we require the use of an encoding process to translate the discrete set of sample values to a more appropriate form of signal.

Regeneration:

The most important feature of PCM system lies in the ability to control the effects of distortion and noise produced by transmitting a PCM signal through a channel. This capability is accomplished by reconstructing the PCM signal by means of a chain of regenerative repeaters located at sufficiently closed spacing along the transmission route. As illustrated in figure below three basic functions are performed by a regenerative repeater: equalization timing and decision making.

MODEL WAVEFORMS:

FOR DC WAVE



Regeneration Repeater

The equalizer shapes the received pulses so as to compensate for the effects of amplitude and phase distortion produced by the non ideal transmission characteristics of the channel. The timing circuit provides a periodic pulse trainer, derived from the received pulses, for sampling the equalized pulses at the instants of time where the signal to noise ratio is maximum. Each sample so extracted is compared to predetermined threshold in the decision making device. In each bit interval, a decision is then made whether the symbol is 1 or 0 on the basis of whether the threshold is exceeded or not. If the threshold is exceeded, a pulse representing symbol '1' is transmitted. In the way, the accumulation of distortion and noise in a repeater span is completely removed.

The basic operations in the receiver are regeneration of impaired signals, decoding and reconstruction of the train of quantized samples. The first operation in the receiver is to regenerate (i.e., reshape and cleanup) the received pulses one last time. These clean pulses are then regrouped in to code words and decoded into a quantized PAM signal. The decoding process involves generating a pulse, the amplitude of which is the linear sum of all the pulses in the code word. The final operation in the receiver is to recover the message signal by passing the decoder output through a low pass reconstruction filter whose cut-off frequency is equal to the message band width W . Assuming that the transmission path is error free, the recovered signal includes no noise with the exception of the initial distortion introduced by the quantization process.

PROCEDURE:

1. Switch on Pulse code modulation and demodulation.
2. Connect the variable DC output to the Analog I/P of modulation section.
3. Connect the clock O/P of bit clock generator to the clock I/P of modulation Section.
4. By varying the variable DC O/P observe the PCM O/P on CRO.
5. Connect the AF output to Analog I/P of modulation section by removing variable DC O/P
6. Connect the PCM O/P to PCM I/P of demodulation section.
7. Observe the DAC O/P at channel 1 of CRO and observe the demodulated O/P at channel 2 of CRO.

OBSERVATIONS TABLE:**AC INPUT (SINE WAVE)**

Name of the signal	Amplitude	Time period	Frequency
Modulating (AF) Signal			
Carrier Signal(Pulse)			
PCM Signal Digital data			
Demodulated Signal			

DC INPUT

Name of the signal	Amplitude	Time period	Frequency
Modulating (DC) Signal			
Carrier Signal(Pulse)			
PCM Signal Digital data			
Demodulated Signal			

PRECAUTIONS:

- 1 .Avoid loose Connections.
2. Switch off the power supply during connections
3. Waveforms must be noted carefully.

RESULT:

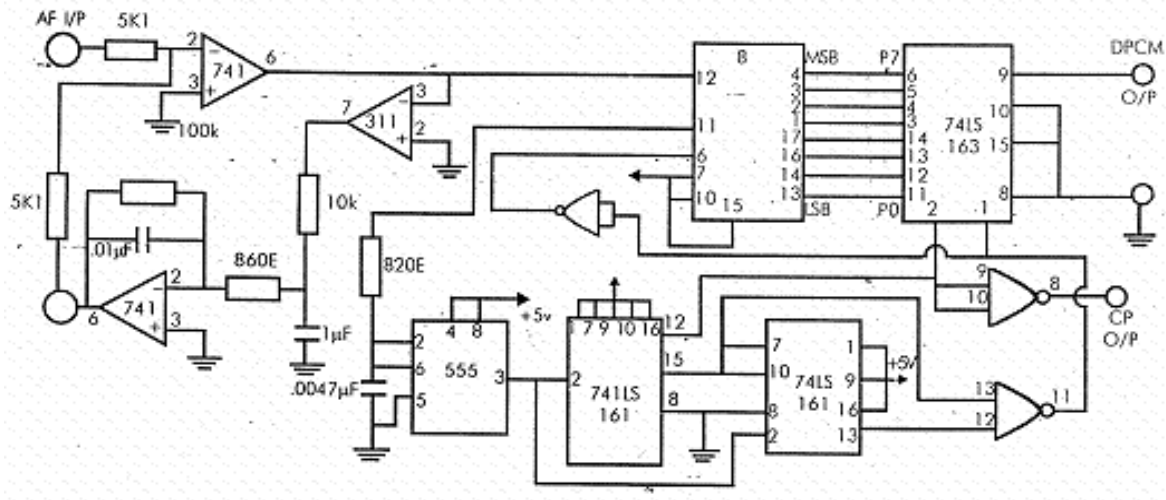
Hence the continuous time signal is converted into digital signal in transmitter side and in the receiver side the original continuous signal is recovered back from the received PCM signal are observed and plotted the appropriate waveforms

Thus PCM modulation and demodulation systems are act as A/D and D/A converters. From the measurements we can observe that the amplitude and frequency values are the same at the receiver also.

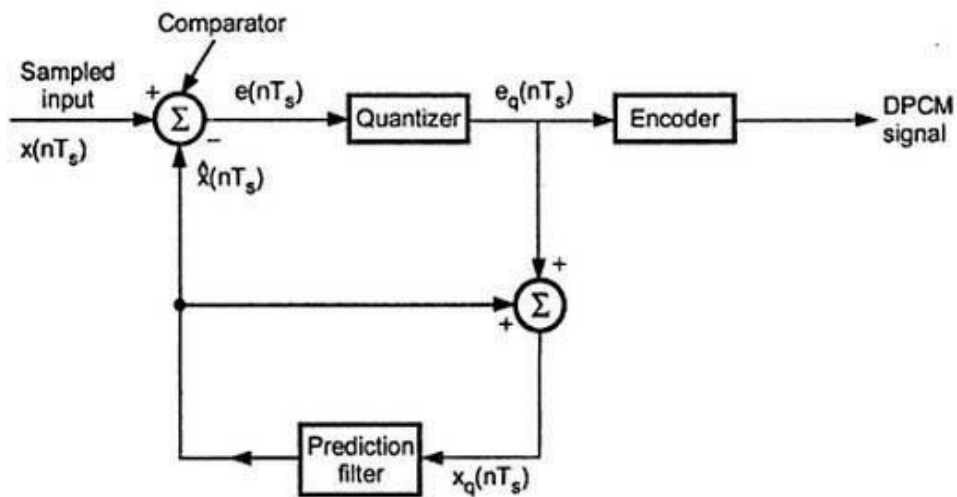
VIVA QUESTIONS:

1. What is the expression for transmission bandwidth in a PCM system?
2. What is the expression for quantization noise/error in PCM system?
3. What are the applications of PCM?
4. What are the advantages of the PCM?
5. What are the disadvantages of PCM?
6. What is the statement of sampling theorem?
7. What is meant by Quantizing?
8. What is meant by Encoding?
9. What is the purpose of Regenerative Repeater in PCM?
10. What is the purpose of Reconstruction filter in Demodulator?

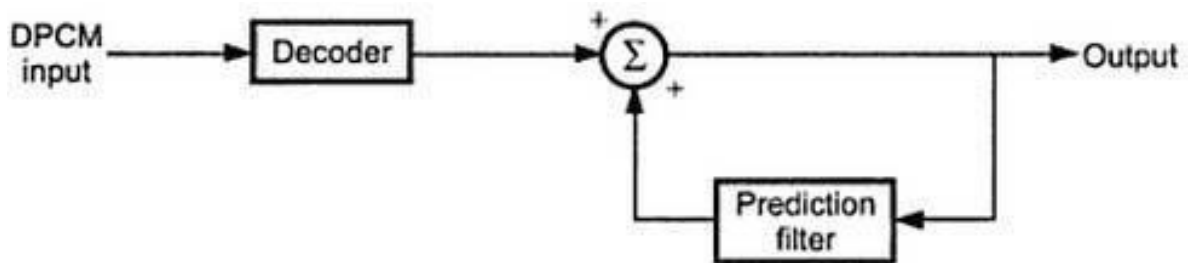
CIRCUIT DIAGRAM OF DPCM



BLOCK DIAGRAM OF DPCM MODULATION:



DEMODULATION:



EXPT. NO:

DATE:

DIFFERENTIAL PULSE CODE MODULATION & DEMODULATION

AIM:

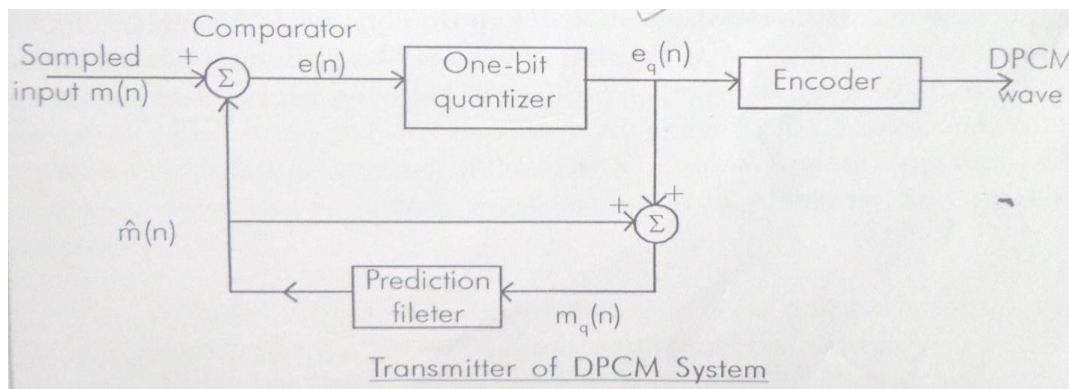
To study the differential PCM modulation and demodulation by sending variable frequency sine wave and variable DC signal input and plot the appropriate waveforms.

APPARATUS:

1. DPCM trainer kit
2. CRO
3. CRO Probes
4. Connecting wires (or) Patch cards

THEORY:

If we know the past behavior of a signal up to a certain point in time, we may use prediction to make an estimate of a future value of the signal. Suppose, a base band signal $m(t)$ is sampled at the rate $f_s = 1/T_s$ to produce the sequence $\{m(n)\}$ whose samples are T_s seconds apart. It is possible to predict future values of the signal $m(t)$, provides motivation for the differential quantization scheme shown in fig. below.

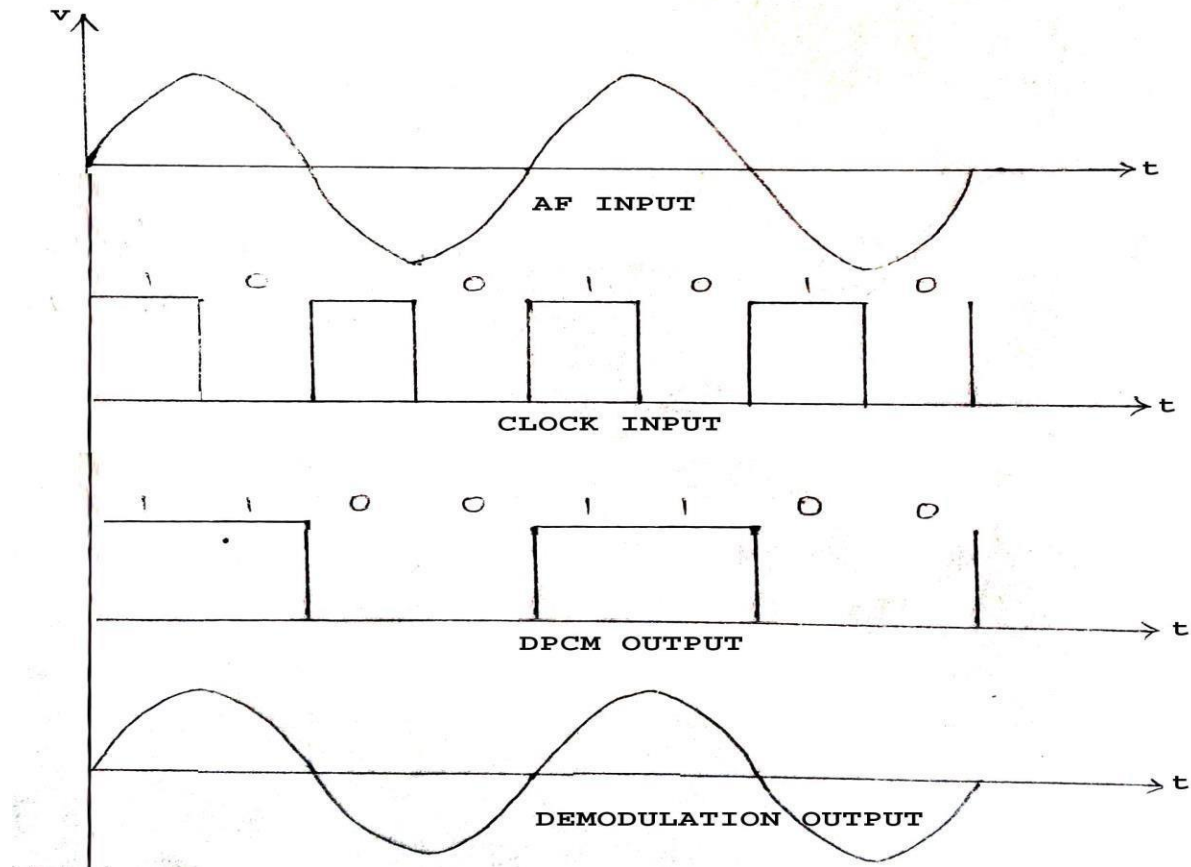


MODULATION BLOCK

In this scheme, the input signal to the quantizer is the difference between the unquantized input sample $m(n)$ and a prediction of it, denoted by $\hat{m}(n)$. This predicted value is produced by using a linear prediction filter whose input, consists of a quantized version of the input sample $m(n)$.

MODEL WAVEFORMS:

FOR SINE WAVE

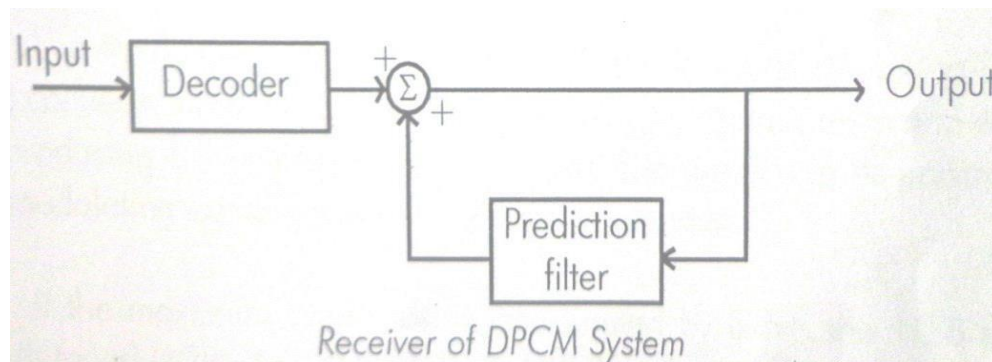


OBSERVATION TABLE:

Name of the signal	Amplitude	Time period	Frequency
Modulating (AF) Signal			
Carrier Signal(Pulse)			
DPCM Signal Digital data			
Demodulated Signal			

The difference signal $e(n)$ is the prediction error, since it is the amount by which the prediction filter fails to predict the input exactly. By encoding the quantizer output, we obtain a variant of PCM known as DPCM. Irrespective of the properties of the prediction filter, the quantized sample $m_q(n)$ at the prediction filter input differs from the original input sample $m(n)$ by the quantization error $q(n)$. Accordingly if the prediction is good, the variance of the prediction error $e(n)$ will be smaller than the variance of $m(n)$, so that a quantizer with a given number of levels can be adjusted to produce a quantization error with a smaller variance than would be possible if the input sample $m(n)$ were quantized directly as in a standard PCM system.

The receiver for reconstructing the quantized version of the input is shown in fig. below.



DEMODULATION BLOCK

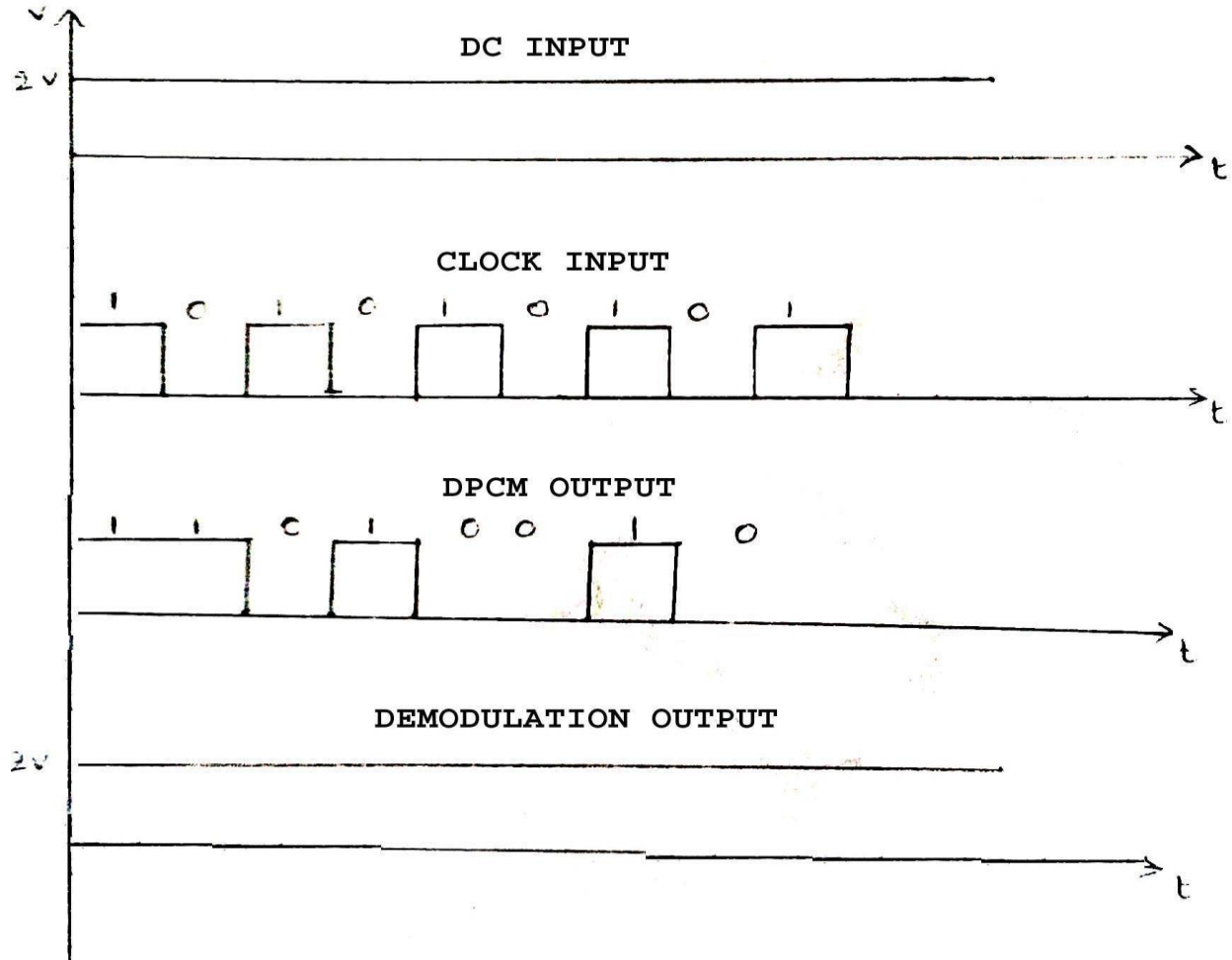
It consists of a decoder to reconstruct the quantized error signal. The quantized version of the original input is reconstructed from the decoder output using the same prediction filter used in transmitter. In the absence of channel noise, we find that the encoded signal at the receiver input is identical to the encoded signal at the transmitter output. Accordingly, the corresponding receiver output is equal to $m_q(n)$, which differs from the original input $m(n)$ only by the quantization error $q(n)$ incurred as a result of quantizing the prediction error $e(n)$.

From these analyses, we observe that, in a noise-free environment, the prediction filter in the transmitter and receiver operates on the same sequence of samples $m_q(n)$. It is with the purpose that a feedback path is added to the quantizer in the transmitter.

DPCM includes Delta Modulation as a special case. By comparing the DPCM system with DM system, they are basically similar, except for two important differences: The use of a one-bit (two-level) quantizer in the delta modulator and the replacement of the prediction filter by a single delay element (i.e., zero prediction order).

MODEL WAVEFORMS:

FOR DC INPUT



OBSERVATION TABLE:

Name of the signal	Amplitude	Time period	Frequency
Modulating (DC) Signal			
Carrier Signal(Pulse)			
DPCM Signal Digital data			
Demodulated Signal			

PROCEDURE:

1. Switch on Differential Pulse code modulation and demodulation trainer kit.
2. Connect the variable DC output to the Analog I/P of modulation section.
3. Measure the clock O/P present in the modulation Section.
4. By varying the variable DC O/P observe the DPCM O/P on CRO.
5. Connect the AF output to Analog I/P of modulation section by removing variable DC O/P
6. Connect the DPCM O/P to DPCM I/P of demodulation section.
7. Observe the DAC O/P at channel 1 of CRO and observe the demodulated O/P at channel 2 of CRO.

PRECAUTIONS:

- 1 .Avoid loose Connections.
2. Switch off the power supply during connections
3. Waveforms must be noted carefully.

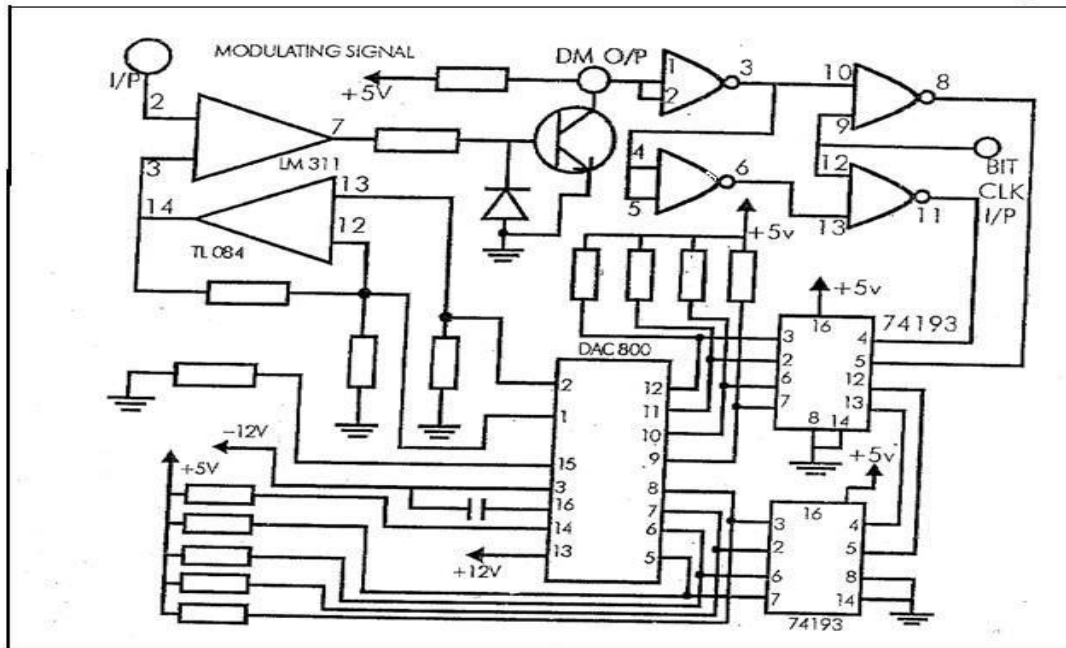
RESULT:

Hence differential PCM modulation and demodulation are observed by sending variable frequency sine wave and variable DC signal input and plotted the appropriate waveforms. From the measurements we can observe that the amplitude and frequency values are the same at the receiver also.

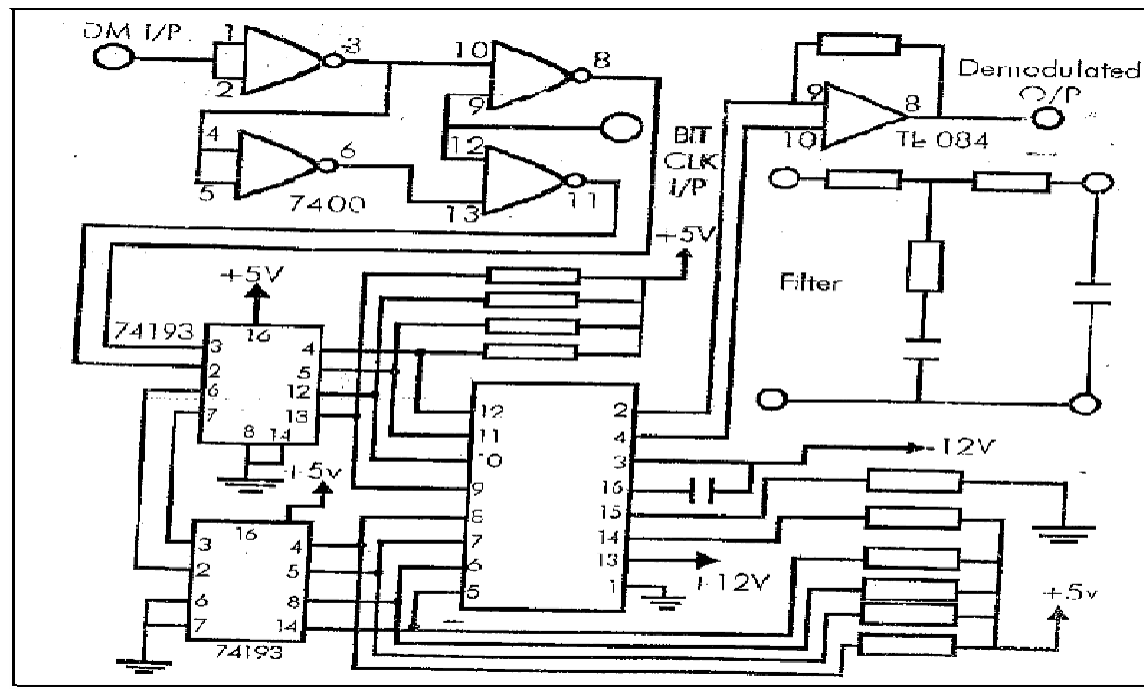
VIVA QUESTIONS

- 1 .What is the differences between PCM and DPCM?
- 2 .What is the bandwidth of DPCM?
- 3 .What is the use prediction filter in DPCM?
4. What are the applications of PCM?
5. What are the advantages of the DPCM?
6. What are the disadvantages of DPCM?

CIRCUIT DIAGRAM OF DELTA MODULATOR:



CIRCUIT DIAGRAM OF DELTA DE-MODULATOR:



EXPT. NO:

DATE:

DELTA MODULATION AND DEMODULATION

AIM:

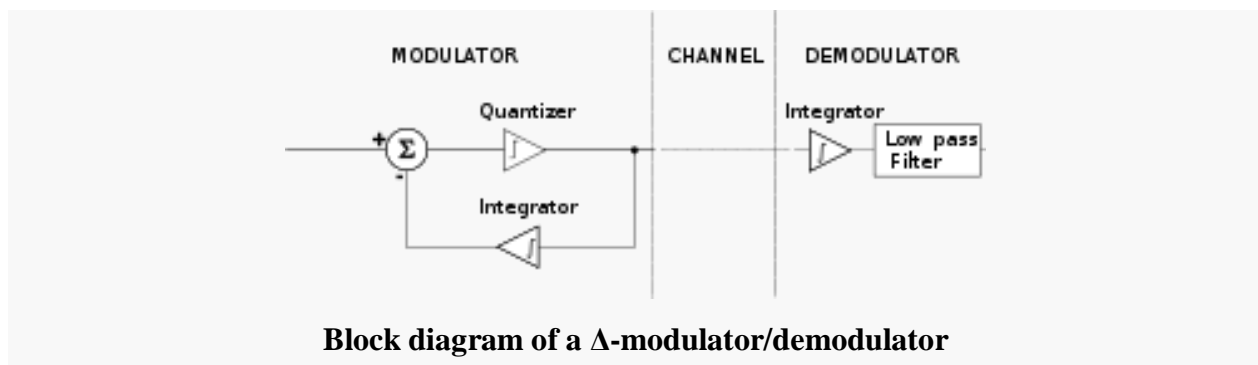
To transmit an analog message signal in its digital form and again reconstruct back the Original analog message signal at receiver by using Delta modulator.

APPARATUS:

1. Delta modulation and demodulation trainer kit
2. CRO
3. CRO Probes
4. Connecting wires (or) Patch cards

THEORY:

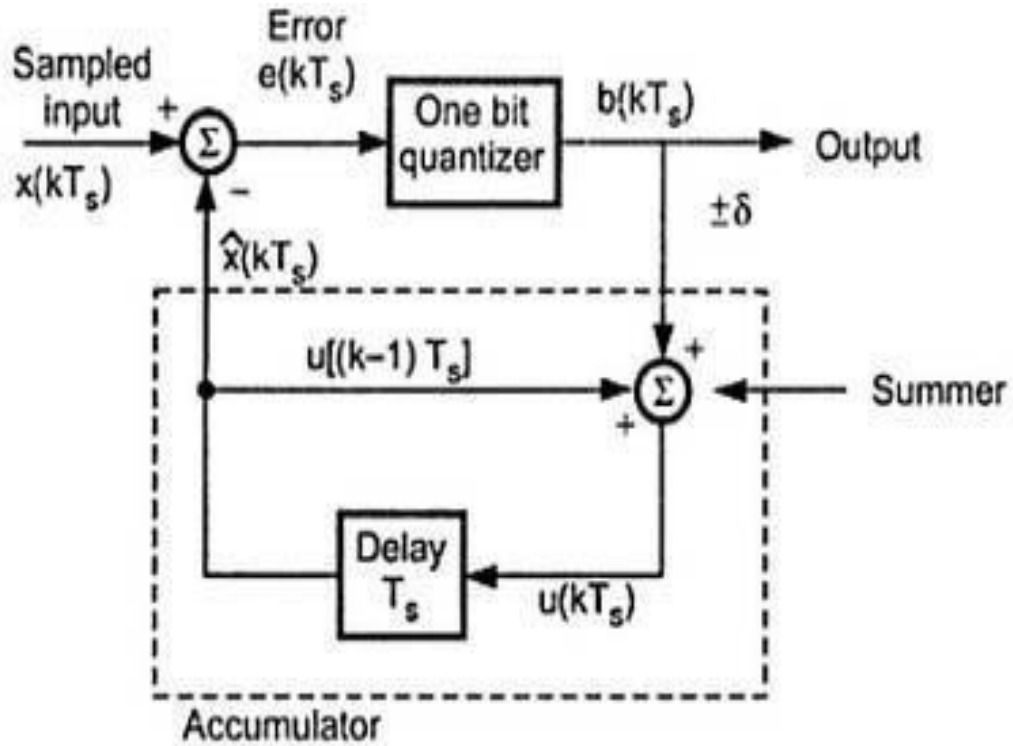
Delta modulator is an advanced version of PCM system, so it is also known as 'Single bit PCM system'. It generates the output signal by comparing the input signal with its quantized approximated output i.e. if the step size increases to $+\Delta$ it gives binary value '1' and if step downs to $-\Delta$ it gives binary value '0'. In this way it reduces the transmission channel band width. Rather than quantizing the absolute value of the input analog waveform, delta modulation quantizes the difference between the current and the previous step, as shown in the block diagram



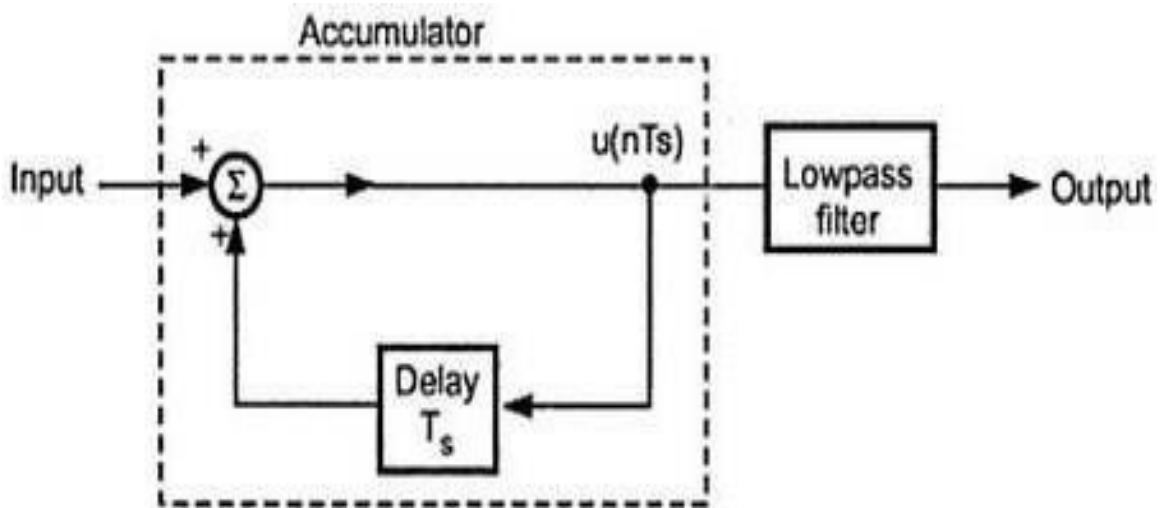
Block diagram of a Δ -modulator/demodulator

The modulator is made by a quantizer which converts the difference between the input signal and the average of the previous steps. In its simplest form, the quantizer can be realized with a comparator referenced to 0 (two levels quantizer), whose output is 1 or 0 if the input signal is

BLOCK DIAGRAM OF DELTA MODULATION:



BLOCK DIAGRAM OF DELTA DEMODULATION:



Positive or negative. It is also a bit-quantizer as it quantizes only a bit at a time. The demodulator is simply an integrator (like the one in the feedback loop) whose output rises or falls with each 1 or 0 received. The integrator itself constitutes a low-pass filter.

In delta modulation, the transmitted data is reduced to a 1-bit data stream. Its main features are:

- The analog signal is approximated with a series of segments
- Each segment of the approximated signal is compared to the original analog wave to determine the increase or decrease in relative amplitude
- The decision process for establishing the state of successive bits is determined by this comparison
- Only the change of information is sent, that is, only an increase or decrease of the signal amplitude from the previous sample is sent whereas a no-change condition causes the modulated signal to remain at the same 0 or 1 state of the previous sample.

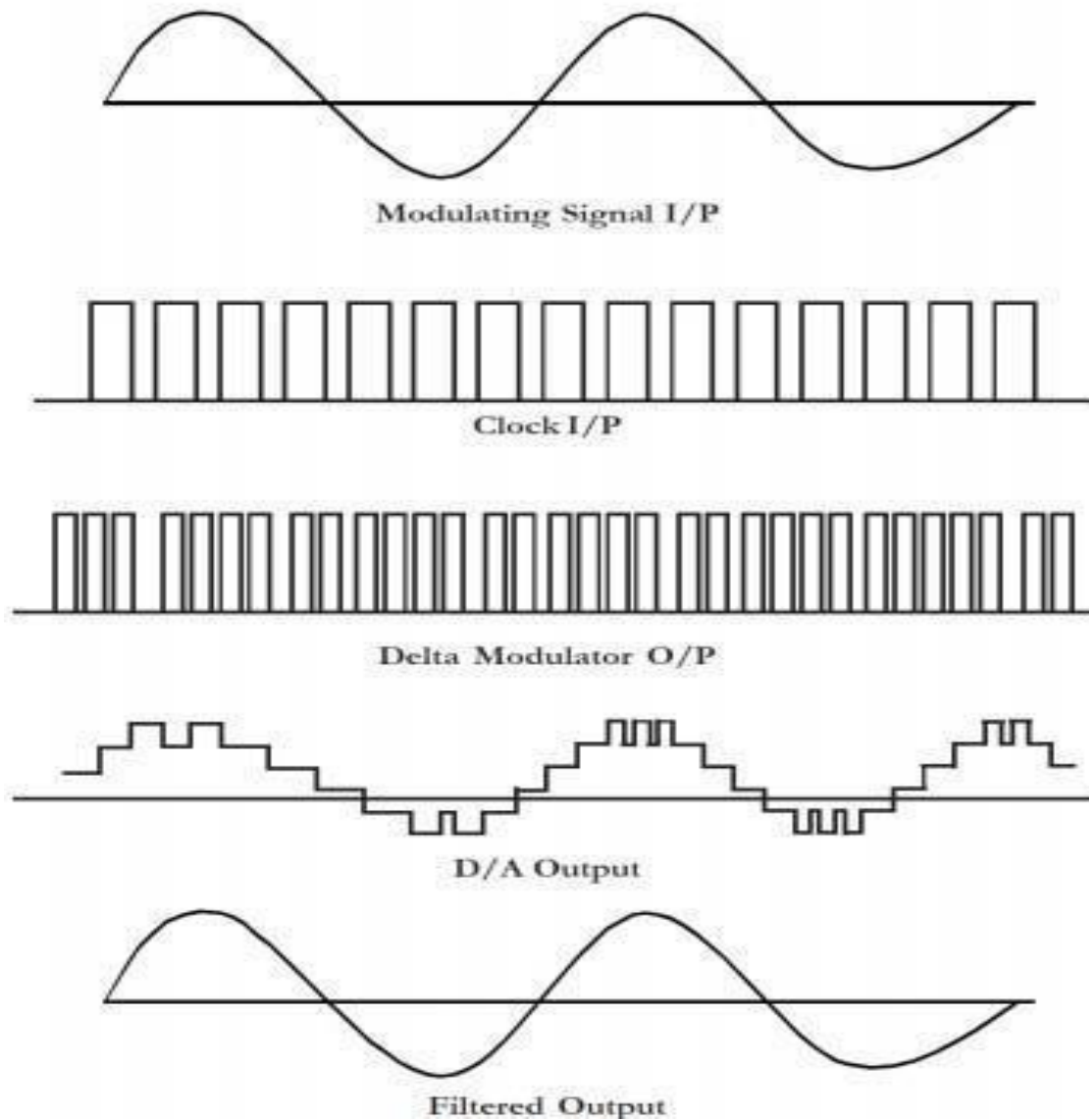
PROCEDURE:

1. Switch on the experimental board.
2. Connect the clock signal of Bit clock generator to the bit clock input of Delta Modulator circuit.
3. Connect modulating signal of the modulating signal generator to the modulating Signal input of the Delta modulator.
4. Observe the modulating signal on Channel 1 of CRO
5. Observe the Delta modulator output on channel 2 of CRO
6. Connect the DM O/P of modulator to the DM I/P of Demodulator circuit
7. Connect the clock signal to the Bit clock I/P of Demodulator circuit.
8. Observe the demodulated O/P on channel 2 of CRO
9. Connect the demodulated O/P to the filter input of demodulator circuit.
10. Observe the demodulated O/P with filter on CRO

PRECAUTIONS:

- 1 .Avoid loose Connections.
2. Switch off the power supply during connections
3. Waveforms must be noted carefully.

MODEL WAVEFORMS:



OBSERVATIONS TABLE:

Name of the signal	Amplitude	Time period	Frequency
Modulating (AF) Signal			
Carrier Signal(Pulse)			
Staircase Signal			
DM Signal			
Demodulated Signal			

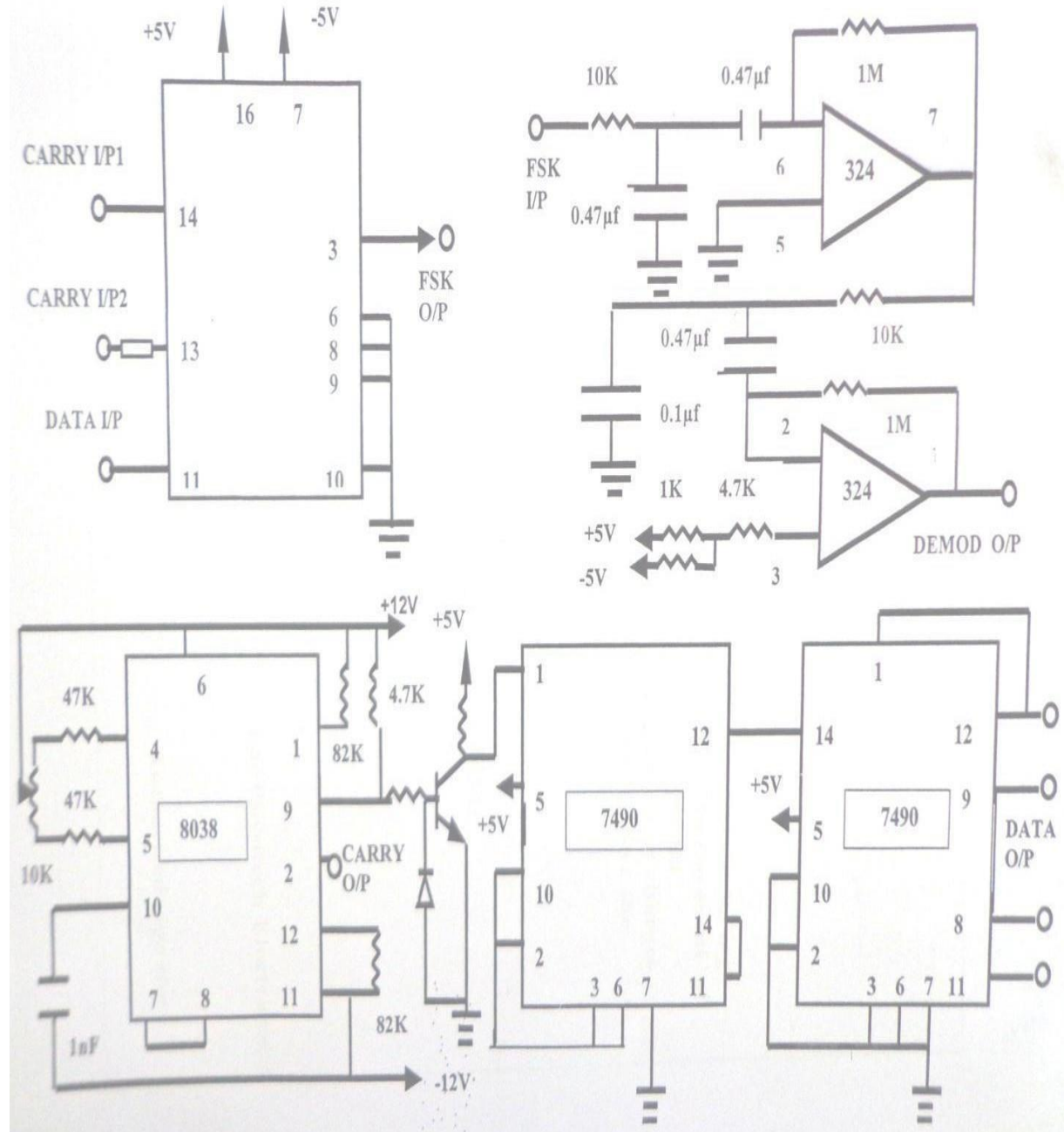
RESULT:

The digital data output of a given analog message signal is observed by using delta modulator and demodulated the original analog message signal from the modulated signal using delta demodulator. The staircase waveform according to digital data is obtained from the analog message signal. From the measurements we can observe that the amplitude and frequency values are the same at the receiver also.

VIVA QUESTIONS:

1. What are the advantages of Delta modulator?
2. What are the disadvantages of delta modulator?
3. How to overcome slope overload distortion?
4. How to overcome Granular or ideal noise?
5. What are the differences between PCM & DM?
6. Define about slope over load distortion?
7. What is the other name of Granular noise?
8. What is meant by staircase approximation?
9. What are the disadvantages of Delta modulator?
10. Write the equation for error at present sample?

FSK MODULATION & DEMODULATION CIRCUIT DIAGRAM



EXPT. NO:

DATE:

FREQUENCY SHIFT KEYING

AIM:

To generate the frequency shift keying signal for a given binary data at modulator end and also demodulate the original data input from the received FSK input at the receiver end.

APPARATUS:

1. FSK modulation & Demodulation trainer kit
2. Function generator
3. CRO
4. Connecting wires and probes.

THEORY:

In FSK, the waveform is generated by switching the frequency of the carrier between two values corresponding to the binary information which is to be transmitted. Here the carrier frequency varies from lowest to highest point i.e. carrier swing is known as Frequency shift keying. FSK signaling schemes find a wide range of applications in low speed digital data transmission systems.

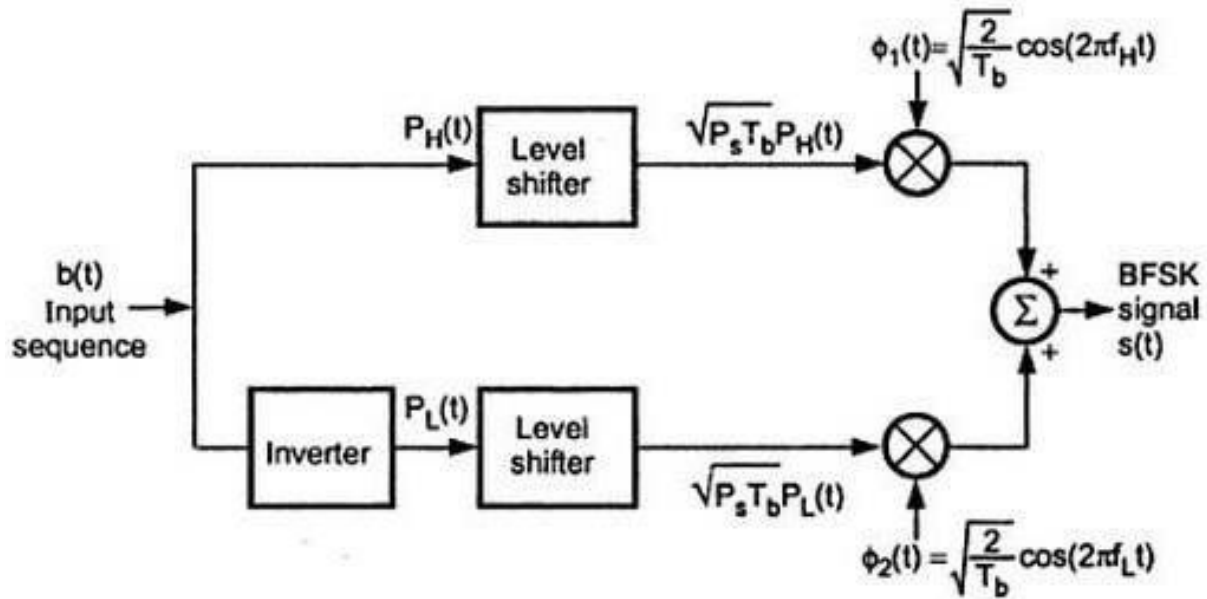
Frequency-shift keying (FSK) is a frequency modulation scheme in which digital information is transmitted through discrete frequency changes of a carrier wave.^[1] The simplest FSK is binary *FSK* (BFSK). BFSK uses a pair of discrete frequencies to transmit binary (0s and 1s) information.^[2] With this scheme, the "1" is called the mark frequency and the "0" is called the space frequency

Minimum-shift keying

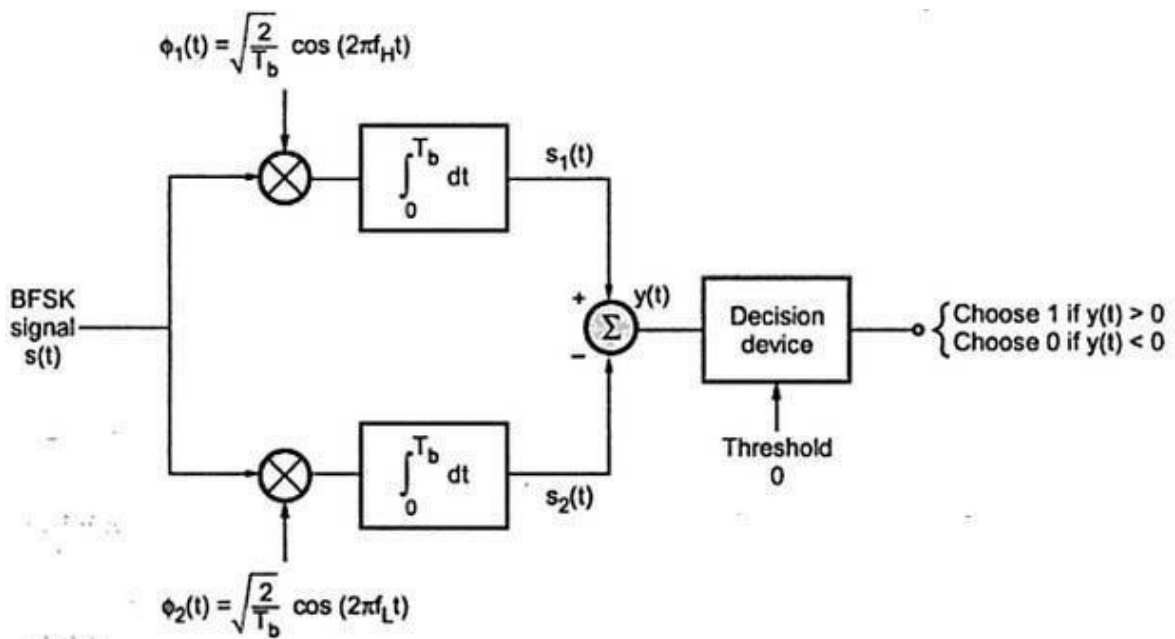
Minimum frequency-shift keying or minimum-shift keying (MSK) is a particular spectrally efficient form of coherent FSK. Consequently, the waveforms that represent a 0 and a 1 bit differ by exactly half a carrier period. The maximum frequency deviation is $\delta = 0.25 f_m$, where f_m is the maximum modulating frequency. As a result, the modulation index m is 0.25. This is the smallest FSK modulation index that can be chosen such that the waveforms for 0 and 1 are orthogonal. A variant of MSK called GMSK is used in the GSM mobile phone standard.

BLOCK DIAGRAM:

FSK MOUDULATION:



FSK DEMODULATION:



Audio FSK

Audio frequency-shift keying (AFSK) is a modulation technique by which digital data is represented by changes in the frequency (pitch) of an audio tone, yielding an encoded signal suitable for transmission via radio or telephone. Normally, the transmitted audio alternates between two tones: one, the "mark", represents a binary one; the other, the "space", represents a binary zero.

AFSK differs from regular frequency-shift keying in performing the modulation at baseband frequencies. In radio applications, the AFSK-modulated signal normally is being used to modulate an RF carrier (using a conventional technique, such as AM or FM) for transmission.

APPLICATIONS:

AFSK is used in the U.S. based Emergency Alert System to notify stations of the type of emergency, locations affected, and the time of issue without actually hearing the text of the alert. AFSK is also used in the United States' Emergency Alert System to transmit warning information. It is used at higher bitrates for Weather copy used on Weatheradio by NOAA in the U.S.

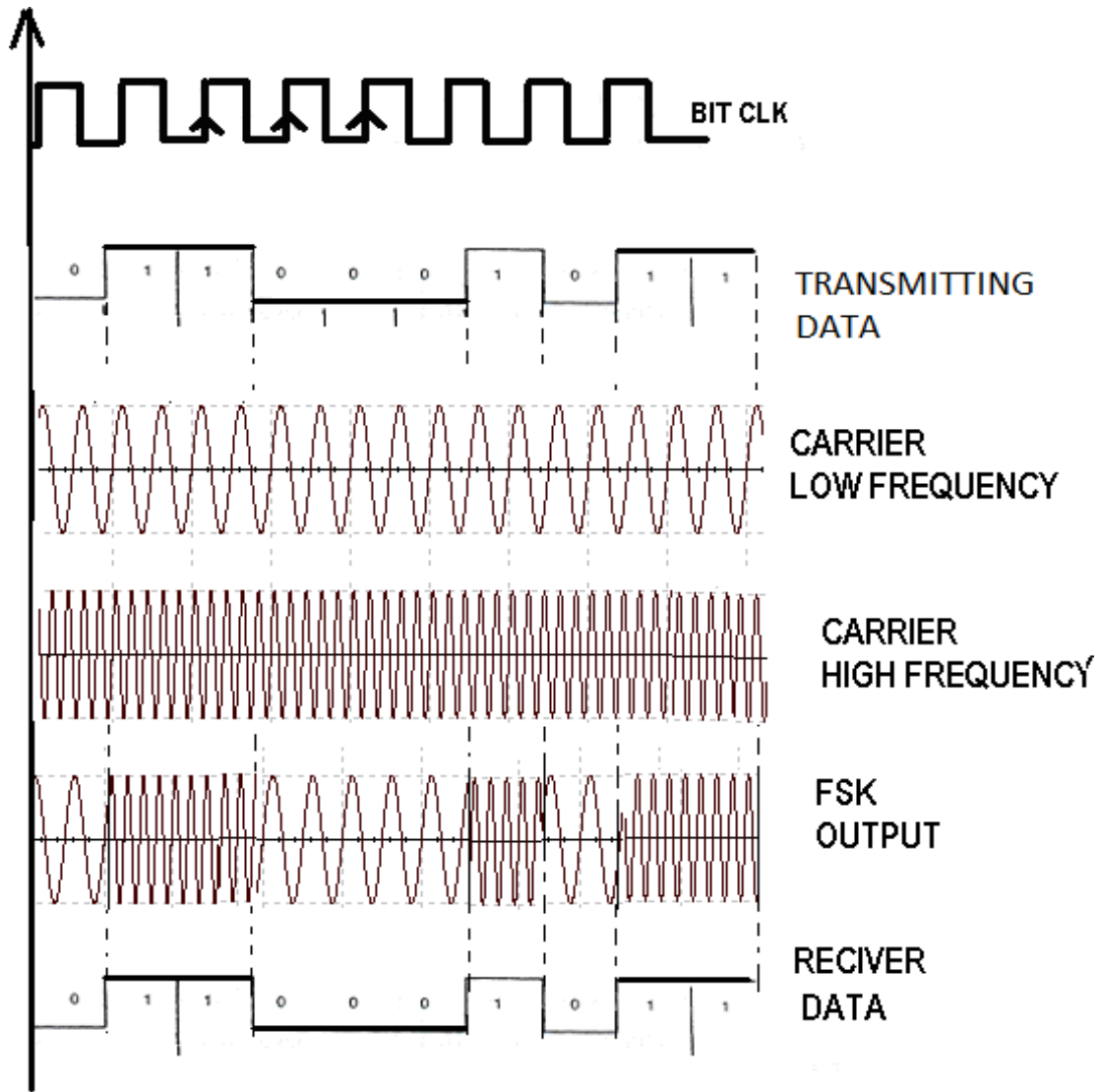
The CHU shortwave radio station in Ottawa, Canada broadcasts an exclusive digital time signal encoded using AFSK modulation

FSK is commonly used in Caller ID and remote metering applications: see FSK standards for use in Caller ID and remote metering for more details

PROCEDURE:

1. Connect the output of the carrier output provided on kit to the input of carrier input 1 terminal.
2. Also connect one of the Data output to the Data input terminal provided on kit.
3. Connect sine wave of certain frequency to the carrier input 2 terminal.
4. Switch ON function generator and FSK modulation and Demodulation kit.
5. Observe the FSK output by connecting it to CRO. Thus FSK modulation can be achieved.
6. For FSK Demodulation, Connect FSK output terminal to the FSK input terminal of demod.
7. Observe the Demodulated wave at Demodulated output terminal by connecting it to CRO.'

MODEL WAVEFORMS



OBSERVATIONS TABLE:

Name of the signal	Amplitude	Time period	Frequency
Modulating Signal Digital data			
Carrier Signal (HF sine wave)			
Carrier Signal (LF sine wave)			
FSK Signal			

Demodulated Signal Digital data			
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PRECAUTIONS:

- 1 .Avoid loose Connections.
2. Switch off the power supply during connections
3. Waveforms must be noted carefully.

RESULT:

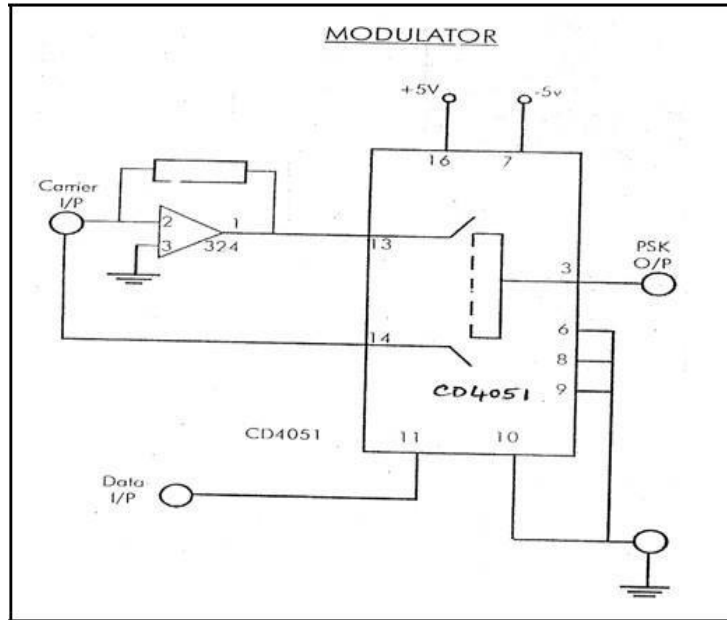
The FSK signal is observed for a given input data and the demodulated data is observed from the received FSK signal. Hence plotted the appropriate waveforms.

And also observed, the Frequency is shifted when the input data is keying from 0 to 1 and vice versa in the modulated FSK signal and retrieved the original input data from the received FSK signal. From the measurements we can observe that the amplitude and frequency values are the same at the receiver.

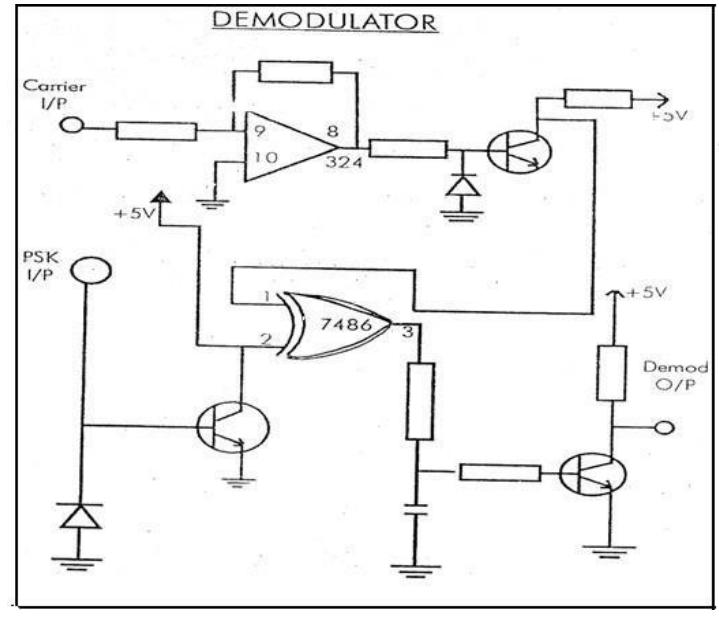
VIVA QUESTIONS:

1. Define Binary FSK signal?
2. What is meant by carrier swing?
3. Define Frequency deviation of FSK signal?
4. What are the advantages of this FSK signal?
5. Give the differences between FSK & FM?
6. Give the necessary equations for BFSK?
7. What are the applications of FSK?
8. What are the disadvantages of this FSK signal?

CIRCUIT DIAGRAM OF PSK MODULATOR



CIRCUIT DIAGRAM OF PSK DE MODULATOR



EXPT. NO:

DATE:

PHASE SHIFT KEYING

AIM:

To generate the phase shift keying signal for the given binary data at modulator end & also demodulate the PSK signal to receive the transmitted binary data.

APPARATUS:

1. PSK Modulation & Demodulation trainer kit
2. CRO
3. CRO Probes
4. Connecting wires (or) Patch cards

THEORY:

Phase shift keying or discrete phase modulation is another technique available for communicating digital information over band pass channels. In PSK signaling schemes the waveforms $s_1(t) = -A\cos\omega_c t$ & $S_2(T) = A\cos\omega_c t$ are used to convey binary digits 0 & 1 respectively. The binary PSK waveform $Z(t)$ can be described by, $Z(t) = D(t) A\cos\omega_c t$. Where $D(T)$ is a random binary waveform with period T_b & levels -1 & 1 . The only difference b/w the ASK & PSK waveform is that in the ASK scheme the carrier is switched on & off whereas in the PSK scheme the carrier is switched b/w levels $+A$ & $-A$.

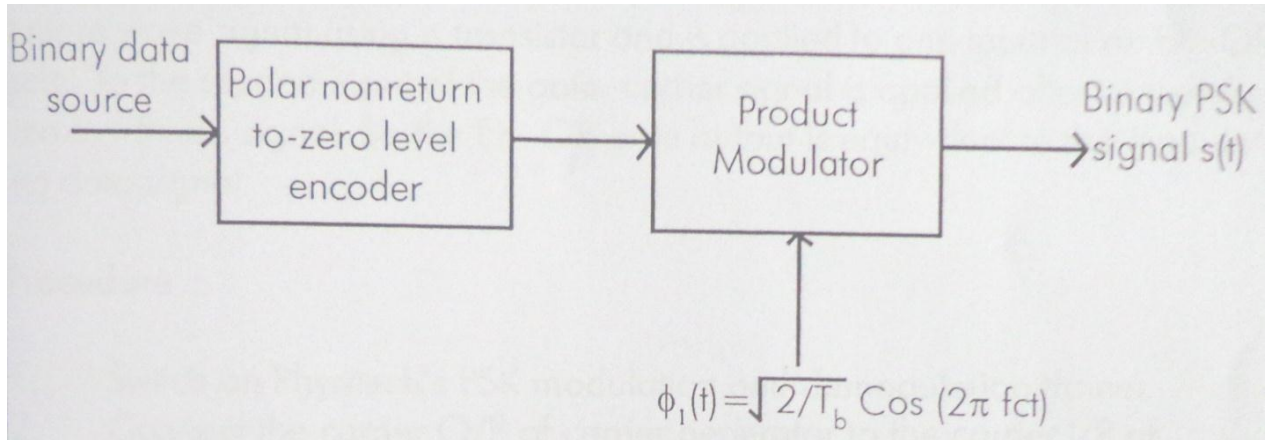
The differentially coherent PSK signaling scheme makes use of a clever technique designed to get around the need for a coherent reference signal at the receiver.

Phase-shift keying (PSK) is a digital modulation scheme that conveys data by changing, or modulating, the phase of a reference signal (the carrier wave).

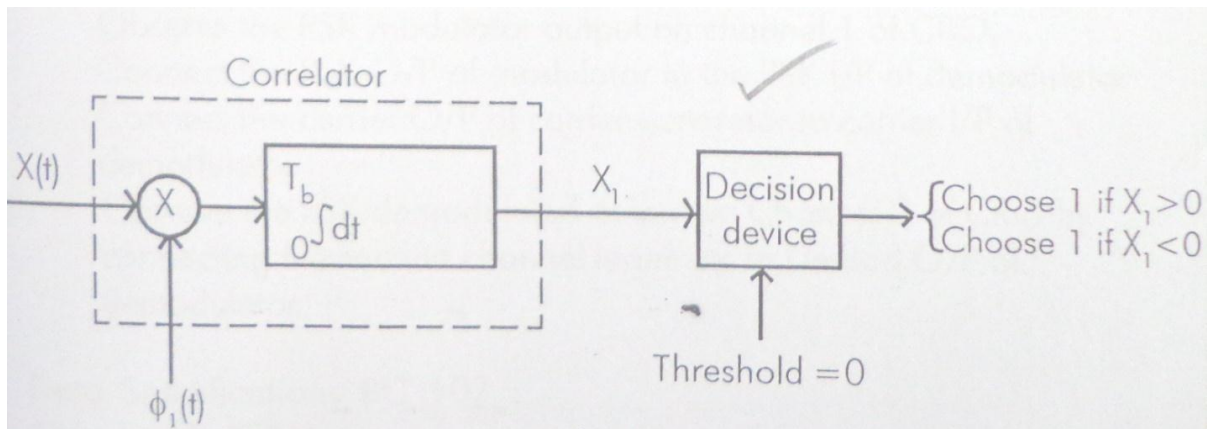
Any digital modulation scheme uses a finite number of distinct signals to represent digital data. PSK uses a finite number of phases; each assigned a unique pattern of binary digits. Usually, each phase encodes an equal number of bits. Each pattern of bits forms the symbol that is represented by the particular phase. The demodulator, which is designed specifically for the symbol-set used by the modulator, determines the phase of the received signal and maps it back to the symbol it represents, thus recovering the original data. This requires the receiver to be able

BLOCK DIAGRAM:

PSK MODULATOR:



PSK DEMODULATOR:



to compare the phase of the received signal to a reference signal — such a system is termed coherent (and referred to as CPSK).

BPSK (also sometimes called PRK, phase reversal keying, or 2PSK) is the simplest form of phase shift keying (PSK). It uses two phases which are separated by 180° and so can also be termed 2-PSK. It does not particularly matter exactly where the constellation points are positioned, and in this figure they are shown on the real axis, at 0° and 180° . This modulation is the most robust of all the PSKs since it takes the highest level of noise or distortion to make the demodulator reach an incorrect decision. It is, however, only able to modulate at 1 bit/symbol (as seen in the figure) and so is unsuitable for high data-rate applications.

BPSK is functionally equivalent to 2-QAM modulation. All convey data by changing some aspect of a base signal, the carrier wave (usually a sinusoid), in response to a data signal. In the case of PSK, the phase is changed to represent the data signal.

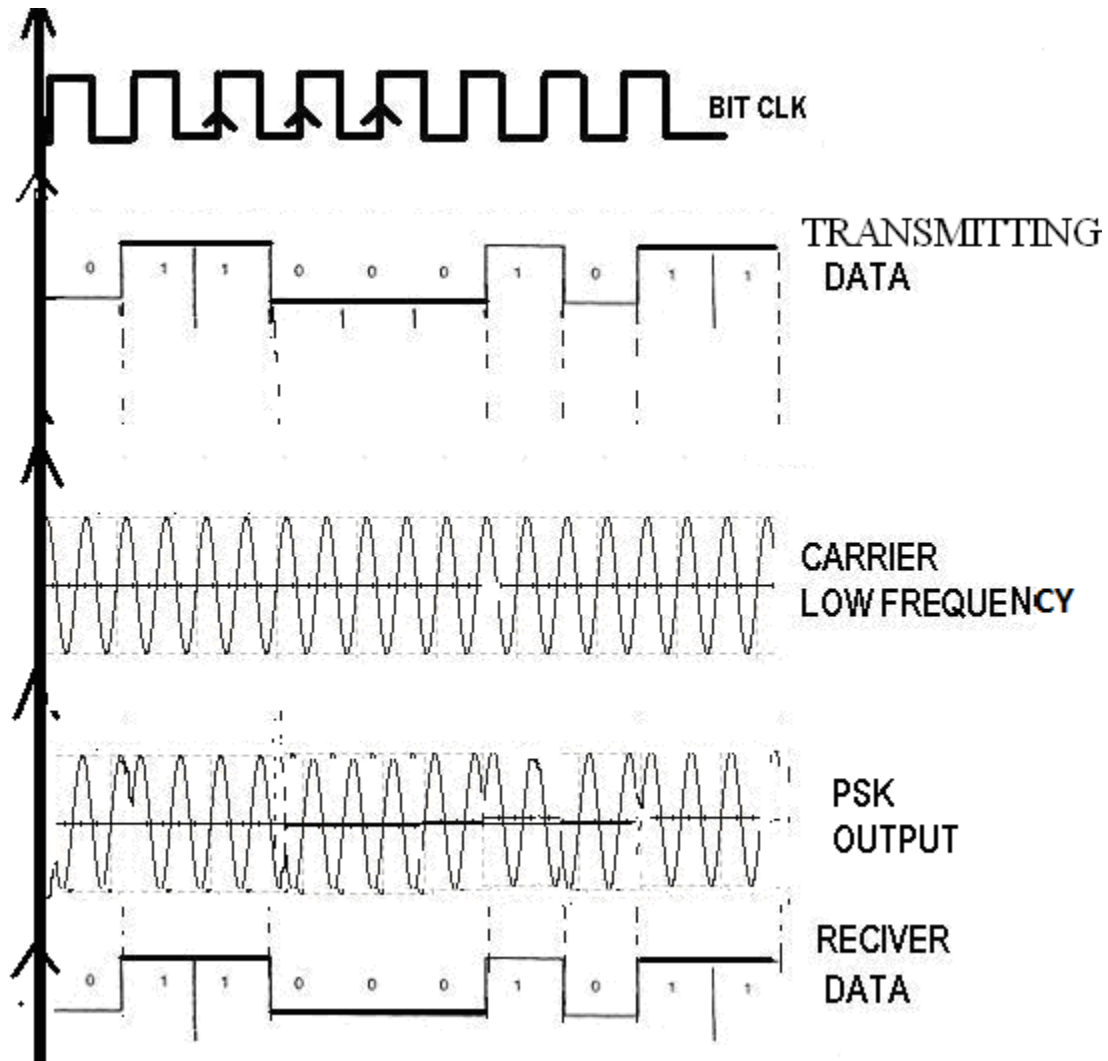
There are two fundamental ways of utilizing the phase of a signal in this way:

- By viewing the phase itself as conveying the information, in which case the demodulator must have a reference signal to compare the received signal's phase against; or
- By viewing the change in the phase as conveying information — differential schemes, some of which do not need a reference carrier (to a certain extent).

PROCEDURE:

1. Switch on Physitech's PSK modulation and demodulation trainer.
2. Connect the carrier O/P of carrier generator to the carrier I/P of modulator.
3. Connect the data O/P of Data generator to the Data I/P of Modulator.
4. Connect CRO terminals to the PSK O/P of modulator.
5. Observe the PSK modulator output on channel 1 of CRO.
6. Connect the PSK O/P of modulator to the PSK I/P of demodulator.
7. Connect the carrier O/P of carrier generator to carrier I/P of demodulator.
8. Observe the PSK demodulated output on channel 2 of CRO by connecting the second channel terminals to Demod O/P of demodulator.

OUTPUT WAVEFORMS:



OBSERVATIONS TABLE:

Name of the signal	Amplitude	Time period	Frequency
Modulating Signal Digital data			
Carrier Signal (HF sine wave)			
PSK Signal			
Demodulated Signal Digital data			

PRECAUTIONS:

- 1 .Avoid loose Connections.
2. Switch off the power supply during connections
3. Waveforms must be noted carefully.

RESULT:

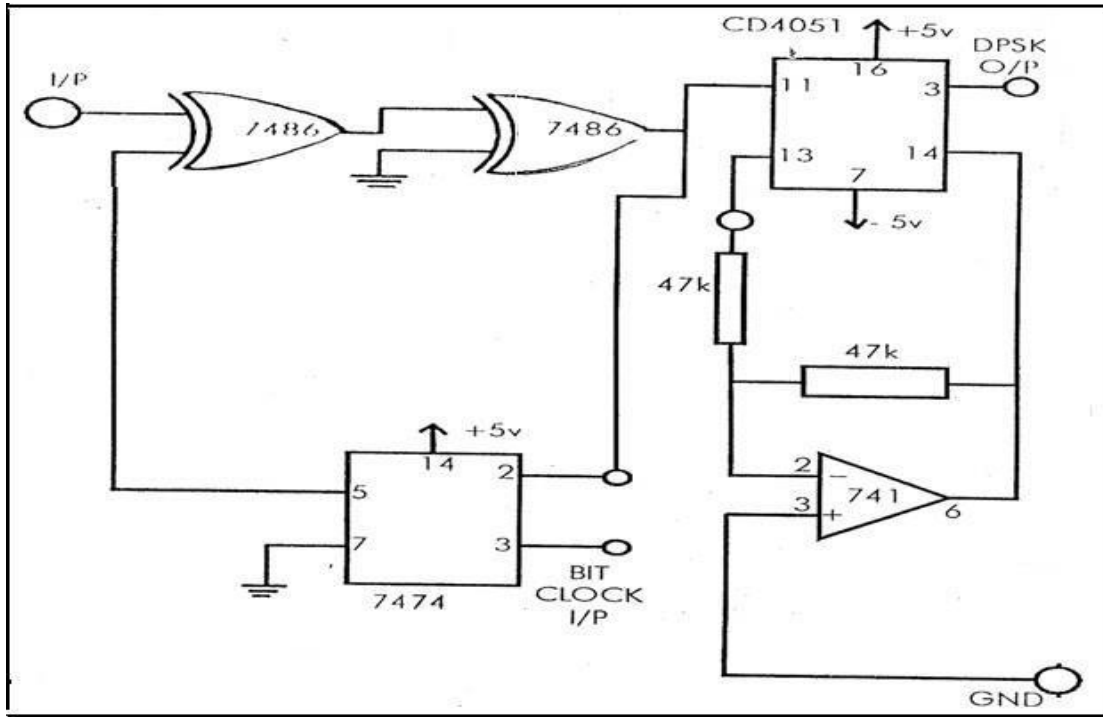
The PSK signal is observed for a given input data and the demodulated data is observed from the received PSK signal. Hence plotted the appropriate waveforms.

And also observed that the PHASE is shifted when the input data is keying from 0 to 1 and vice versa in the modulated PSK signal and retrieved the original input data from the received PSK signal. From the measurements we can observe that the amplitude and frequency values are the same at the receiver also.

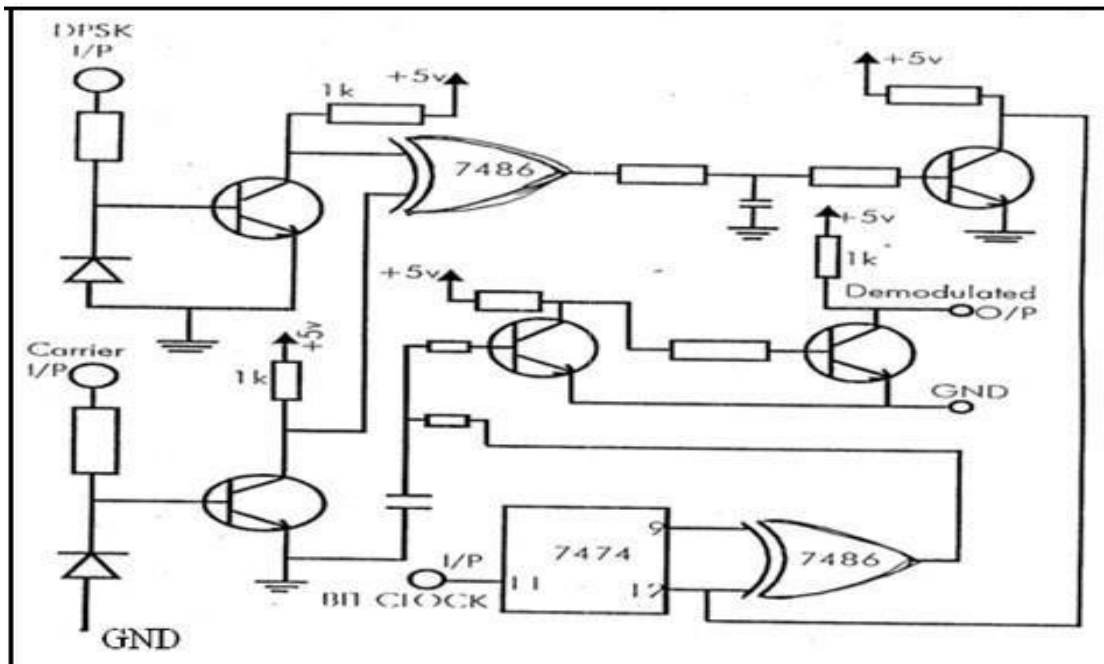
VIVA QUESTIONS:

1. What is the bandwidth requirement of BPSK?
2. What is the expression for error probability of BPSK reception using coherent matched filter detection?
3. What are the draw backs of BPSK?
4. Draw the Power spectral density of BPSK?
5. What are the major differences between DPSK&BPSK?
6. What are the advantages of BPSK over a PSK signal?
7. Give the necessary equations for BPSK?
8. What are the various types of PSK?

CIRCUIT DIAGRAM OF DPSK MODULATOR:



CIRCUIT DIAGRAM OF DPSK DE MODULATOR:



EXPT. NO:

DATE:

DIFFERENTIAL PHASE SHIFT KEYING

AIM:

To generate differentially phase shift keying signal at modulator end and also demodulate the original binary data from the DPSK signal at the demodulator end.

APPARATUS:

1. DPSK modulation and demodulation trainer kit
2. CRO
3. CRO Probes
4. Connecting wires (or) Patch cards

THEORY:

We may view DPSK as the non-coherent version of PSK. It eliminates the need for adjustment coherent reference signal at the receiver by connecting two basic operations at the transmitter.

1. Differential encoding at the transmitter.
2. Phase shift keying

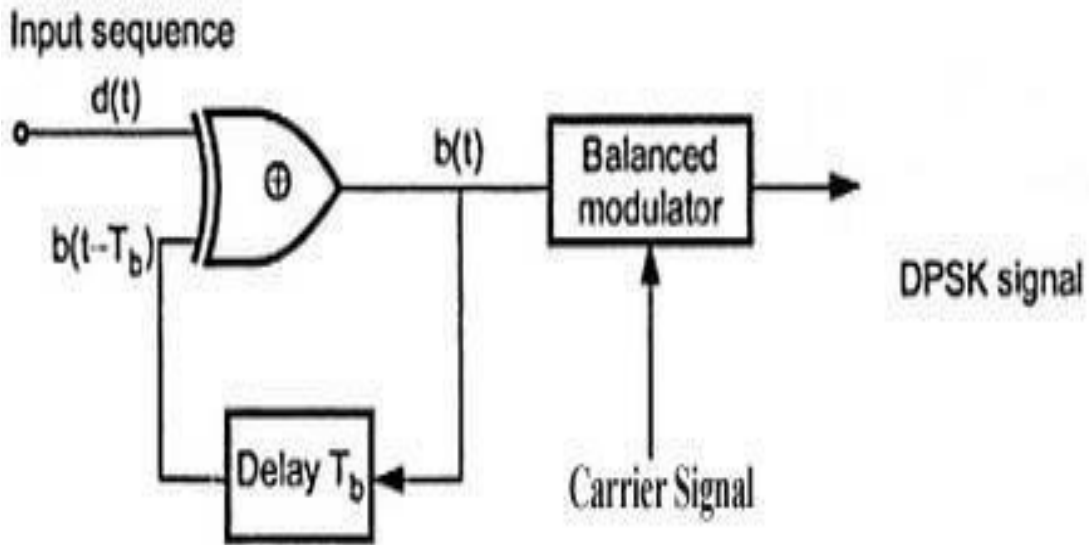
Hence differential encoding means the given input data will be done EX-OR operation with the previous encoded bit. Now the process of Phase shift keying will be done for both differentially encoded data and the carrier signal.

Differential encoding:

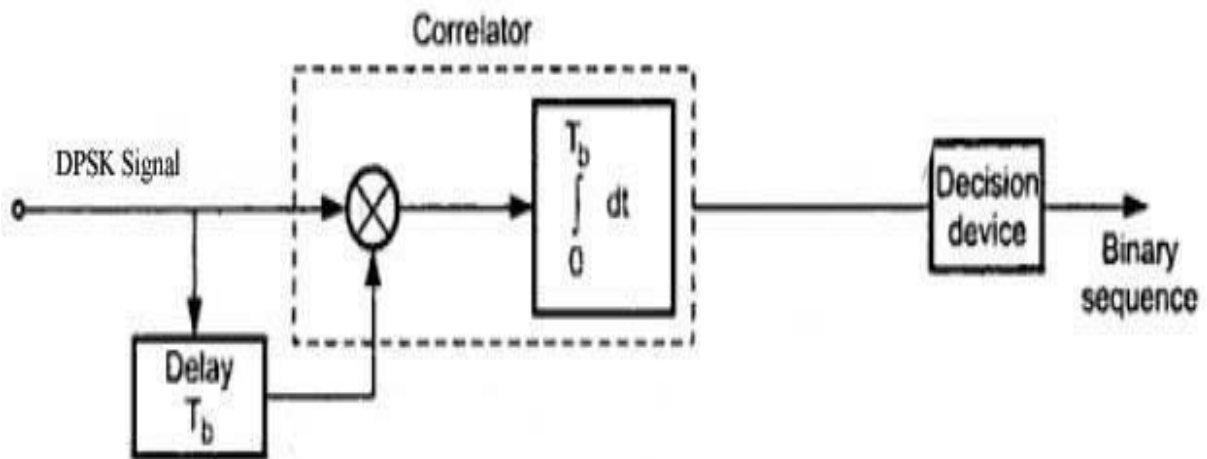
Differential phase shift keying (DPSK) is a common form of phase modulation that conveys data by changing the phase of the carrier wave. As mentioned for BPSK and QPSK there is an ambiguity of phase if the constellation is rotated by some effect in the communications through which the signal passes. This problem can be overcome by using the data to change rather than set the phase. For example, in differentially encoded BPSK a binary '1' may be transmitted by adding 180° to the current phase and a binary '0' by adding 0° to the current phase. Another variant of DPSK is Symmetric Differential Phase Shift keying, SDPSK, where encoding would be $+90^\circ$ for a '1' and -90° for a '0'.

BLOCK DIAGRAM:

DPSK MODULATOR:

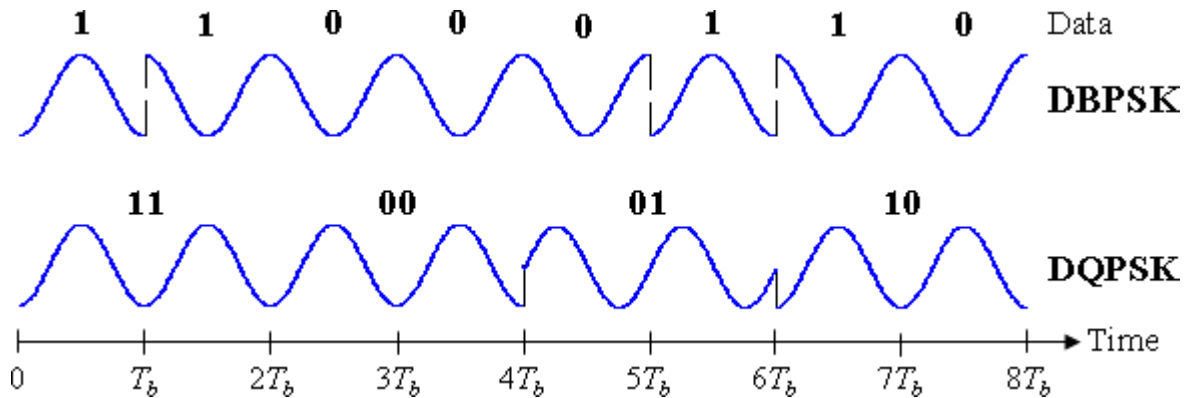


DPSK DEMODULATOR:



In differentially encoded QPSK (DQPSK), the phase-shifts are 0° , 90° , 180° , -90° corresponding to data '00', '01', '11', '10'. This kind of encoding may be demodulated in the same way as for Non-differential PSK but the phase ambiguities can be ignored. Thus, each received symbol is demodulated to one of the M points in the constellation and a comparator then computes the difference in phase between this received signal and the preceding one. The difference encodes the data as described above. Symmetric Differential Quadrature Phase Shift Keying (SDQPSK) is like DQPSK, but encoding is symmetric, using phase shift values of -135° , -45° , $+45^\circ$ and $+135^\circ$.

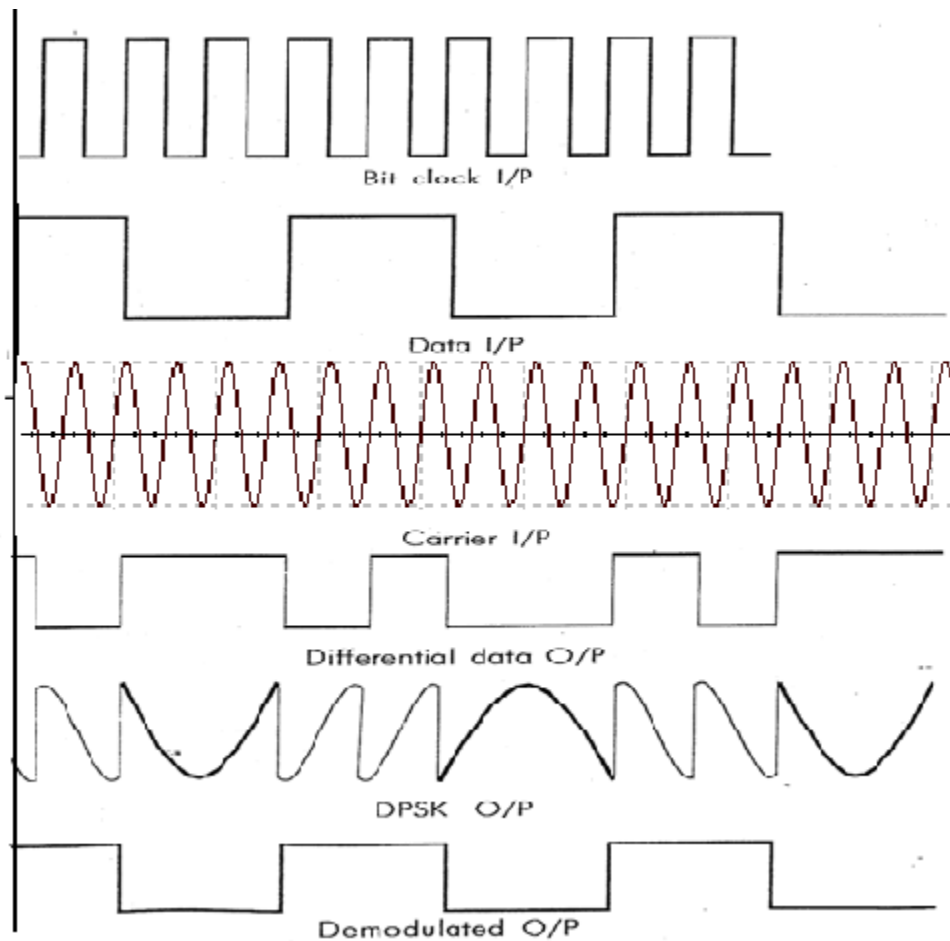
The modulated signal is shown below for both DBPSK and DQPSK as described above. In the figure, it is assumed that the signal starts with zero phase, and so there is a phase shift in both signals at $t = 0$.



Timing diagram for DBPSK and DQPSK. The binary data stream is above the DBPSK signal. The individual bits of the DBPSK signal are grouped into pairs for the DQPSK signal, which only changes every $T_s = 2T_b$.

Analysis shows that differential encoding approximately doubles the error rate compared to ordinary M -PSK but this may be overcome by only a small increase in E_b/N_0 . Furthermore, this analysis (and the graphical results below) are based on a system in which the only corruption is additive white Gaussian noise(AWGN). However, there will also be a physical channel between the transmitter and receiver in the communication system. This channel will, in general, introduce an unknown phase-shift to the PSK signal; in these cases the differential schemes can yield a better error-rate than the ordinary schemes which rely on precise phase information. For a signal that has been differentially encoded, there is an obvious alternative method of demodulation.

MODEL WAVE FORMS:



OBSERVATIONS TABLE:

Name of the signal	Amplitude	Time period	Frequency
Modulating Signal Digital data			
Carrier Signal (HF sine wave)			
DPSK Signal			
Demodulated Signal Digital data			

PROCEDURE:

1. Switch on differential phase shifting keying trainer.
2. Connect the carrier output of carrier generator to the 13th pin of CD4051 (Analog mux) of Modulator.
3. Connect the bit clock output to the Bit clock input at pin 3 of 7474(8 – bit converter) of Modulator.
4. Connect the data output of data generator to the input of modulator circuit.
5. Observe the differential data output at pin 2 of 7474 IC on channel; 1 of CRO.
- 6 .Observe the DPSK modulated output on channel 2 of CRO
- 7 .During demodulation, connect the DPSK modulated output to the DPSK I/P Of Demodulator.
- 8 .Connect the Bit clock O/P to the Bit clock I/P of Demodulator and also connect the carrier O/P to the carrier I/P of demodulator.
- 9 .Observe the demodulated data O/P at demodulated O/P of Demodulator.
10. The frequency of modulation data signal should be equal to the demodulated O/P.

PRECAUTIONS:

- 1 .Avoid loose Connections.
2. Switch off the power supply during connections
3. Waveforms must be noted carefully.

RESULT:

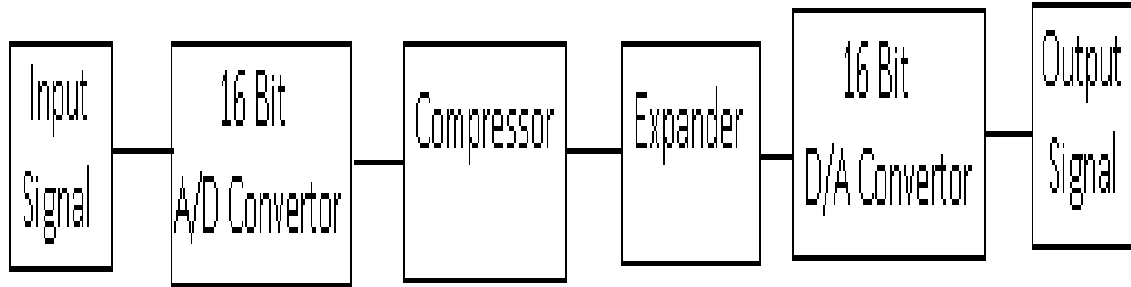
The DPSK signal is observed for a given input data and the demodulated data is observed from the received DPSK signal. Hence plotted the appropriate waveforms.

And also observed that the PHASE is shifted when the DIFFERENTIAL input data is keying from 0 to 1 and vice versa in the modulated DPSK signal and retrieved the original input data from the received DPSK signal. From the measurements we can observe that the amplitude and frequency values are the same at the receiver also.

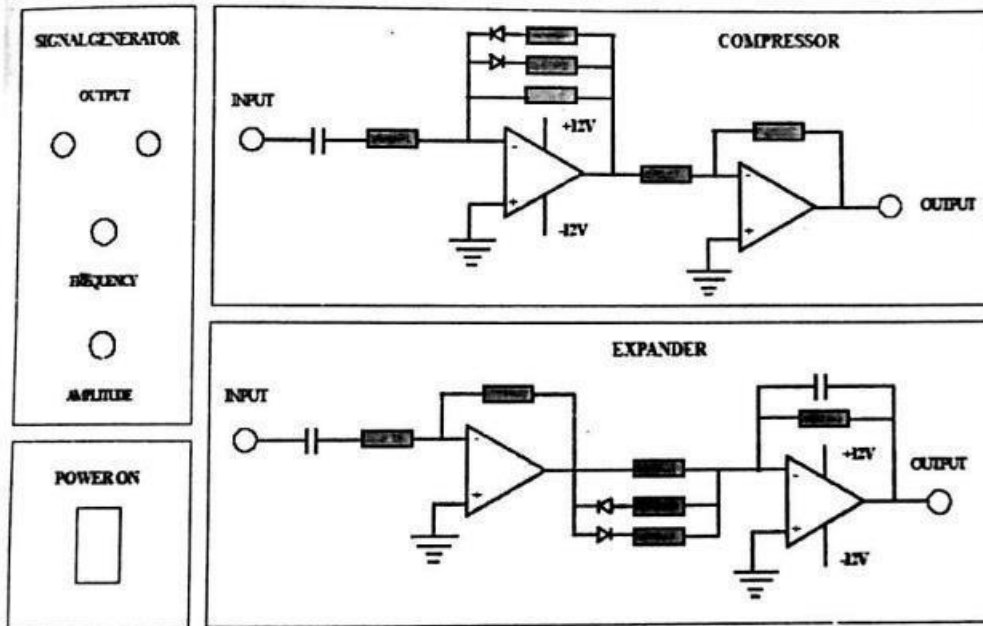
VIVA QUESTIONS:

1. What is the difference between PSK&DPSK?
2. What is the band width requirement of a DPSK?
3. Explain the operation of DPSK detection?
4. What are the advantages of DPSK?
5. What is meant by differential encoding?
6. In Differential encoding technique which type of logic gates are used?
7. Is there any change in the output data when the starting reference bit is changed?

BLOCK DIAGRAM OF COMPANDING:



Block diagram:



EXPT. NO:

DATE:

COMPANDING

AIM:

To observe the waveforms of compander and expander circuits

APPARATUS:

1. Companding trainer kit
2. CRO
3. Patch cards

THEORY:

Companding is a compressing and expanding technique used in digital communications system. With this technique we can improve noise performance at low level signals and improve the dynamic range of the signal that can be handled by the channel. While coding the signal we give higher resolution at low levels and lower resolution at high levels. This makes the signal compressed in the code domain. By applying the reverse mapping at the decoding end we get back the original signal. By allocating more bits at lower signals we reduce the quantization noise (the noise created by the LSB). A-law and μ -law algorithms encode 14-bit and 13-bit signed linear PCM samples to logarithmic 8-bit samples. Companding is non uniform quantization. If the quantizer characteristics is non linear and the step size is not constant instead if it is variable, dependent on amplitude of input signal then the quantization is known as Non uniform quantization. In this the step size is reduced with the reduction in the signal level.

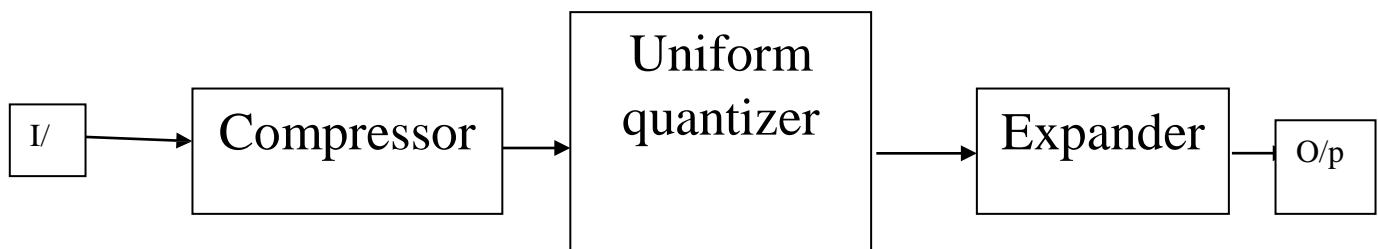


Fig: Companding Model

WAVE FORMS:

Message Signal with 100Hz frequency

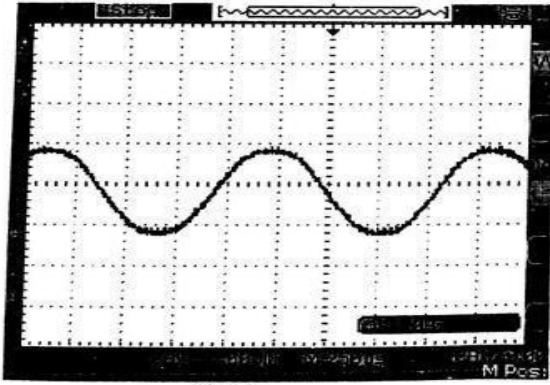


Fig.1

Companing Signal with respect to Message Signal :

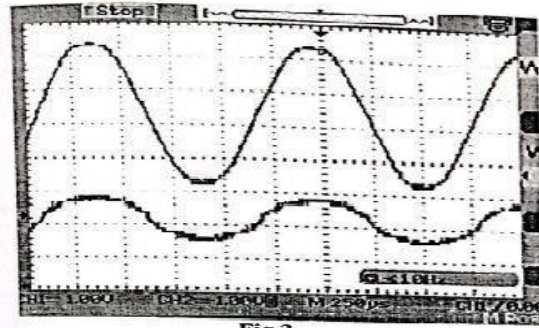


Fig.2

Expanding Signal with respect to Message Signal :

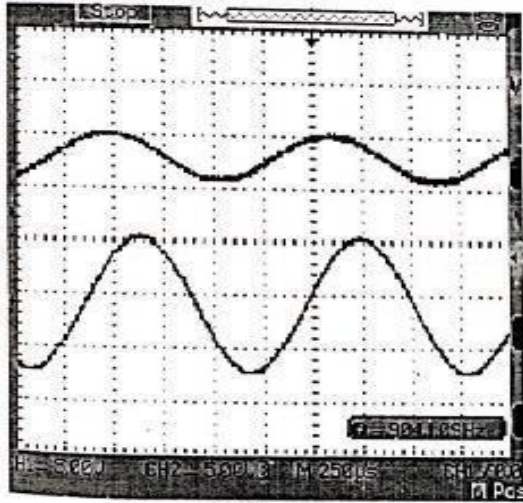


Fig.3

Reconstructed signal with respect to message signal:

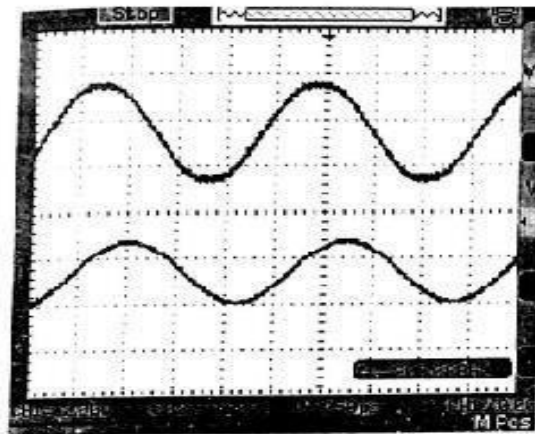


Fig.4

OBSERVATIONS TABLE 1:

Name of the signal	Amplitude	Time period	Frequency
Modulating Signal			
Companing Signal			
Expanding Signal			
Reconstructed Signal			

There are mainly two companding methods. They are:

1. μ - law Companding:

$$Z(x) = (\text{sgn}x) \ln (1+\mu|x|/x_{\max})/\ln (1+\mu) \quad \text{where } 0 \leq |x|/x_{\max} \leq 1.$$

2. A-law Companding:

$$\frac{z(x)}{x_{\max}} = \frac{A|x|/x_{\max}}{1 + \ln A} \quad \text{for } 0 \leq \frac{|x|}{x_{\max}} \leq \frac{1}{A}$$
$$\frac{z(x)}{x_{\max}} = \frac{1 + \ln \left[\frac{A|x|}{x_{\max}} \right]}{1 + \ln A} \quad \text{for } \frac{1}{A} \leq \frac{|x|}{x_{\max}} \leq 1$$

PROCEDURE:

COMPANDER

1. Switch on the trainer kit.
2. Observe the message signal with variable frequency 100Hz to 500Hz and variable amplitude 0 to 5V.
3. Set the message signal as 100Hz frequency and 4V peak to peak amplitude by using CRO as shown in figure 1.
4. Connect the signal to input of compander circuit.
5. Observe the logarithmic decrease in amplitude of output of companding signal with respect to message signal as shown in figure 2.
6. Note down the waveform of compander circuit.

EXPANDER:

1. Set the message signal as 100Hz frequency and 4V peak to peak amplitude and connect it to the input of expander kit.
2. Connect compander output to input of expander.
3. Observe and note down message signal and reconstructed signal as shown in figure 4

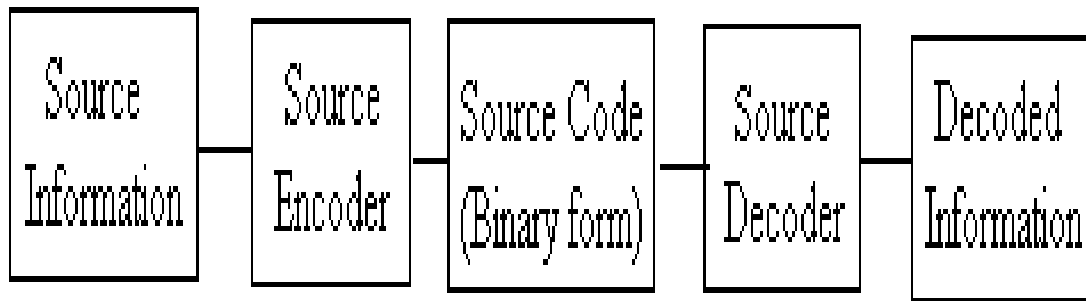
RESULT:

Thus we have observed the waveforms of Compander and Expander circuits.

\VIVA QUESTIONS:

1. What is the use of Companding?
2. How many types of Companding are they & what are they?
3. Companding comes under which type of quantization ?(whether uniform or non uniform)
4. What are the blocks present in Compander?
5. What is the purpose of compressor and expander?

BLOCK DIAGRAM OF SOURCE ENCODER AND DECODER:



EXPT. NO:

DATE:

SOURCE ENCODER AND DECODER

AIM:

To select information having some probability of occurrence of each symbol and applying a source coding using one of the sources coding techniques i.e.huffman` coding, observing the size of the coded information, sending the minimized packet, decoding at the receiving end getting back the full information sent.

APPARATUS:

1. Source encoder and decoder trainer kit
2. Adapter

THEORY:

A Conversion of the output of discrete memory less source (DMS) into a sequence of binary symbols is called source coding. The device that performs this conversion is called Source Encoder.

Objective of Source Coding is to minimize the average bit rate required for representation of the source by reducing the redundancy of the information source.

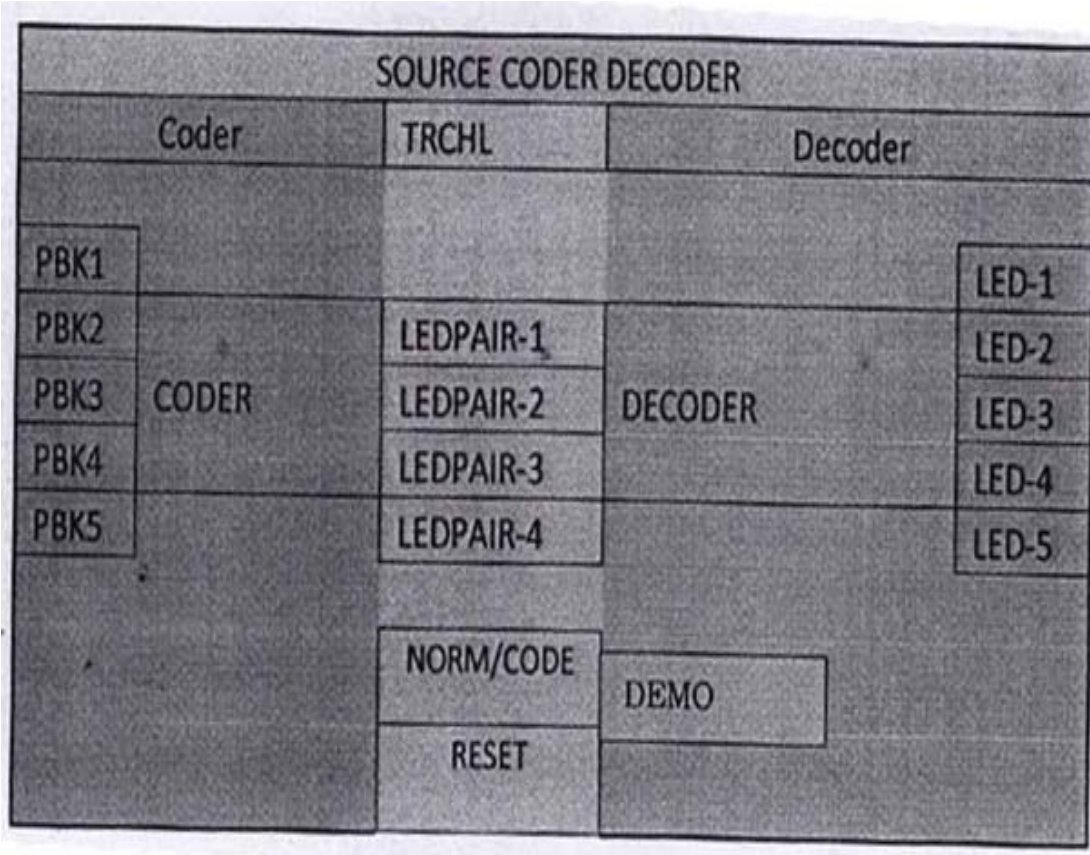
Source coding is a technique of compressing the source information size based on the probability of occurrence of each information symbol. Decoding is the reverse process to get back the full source information. In everyday we employ this in transferring big files, particularly image or voice files by zipping them and transferring to the destination and unzipping at the destination.

Example: Huffman coding. This Huffman coding is a type of variable length codes. By using this coding, frequently occurring symbol is assigned with less no. of bits in the code where as rarely occurred symbol is assigned with large no. of bits. Source coding provides security to our information.

The important parameters of Source encoder and decoder are:

- | | |
|--------------------|-----------------------------|
| 1. Block size | 3.Average data rate |
| 2. Codeword length | 4.Efficiency of the encoder |

EXPERIMENT PROCEDURE:



OBSERVATIONS TABLE:

Information Text	Information bits without coding	Information bits with coding	Decoded Information
ABRAKADABRA			
DABRAKAABRA			
ABRAKAABRAKA			
DABRAKADABRA			

PROCEDURE:

1. First observe the signal chain.
2. Now verify how many bits are taken to transmit ABRAKADABRA in normal and source coded mode.
3. Apply an input symbol by pressing any one of the input symbol keys. Ex: To transmit a letter 'A', press the key marked 'A'.
4. Observe how many bits are transmitted for the given input symbol and note down the bits being transmitted on the LEDs(1=RED, 0=GREEN)
5. Observe if the corresponding output LED is glows corresponding to the symbol pressed at the input.

PRECAUTIONS:

- 1 .Avoid loose Connections.
2. Switch off the power supply during connections
3. Waveforms must be noted carefully.

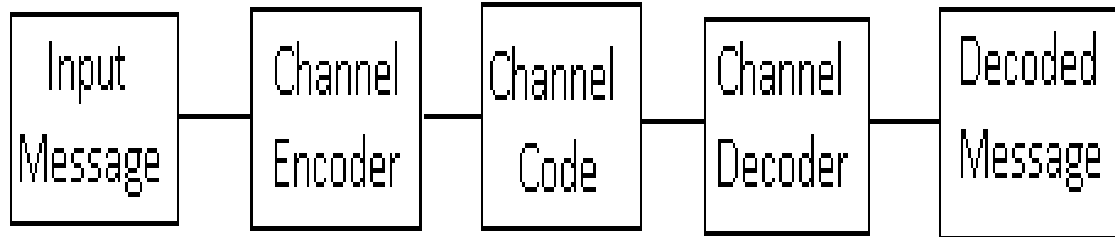
RESULT:

The source code for the given information & the size of the coded information are observed and also decoded the full information sent getting back at the receiving end.

VIVA QUESTIONS:

1. What is the purpose of source encoding and decoding?
2. What are the advantages of source coding?
3. What are the different source coding techniques used?
4. After source coding, message is in which form?
5. Define Shannon's source coding theorem.

BLOCK DIAGRAM OF LINEAR BLOCK CODES ENCODER AND DECODER



EXPT. NO:

DATE:

LINEAR BLOCK CODES ENCODER AND DECODER

AIM:

To observe that the errors received through a noisy channel can be removed by employing the error detection and correction code.

APPARATUS:

1. Linear block codes encoder and decoder trainer kit
2. Adapter

THEORY:

Linear block codes are called as channel coding techniques which are used for the purpose of detecting and correcting errors occurring in the communication channel. When information is represented in blocks of 'k' bits we can add a few extra bits to the information bits i.e. provide some redundancy and detect/correct the errors from the received data. More redundancy we provide more correction we can have. Most common example of linear block codes is Hamming code. By using hamming code technique with given no. of extra bits we can extract maximum advantage of detection and correction. Hamming code (n,k) places symbols represented by 'n' bits having 'k' information bits at a maximum distance from each other allowing us to detect more errors correct more errors. In this experiment hamming (7, 4) is used i.e. 4 information bits, 7 bits in the codeword and 3 parity bits (i.e.-k=7-4=3).

A code is linear if the sum of any two code vectors produces another code vector. This shows that any code vector can be expressed as a linear combination of other code vectors.

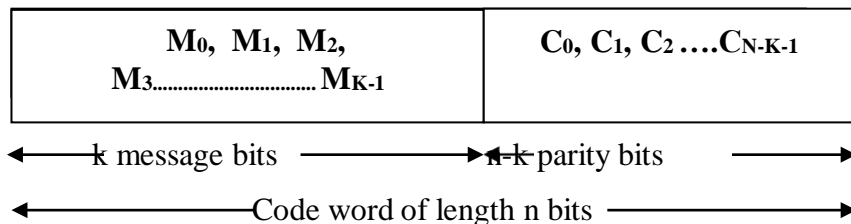
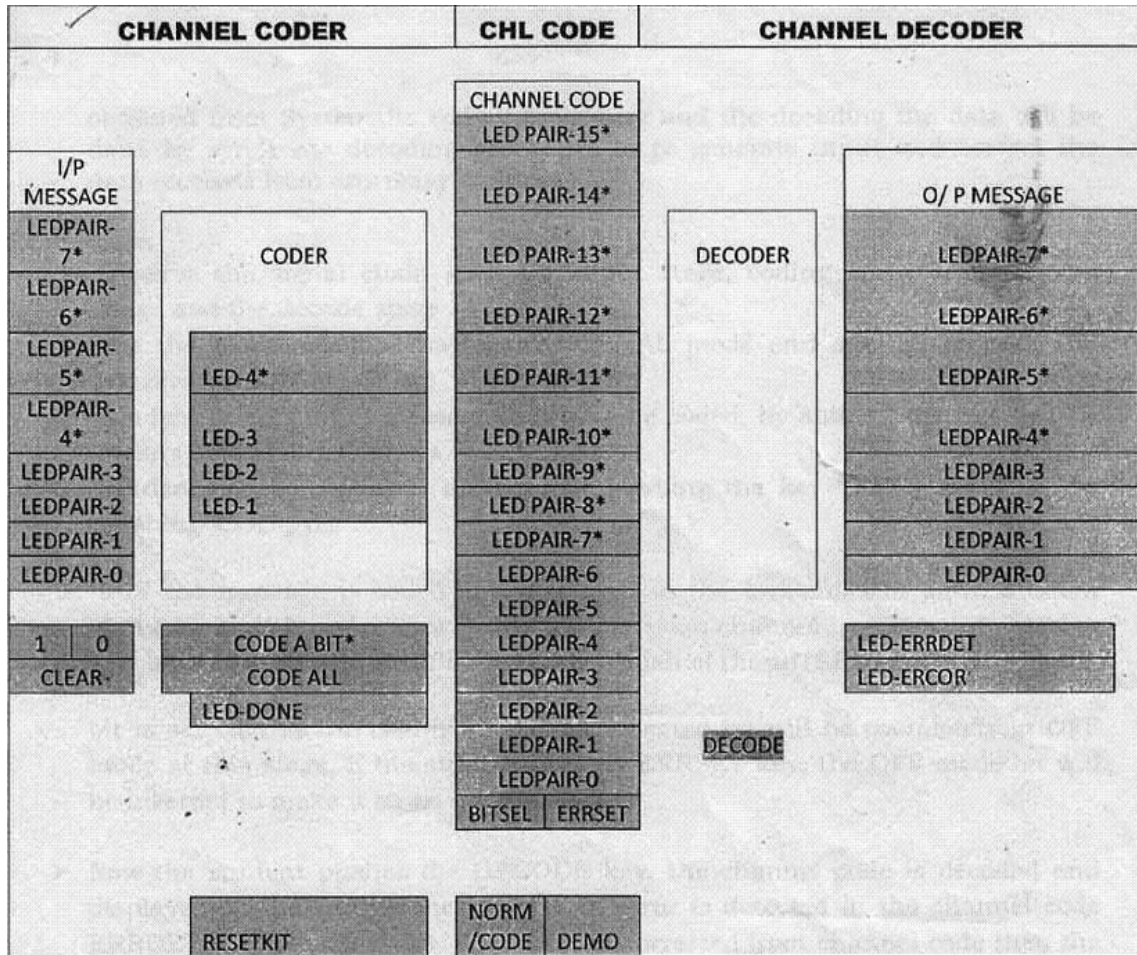


Fig: Structure of code word for a linear block code

EXPERIMENT PROCEDURE:



OBSERVATIONS TABLE:

Type of error	4-bit Input message(abcd)	3-Parity bits(xyz)	Channel Code(7bit) (xyzabcd)	Decoder Input	Decoder Output	Error detected	Error corrected
No error	1010					-----	-----
Single error	1000					Yes/No	Yes/No
Double error	0111					Yes/No	Yes/No

PROCEDURE:

1. Observe the signal chain, i.e. the input stage, coding stage, transmission stage and the Decode stage.
2. Put the mode selection switch in NORMAL/CODE mode and see the process and observe Output.
3. Now select input message that is to be coded, by shifting the bits 0/1 by means of pressing the Keys 0, 1, CLEAR.
4. Now the parity bits are added for the given input message by pushing the key 'CODE A BIT' Or by pushing 'CODEALL'.
5. Now the message is coded and displayed in the transmission path. Now push the DECODE Key, the channel code is decoded and displayed as the output message having no errors.
6. Due to channel noise, errors are introduced in the channel, now errors are introduced by means of pressing the keys BITSEL and ERRSET. On every push of the BITSEL one bit is Selected in the channel code, now if ERRSET is pressed the existing bit is inverted to make it as an error.
7. Now again DECODE key is pressed to display output message. If an error is detected in the Channel code ERRDETECTED LED glows, if an error is corrected from the channel code ERRCORRECTED LED glows in the decoder output stage.
8. Observe the decoded output by introducing single and double errors in the channel code.
9. Note down the observations in a table given.

PRECAUTIONS:

- 1 .Avoid loose Connections.
2. Switch off the power supply during connections
3. Waveforms must be noted carefully.

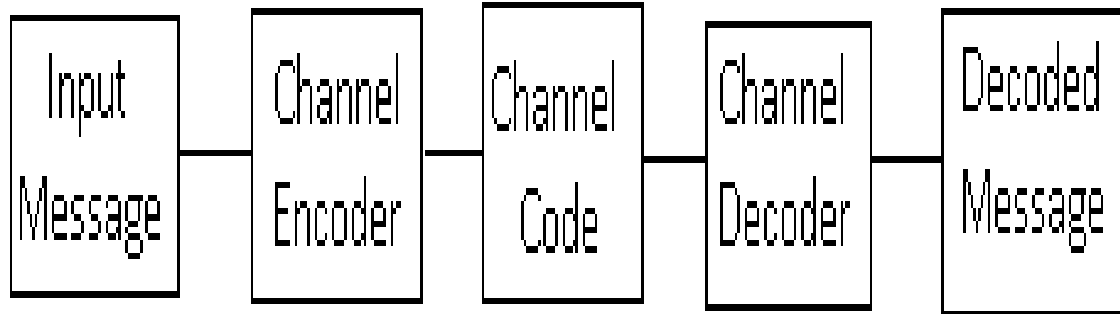
RESULT:

Hence errors in the transmission channel can be detected and corrected with linear block coding In (7,4) hamming code, single bit errors in any position are detected & corrected and double errors are only detected but not corrected.

VIVA QUESTIONS:

1. What is the purpose of channel coding?
2. What are the advantages of channel coding?
3. What are the different channel coding techniques used?
4. After channel coding, message is in which form?
5. Define Shannon's channel coding theorem.
6. How many errors can be detected and corrected using Linear Block Codes?
7. What is the decoding technique used for Linear Block Codes for getting original message?

BLOCK DIAGRAM OF BINARY CYCLIC CODES ENCODER AND DECODER:



EXPT. NO:

DATE:

BINARY CYCLIC CODES ENCODER AND DECODER

AIM:

To observe that the errors received through a noisy channel can be removed by employing the error detection and correction code by using algebraic structure.

APPARATUS:

1. Binary cyclic codes encoder and decoder trainer kit
2. Adapter

THEORY:

An (n, k) linear code C is called a cyclic code if every cyclic shift of a code vector in C is also a code vector in C . In coding theory cyclic codes are the subclass of linear block error correcting codes that have convenient algebraic structures for efficient error detection and correction. A linear code is called cyclic code if every cyclic shift of code vector produces some other code vector i.e. the cyclic shift to the data in an array should also represent the data in the same array. Example: array $\{(0000) (0101) (1010) (1111)\}$

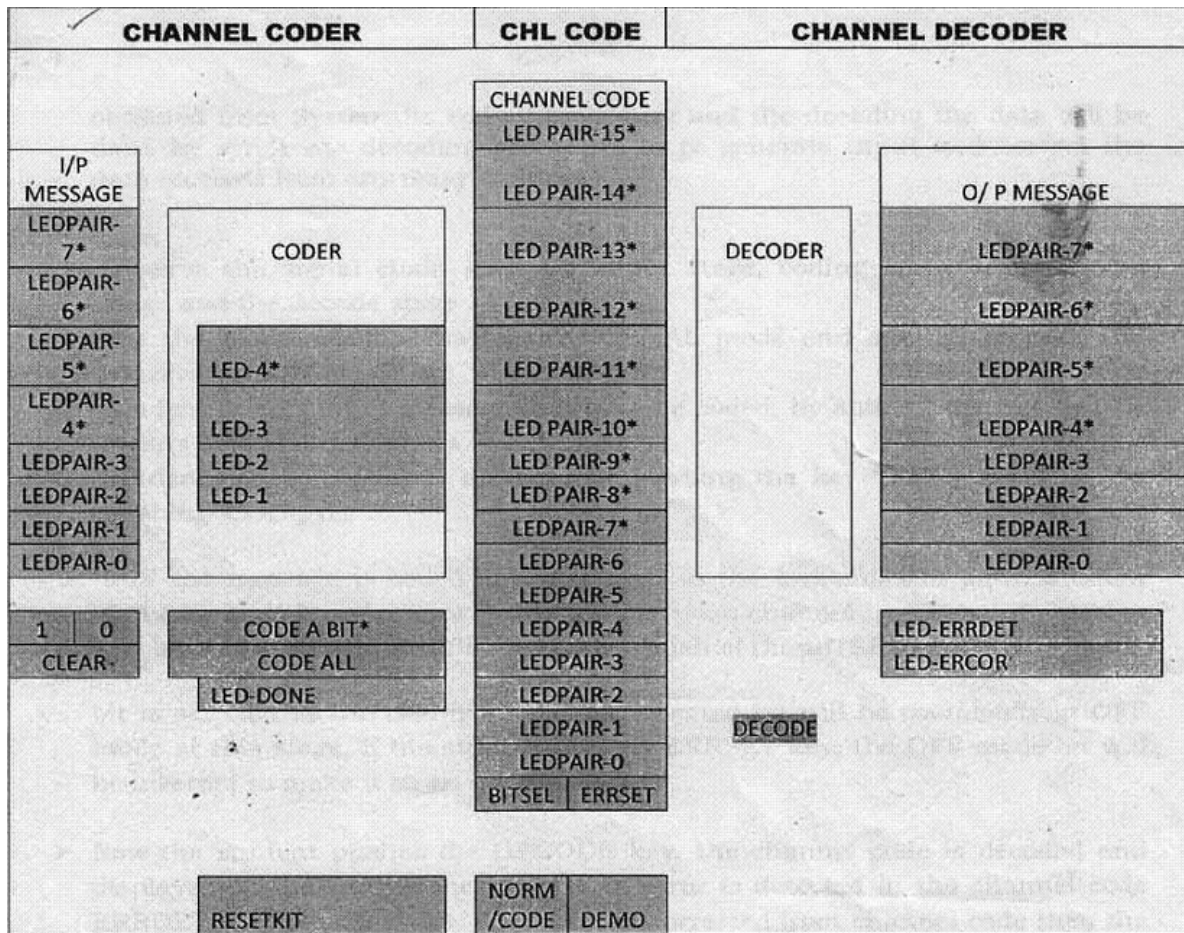
Cyclic codes form an important subclass of linear codes. These codes are attractive for two reasons: first, encoding and syndrome computation can be implemented easily by employing Shift registers with feedback connections (or linear sequential circuits); and second, because they have considerable inherent algebraic structure

Cyclic codes are of interest and importance because

- They possess rich algebraic structure that can be utilized in a variety of ways.
- They have extremely concise specifications.
- They can be efficiently implemented using simple shift registers.
- Many practically important codes are cyclic.

Error detection in cyclic codes is simpler but error correction is little complicated since the combination logic circuits in error detector are complex

EXPERIMENT PROCEDURE:



OBSERVATIONS TABLE:

Type of error	4-bit Input message(abcd)	3-Parity bits(xyz)	Channel Code(7bit) (abcdxyz)	Decoder Input	Decoder Output	Error detected	Error corrected
No error	1010					-----	-----
Single error	1000					Yes/No	Yes/No
Double error	0111					Yes/No	Yes/No

PROCEDURE:

1. Observe the signal chain, i.e. the input stage, coding stage, transmission stage and the decode Stage.
2. Put the mode selection switch in NORMAL/CODE mode and see the process and observe Output.
3. Now select input message that is to be coded, by shifting the bits 0/1 by means of pressing the Keys 0, 1, CLEAR.
4. Now the parity bits are added for the given input message by pushing the key 'CODE A BIT' Or by pushing 'CODEALL'.
5. Now the message is coded and displayed in the transmission path. Now push the DECODE Key, the channel code is decoded and displayed as the output message having no errors.
6. Due to channel noise, errors are introduced in the channel, now errors are introduced by means of pressing the keys BITSEL and ERRSET. On every push of the BITSEL one bit is Selected in the channel code, now if ERRSET is pressed the existing bit is inverted to make it As an error.
7. Now again DECODE key is pressed to display output message. If an error is detected in the Channel code ERRDETECTED LED glows, if an error is corrected from the channel code ERRCORRECTED LED glows in the decoder output stage.
8. Observe the decoded output by introducing single and double errors in the channel code.
9. Note down the observations in a table given.

PRECAUTIONS:

- 1 .Avoid loose Connections.
2. Switch off the power supply during connections
3. Waveforms must be noted carefully.

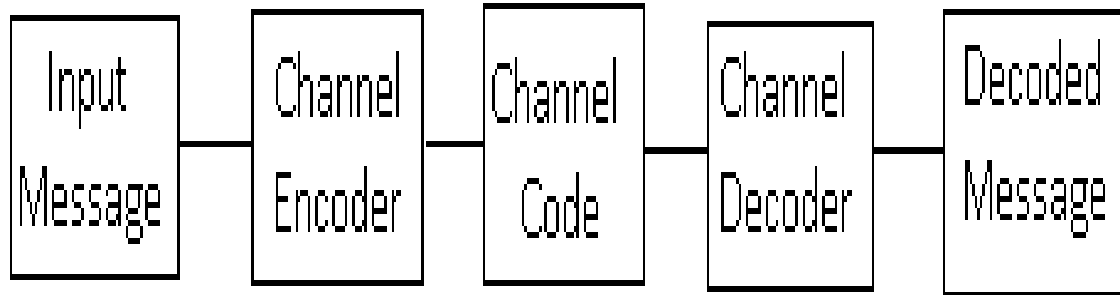
RESULT:

Hence errors in the transmission channel can be detected and corrected with cyclic coding i.e., single bit errors in any position are detected & corrected and double errors are only detected but not corrected.

VIVA QUESTIONS:

1. What is the purpose of channel coding?
2. What are the advantages of channel coding?
3. What are the different channel coding techniques used?
4. After channel coding, message is in which form?
5. Define Shannon's channel coding theorem.
6. How many errors can be detected and corrected using Binary Cyclic Codes?
7. What is the decoding technique used for Binary Cyclic Codes for getting original message?

BLOCK DIAGRAM OF CONVOLUTION CODES ENCODER AND DECODER:



EXPT. NO:

DATE:

CONVOLUTION CODES ENCODER AND DECODER

AIM:

To observe error correcting performance and decoding ability of convolution codes.

APPARATUS:

1. Convolution codes encoder and decoder trainer kit
2. Adapter

THEORY:

Convolution codes are one of the type of channel coding techniques used for the purpose of detecting and correcting errors, occurring in the communication channel. Convolution coding is a special case of error control coding. Unlike a block coder, convolution code is a memory based device. Even though a convolution encoder accepts a fixed no. of message symbols and produces a fixed no. of code symbols, its computations depend not only on the current set of input symbols but on some of the previous input symbols.

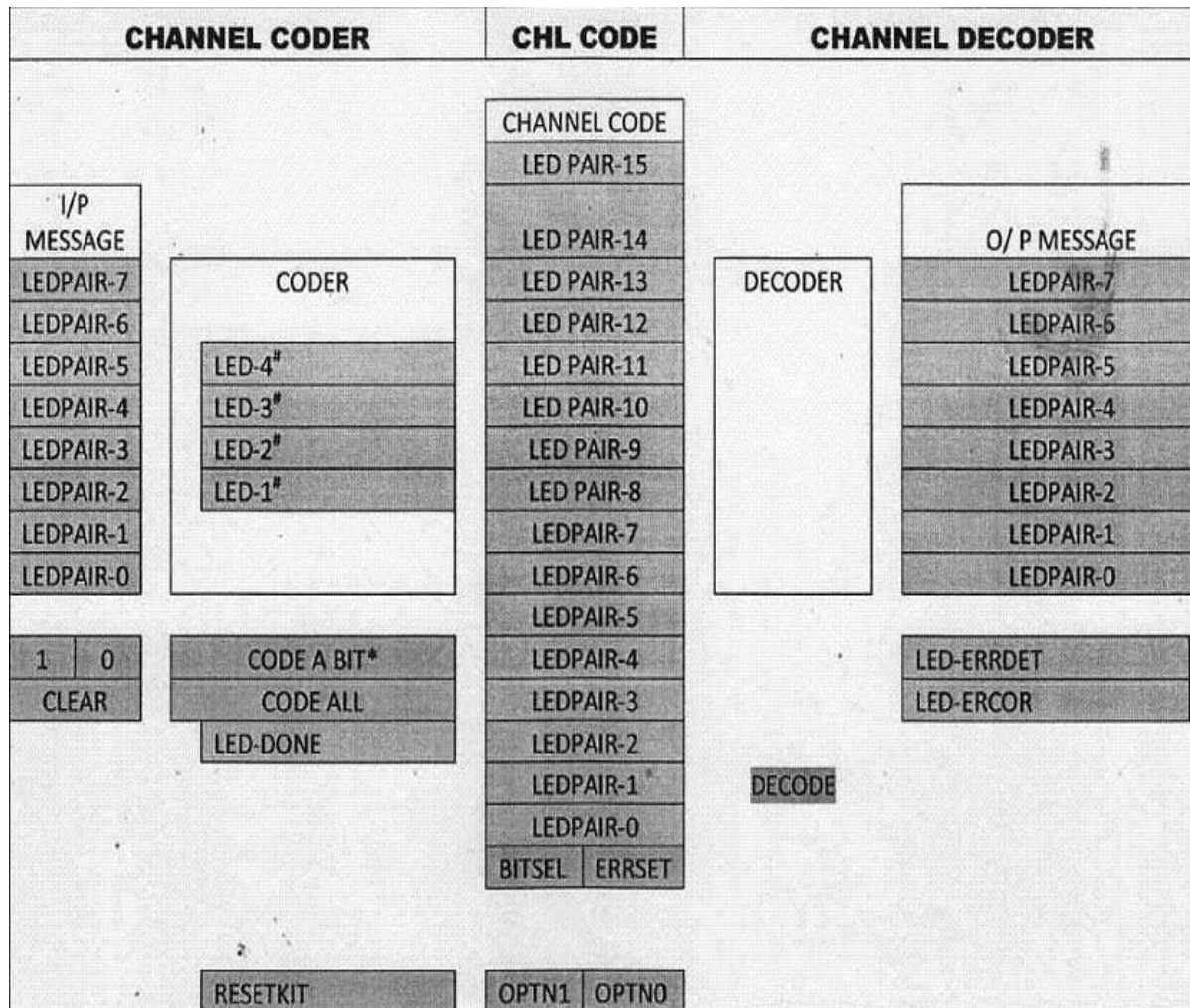
Convolution code normally represented as $(n, m, \text{ and } k)$ where

n = no. of output bits , m = no. of message bits , k =no. of constraint bits/memory bits

Convolution codes are decoded by using Viterbi algorithm.

Convolutional codes are commonly described using two parameters: the code rate and the constraint length. The code rate, k/n , is expressed as a ratio of the number of bits into the convolutional encoder (k) to the number of channel symbols output by the convolutional encoder (n) in a given encoder cycle. The constraint length parameter, K , denotes the "length" of the convolutional encoder, i.e. how many k -bit stages are available to feed the combinatorial logic that produces the output symbols. Closely related to K is the parameter m , which indicates how many encoder cycles an input bit is retained and used for encoding after it first appears at the input to the convolutional encoder. The m parameter can be thought of as the memory length of the encoder. Convolutional codes are widely used as channel codes in practical communication

EXPERIMENT PROCEDURE:



OBSERVATIONS TABLE:

Type of error	4-bit Input message	Change d input (6-bit)	Channel Code(12bit)	Decoder Input	Output Message (6-bit)	4-bit Output message	Error detected	Error corrected
No error	1010	101000	111101111000	111101111000	101000	1010	----	-----
Single error	1000	100000					Yes/No	Yes/No
Double error	0111	011100					Yes/No	Yes/No

systems for error correction. The encoded bits depend on the current k input bits and a Trellis coded modulation (TCM) and turbo codes are two such examples. In TCM, redundancy is added by combining coding and modulation into a single operation. This is achieved without any reduction in data rate or expansion in bandwidth as required by only error correcting coding schemes.

Basic concepts of Convolution Codes

1. State Diagram Representation
2. Tree Diagram Representation
3. Trellis Diagram Representation

PROCEDURE:

1. Observe the signal chain, i.e. the input stage, coding stage, transmission stage and the decode Stage.
2. Put the mode selection switch in NORMAL/CODE mode and see the process and observe Output.
3. Now select input message that is to be coded, by shifting the bits 0/1 by means of pressing to Keys 0, 1, CLEAR.
4. Now the parity bits are added for the given input message by pushing the key 'CODE A BIT' Or by pushing 'CODEALL'.
5. Now the message is coded and displayed in the transmission path. Now push the DECODE Key, the channel code is decoded and displayed as the output message having no errors.
6. Due to channel noise, errors are introduced in the channel, now errors are introduced by Means of pressing the keys BITSEL and ERRSET. On every push of the BITSEL one bit is Selected in the channel code, now if ERRSET is pressed the existing bit is inverted to make it As an error.
7. Now again DECODE key is pressed to display output message. If an error is detected in the Channel code ERRDETECTED LED glows, if an error is corrected from the channel code ERRCORRECTED LED glows in the decoder output stage.
8. Observe the decoded output by introducing single and double errors in the channel code.
9. Note down the observations in a table given.

PRECAUTIONS:

- 1 .Avoid loose Connections.
2. Switch off the power supply during connections
3. Waveforms must be noted carefully.

RESULT:

Hence errors in the transmission channel can be detected and corrected in the receiver using convolution coding and viterbi decoding.

VIVA QUESTIONS:

1. What is the purpose of channel coding?
2. What are the advantages of channel coding?
3. What are the different channel coding techniques used?
4. After channel coding, message is in which form?
5. Define Shannon's channel coding theorem.
6. How many errors can be detected and corrected using Convolution Codes?
7. What is the decoding technique used for convolution codes for getting original message?