EXPERIMENT NO. 1

Aim:

AM Generation (DSB-FC): Calculation of modulation index by graphical method, Power of AM Wave for different modulating signal and Observe Spectrum.

Objectives:

1. To understand generation & Detection of DSB-FC AM signal .
2. To study trapezoidal method for calculation of modulation index.
3. To understand power measurement of sidebands of DSB-FC AM Signal using Spectrum analyser

Apparatus:

DSB-FC generation kit , CRO, Spectrum Analyzer.

Theory:

Modulation is the act of translating some low frequency or baseband signal (voice, music, and data) to a higher frequency. Why do we modulate signals? There are at least two reasons: to allow the simultaneous transmission of two or more baseband signals by translating them to different frequencies, and to take advantage of the greater efficiency and smaller size of higher-frequency antenna.

In the modulation process, some characteristic of a high-frequency sinusoidal carrier is changed in direct proportion to the instantaneous amplitude of the baseband signal. The carrier itself can be described by the equation:

$$e = A \cos (\omega t + \varphi)$$

Where;

A = peak amplitude of the carrier,

ω = angular frequency of the carrier in radians per second,

t = time, and

φ = initial phase of the carrier at time $t = 0$.

In the expression above, there are two properties of the carrier that can be changed, the amplitude (A) and the angular position (argument of the cosine function). Thus we have amplitude

- **Calculations** ⇒.

- For Graphical Method:

1] $MI = \frac{V_{max} - V_{min}}{V_{max} + V_{min}} \times 100 = \frac{3-1}{3+1}$

$= 50\%$.

2] $MI = \frac{V_{max} - V_{min}}{V_{max} + V_{min}} \times 100 = \frac{3.7 - 1.4}{3.7 + 1.4} \times 100$

$= 45\%$.

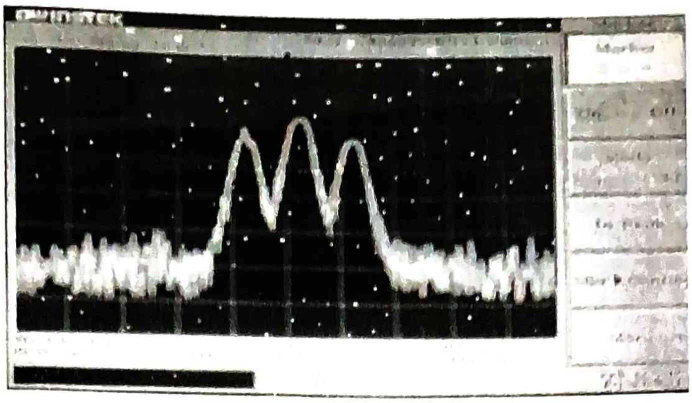
- For Trapezoidal Method:

1] $MI = \frac{L_1 - L_2}{L_1 + L_2} \times 100 = \frac{3 - 1.1}{3 + 1.1} \times 100$

$= 46.34$.

2] $MI = \frac{L_1 - L_2}{L_1 + L_2} \times 100 = \frac{6.8 - 3.2}{6.8 + 3.2} \times 100$

$MI = 36\%$.



4. To calculate power of Carrier and sidebands, measure amplitude of markers on spectrum.

Observation Table

Sr. No.	Test points	Frequency(Hz)	Voltage(V)
1	Modulating signal	1.5 kHz	3×0.5 = 1.5V
2	Carrier signal	350 kHz	4.4×0.5 = 2.5V
3	Modulated signal	352 kHz	3×0.5 = 1.5V
4	Demodulated Signal	1.4 kHz.	16×0.5 = 8V.

Graphical Methods

	Test points	Graphical Display		Modulation Index
Sr. No.		Voltage(V)		
1	Vmax (Maximum Modulating voltage)	3	1	50%
2	Vmin (Minimum Modulating voltage)	3.7	1.4	45%.

Trapezoidal Display

	Test points	Trapezoidal Display		Modulation Index
Sr. No.		Voltage(V)		
1	L1	6.8	3	36%.
2	L2	3.2	1.1	46.34

Sr. No.	Parameter	Graphical Display	Trapezoidal display
1	Modulation Index	45%	36%.

Calculation:

$$\text{Modulation index} = \frac{V_{max} - V_{min}}{V_{max} + V_{min}} \times 100$$

Spectrum Observation Table:

Sr. No.	Test points	Frequency	Power in dbuv
1	Carrier	434 kHz	99.0 dBuV
2	USB	461 kHz.	66.9 dBuV
3	LSB	435 kHz	66.9 dBuV

Conclusion:

Observed FM Waveform by giving modulating signal & carrier signal to modulator optitude moduted waveform is observed.

Questions:

1. Define Modulation? Need of Modulation?
2. Write mathematical expression for AM?
3. Application of AM?
4. Difference between DSB-FC,DSB-SC,SSB,VSB?
5. Advantages and disadvantages of AM?
6. Draw waveforms and spectrum of AM on graph paper?

18/2/25

Experiment No. 1

1] define modulation? Need of Modulation.

⇒ Modulation is defined as, the process in which some parameter of carrier wave is varied in proportion with the instantaneous magnitude of the modulation signal.

Need:

1) Reduction in height of antenna.
2) Avoids mixing of signals.
3) Increases the range of Communication.
4) Multiplexing become possible.
5) Improves quality of Reception.

2] Write mathematical expression for AM?

⇒ $e_{AM} = E_c [1 + m \cdot \cos(2\pi f_m \cdot t)] \cdot \cos(2\pi f_c t)$.

3] Application of FM.

⇒ 1) Radio broadcasting.
2) Picture transmission in a TV system.

4] Advantages & disadvantages of AM?

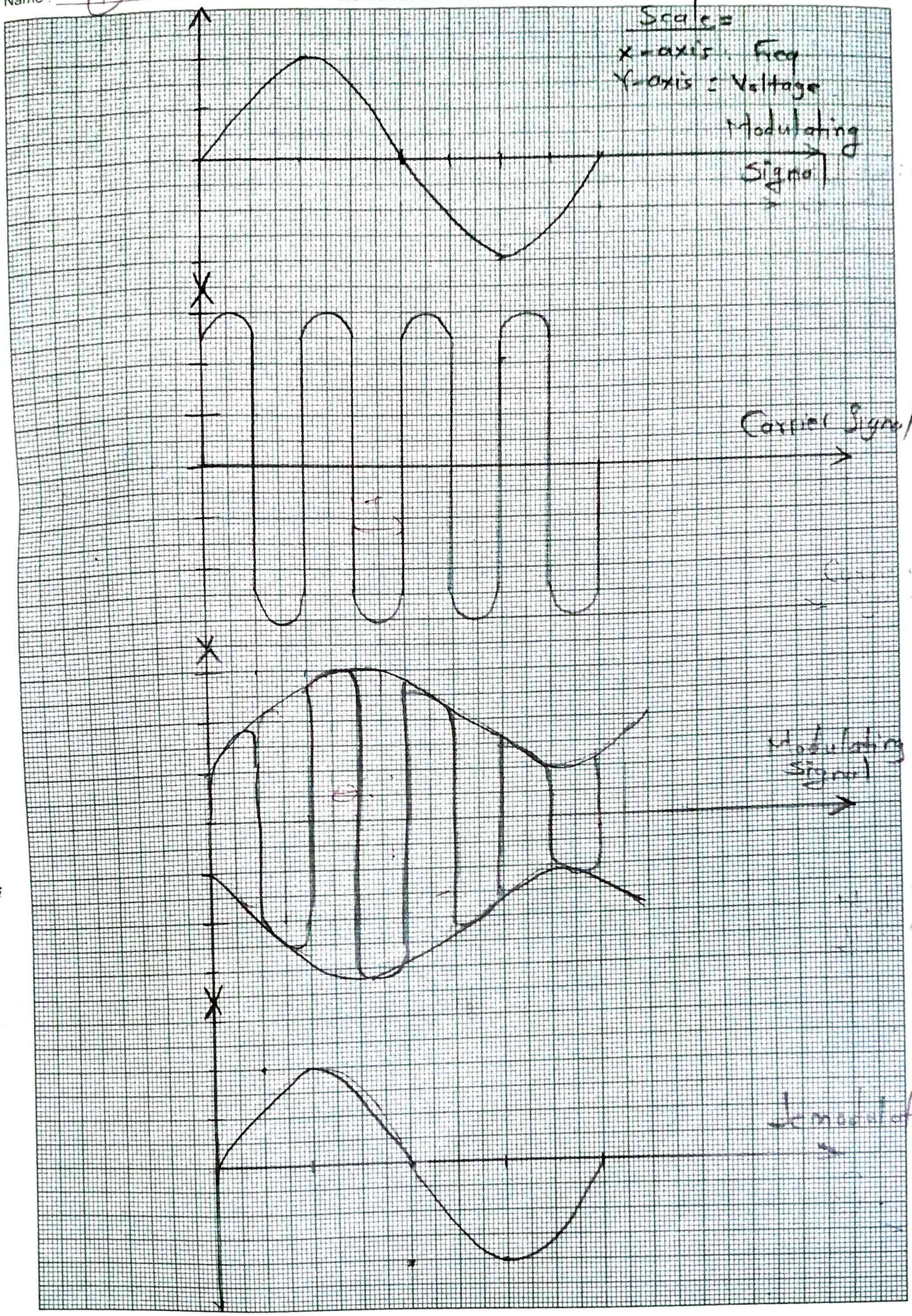
⇒

Advantages	Disadvantages
1] Less Complex.	1] Power wastage take place
2] Simple & detection is easy	2] Needs larger bandwidth.
3] Cost efficient	3] Get affected due to noise.
4] AM waves can travel longer distance.	
5] Low bandwidth.	

5) Difference between DSB-FC, SC, VSB, SSB?

⟹

	DSB-FC.	DSB-SC	VSB	SSB
Carrier Transmission.	NA.	Fully	Fully	NA.
Sidebands present.	NA	NA.	1 S.B. Completely	1 Full + One part
Bandwidth.	2fm	2fm	fm.	fm.
Transmission.	minimum	moderate	maximum	maximum
No. of Modulators.	1	1	1	1.
Applications.	Radio broad-casting.	Radio broadcast-ing	Point to Point mobile comm.	TV Broadca-sting.


Scale =
x-axis : Freq
Y-axis = Voltage
Modulating Signal
Carrier Signal
Modulating Signal
demodulation

EXPERIMENT NO. 2

Aim:

Frequency modulator & demodulator using Varicap/Varactor Diode and NE 566 VCO, IC 565 (PLL based detection), calculation of modulation index & BW of FM.

Objectives:

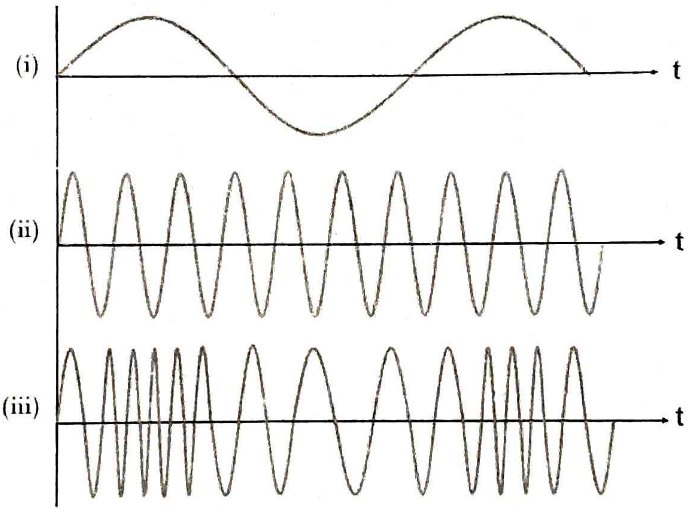
1. To understand generation of frequency modulated Signal.
2. To understand demodulation of FM.
3. Calculation of Modulation Index & BW of FM.

Apparatus:

FM generation & Detection using Varicap/Varactor Diode and NE 566 VCO, IC 565 (PLL based detection) kit , CRO, Spectrum Analyzer.

Theory

A category of angle modulation in which the frequency of the carrier wave is changed according to the amplitude of the message signal is known as frequency modulation. It is abbreviated as FM and is a widely used analog modulation technique.



(i) Modulating signal
(ii) Carrier waveform
(iii) Frequency modulated signal

Single tone frequency modulation

Let the modulating signal be m(t) having amplitude V_m and frequency f_m

१०

FM Rest Frequency and Sideband Nulls

As mentioned before, an interesting aspect of FM is that for certain values of modulation index the rest-frequency component of the FM wave can disappear! See Table 4-4, Column 2 for Bessel-function "zeros" of the rest frequency. A term given to the m_f zero values is *eigenvalues*.

TABLE 4-4 Zeros of the Bessel Functions

Number of *Zero*	$J_0 (m_f)$	$J_1 (m_f)$	$J_2 (m_f)$	$J_3 (m_f)$
0	2.41	3.83	5.14	6.38
1	5.53	7.00	8.42	9.76
2	8.65	10.17	11.62	13.02
3	11.79	13.32	14.80	16.22
4	14.93	16.47	17.96	19.41

5. Now calculate modulation index as per Eq. (1).

Modulating index for 1st Zero = 0.65 x37/10 = 2.40

Modulating index for 2nd Zero = 1.50 x37/10 = 5.55

Modulating index for 3rd Zero = 2.35 x37/10 = 8.69

Modulating index for 4th Zero = 3.20 x37/10 = 11.84

Now compare measured and theoretical Eigen values as per above table. These are exactly same.

Observation Table

Sr. No.	Test points	Frequency(Hz)	Voltage(V)
1	**Modulating Signal**	10.10 kHz	1.84V
2	**Carrier Signal**	94.79 kHz	4.80V
3	**Demodulator Output**	9.48 kHz.	5.52 mV
4	**LPF Output**	122.7 kHz.	1.02 mV
5		160.6 kHz.	4.76V.

Sr. No.	Modulating Voltage (V)		Frequency (kHz).	Power dBmV
1	0V	Carrier	147.	17
2	0.2	Carrier	148	17
		USB1	156	7
		LSB1	136	6
3	0.5	Carrier	148	17
		USB1	156	17
		LSB1	136	16.8
		USB2	166	5
		LSB2	127	3
4	0.8	Carrier	145	17
		USB1	157	17
		LSB1	127	9.67
		USB2	167	10
		LSB2	176	-1.5
		USB3	138	17
		LSB3	116	-2.5.

Conclusion:

In frequency modulation, frequency of carrier signal changes in accordance with the modulating signal also observed the eigen value of ~~theorem~~ FM in frequency domain.

Questions:

1. What is Frequency Modulation?
2. Which modulator and demodulator we are used for FM generation and reception?
3. Explain the function of PLL FM demodulator?
4. Advantages of FM over AM?
5. Compare AM and FM?
6. Application of FM?
7. Explain Bessel's Function?
8. Explain Narrowband and Wideband FM?

* Experiment No.2 *

1] Define FM modulation

⇒ FM modulation in which the instantaneous frequency of the carrier is varied in proportion with instantaneous amplitude of the modulating signal.

2] Explain the fxn of PLL FM deModulator.

⇒ A phase locked loop basically a -ve FB system, it consists of 3 major components of 3 major components such as re-multiplier a loop filter and a vtg controlled oscillator connected together in the form of a FB loop.

A phase locked loop is primarily used in tracking the phase & freq. of the carrier component of an incoming FFT signal.

PLL is also used for synchronous demodulation of AM-SC signals or signals with few cycles of pilot carries.

PLL is also useful for demodulating FM signal in presence of large noise and low signal power.

3] Advantages of FM over AM.

⇒ 1] Better Noise Immunity.

2] Rejection of Interference.

3] Higher Transmission efficiency.

4] Lower power requirements.

5] Larger area.

6] All the transmitted power is useful.

3] Compare AM and FM.

	Parameters.	AM.	FM.
1]	Variable parameter of the Carrier.	Amplitude.	Frequency
2]	Mathematical eqn.	$S(t) = E_c \sin(\omega_c t + m_F \sin\omega_m t)$	$S(t) = A_c \cos[2\pi f_c t + 2\pi k_f \int m$
3]	Amplitude of Modulating signal.	Varies continuously.	~~Const~~ Affects Freq. deviation
4]	Bandwidth.	Constant $2F_m$	$2(\Delta F + F_m)$

Name :

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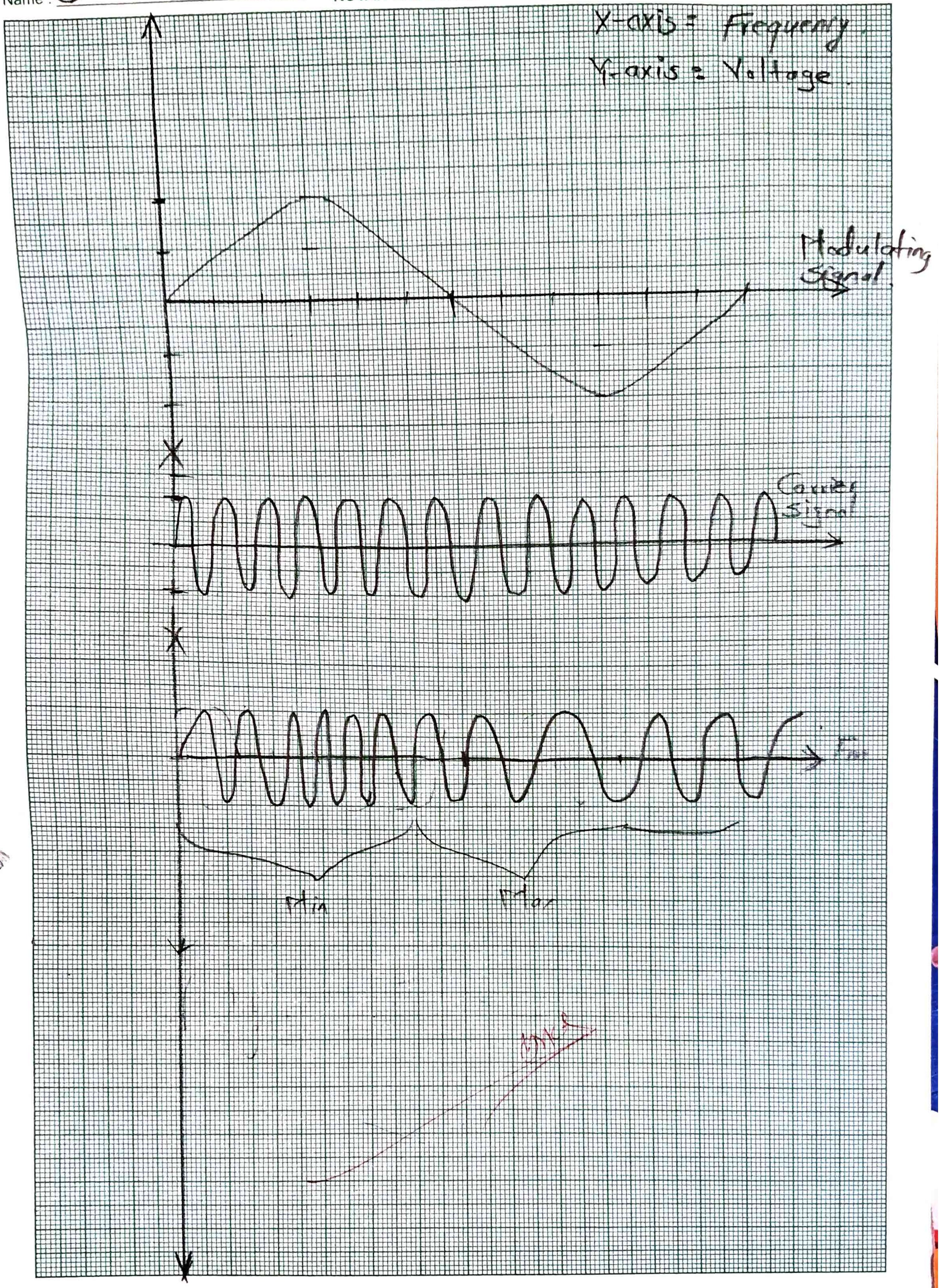


EXPERIMENT NO. 3

Aim:

Verification of Sampling Theorem, PAM Techniques, (Flat top & Natural sampling), reconstruction of original signal, Observe Aliasing Effect in frequency domain.

Objectives:

1. To study verification of Sampling Theorem
2. To study PAM Techniques (Flat top & Natural sampling)
3. To reconstruct original signal using Interpolation filter
4. To study aliasing effect in frequency domain
5. To observe Spectrum of Sampled signal on Spectrum analyzer
6. How to overcome aliasing effect?

Apparatus:

Sampling theorem kit , CRO, Spectrum Analyzer.

Theory:

Pulse Amplitude Modulation & Demodulation(PAM):

When a perfectly rectangular pulse waveform is transformed from the time domain to the frequency domain (Figure-1), the resulting envelope follows a function of the form:

$$Y = \frac{\text{Sin } X}{X}$$

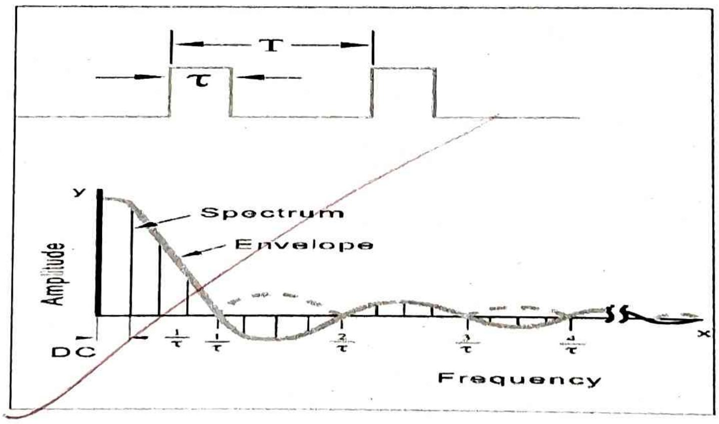


Figure - 1

Observation Table:

Sr. No.	Test points	Frequency(Hz) (Hz)	Voltage(V) (V)
1	Sine wave generator output	258.7	2.38
2	Sampling Pulse generator output	6.91	4.88
3	Output of sampler(Natural)	367.3	2.38
4	Output of sampler(Flat top)	1.786	2.32
5	Output of demodulator (LPF)	250.6.	1.90.

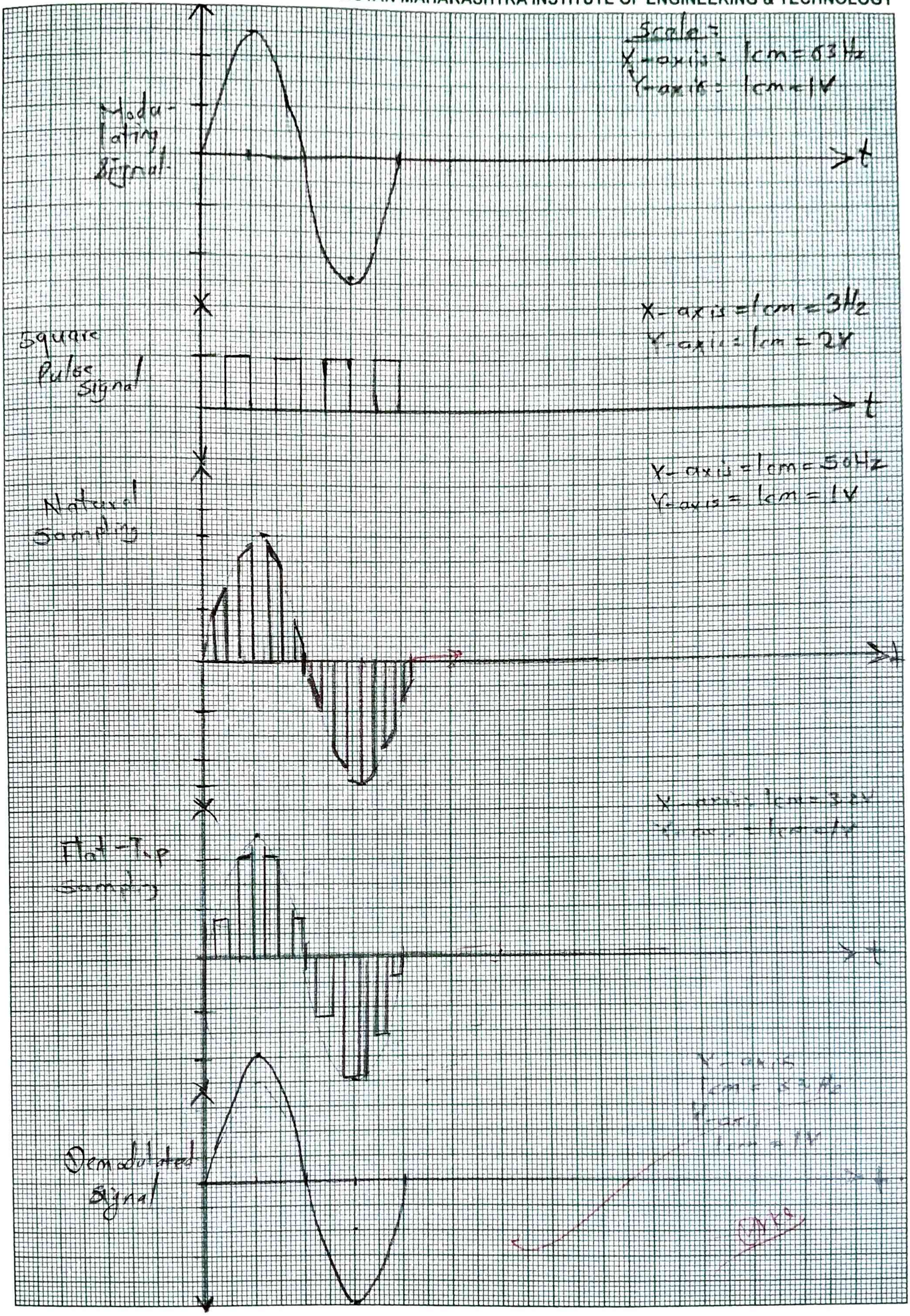
Conclusion:

Thus, we observed that the sampling of natural sampling & aliasing affect with the nyquist rate of $F_S \geq 2F_M$.

Oral Questions:

1. State the sampling Theorem?
2. What is Nyquest criteria and Nyquest Interval?
3. What is Aliasing Effect?
4. What are the different types of sampling technique
5. Difference between natural sampling and flat top sampling
6. Difference between impulse. natural, flat top sampling?
7. Draw the waveforms of natural sampling and flat top sampling?
8. Draw the diagram of generation of natural sampling and flat top sampling?
9. What is the solution for aliasing effect?
10. What is the sampling process?

Name : 3


Scale:-
X-axis:- 1cm = 3Hz
Y-axis:- 1cm = 1V
Modulating Signal
t
X-axis = 1cm = 3Hz
Y-axis = 1cm = 2V
Square Pulse Signal
t
X-axis = 1cm = 50Hz
Y-axis = 1cm = 1V
Natural Sampling
Flat-Top Sampling
Demodulated Signal

Experiment No. 3.

1) State the Sampling theorem.

⟹ Let, m(t) be a signal which is band limited such that its highest Freq. component is f_m, let the values of m(t) be determined at regular intervals separated by times $T_s \leq 1/2 f_m$ where m is an integers uniquely determine the signal & the signal may be reconstructed from these samples with no distortion.

2) What Nyquist Criteria?

⟹ The Nyquist Criteria requires that the Sampling Freq. be at least twice the highest Freq. contained in the signal or along signal is sampled at discrete interval $t_s = \frac{1}{f_s}$ which must be carefully choosen to ensure an accurate repre. of original Analog signal.

3) What Aliasing effect?

⟹ It's an undesirable effect that is seen in sampled system when the system i/p Frequency is greater than half the sample Frequency & the sample points don't adequately present the i/p signal. I/P at the there higher Freq. are observed at a lower wiared Freq.

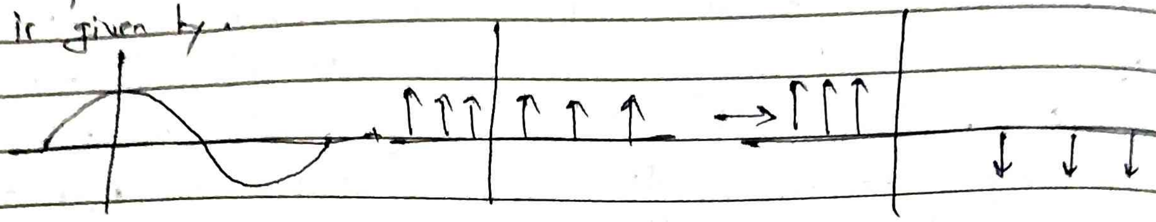
4) What are diffe types of Sampling techniques?

⟹ 1] Impulse sampling.
2] Natural Sampling.
3] Flat Top Sampling.

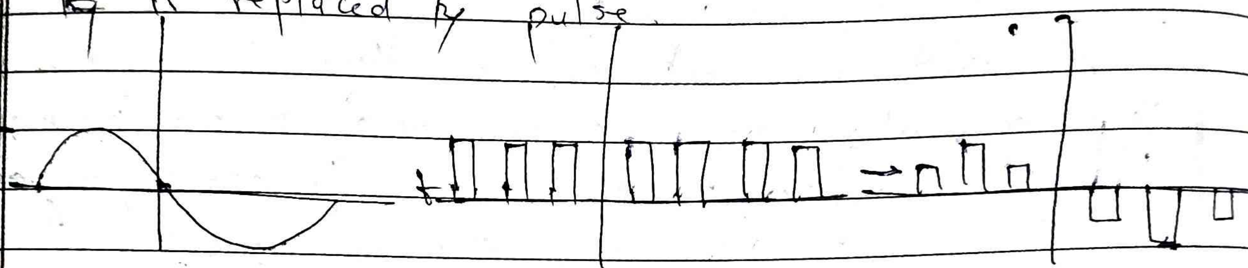
1] Impulse Sampling ⟹ It can be performed by multiplying i/p signal x(t) width $\sum_{n=-\infty}^{\infty} \delta(t-nT)$ of T.

NMIET

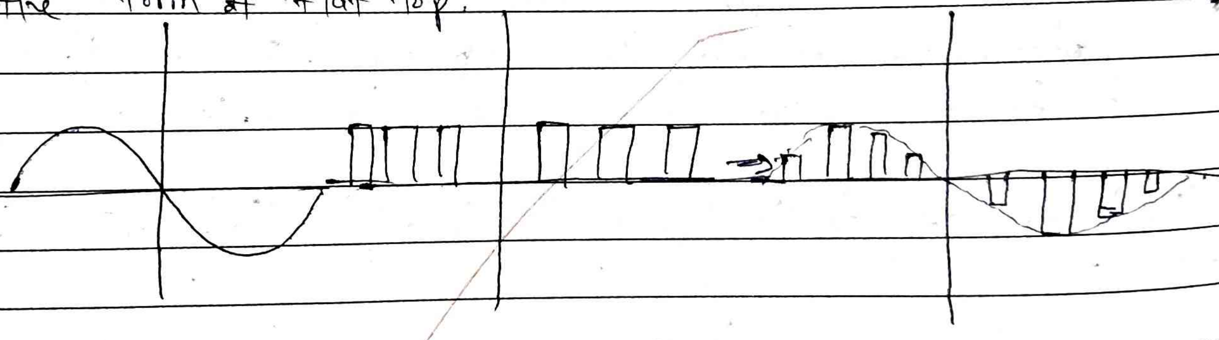
Hence the amplitude n=∞ of impulse changes with response amplitude of i/p signal x(t) the o/p sample is given by.



[2] Natural Sampling ⇒
Similar to impulse sampling except the impulse train is replaced by pulse.



[3] Flat-top Sampling ⇒
Noise is introduced at the top of the transmission pulse which can be easily removed if the pulse is in the form of Flat top.



EXPERIMENT NO.4

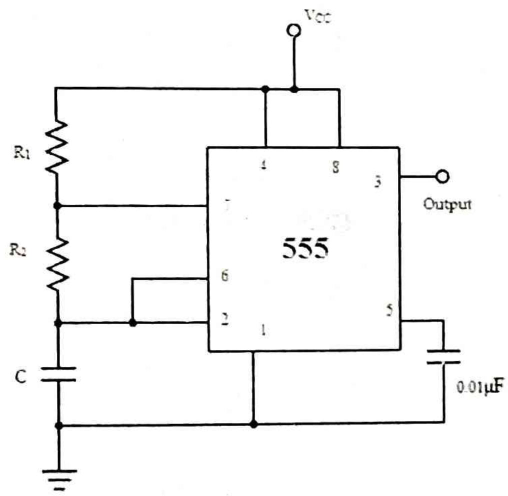
AIM: Generation and Detection of PWM using IC 555

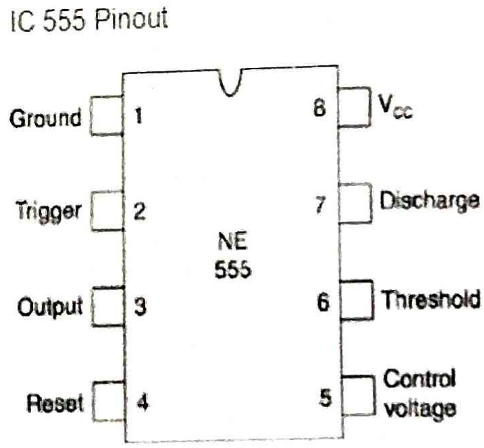
OBJECTIVE: Study the PWM Generation & Detection using IC 555.

APPARATUS/SW: Trainer kit of PWM, DSO.

THEORY:

PWM (Pulse Width Modulation) is one of the modulation techniques in which the width of the carrier wave varies with the amplitude of the message signal. In this technique the pulse is used as a carrier signal and the message signal can be any analog signal. Hence as the width is changing it can be used to control the power given to the devices. Thus the major application of PWM is to control the power given to the electrical appliances like motors.
IC 555 WORKS as an Astable Multivibrator is an oscillating circuit without a stable state i.e., it automatically switches between the two states. Hence, an Astable Multivibrator is also known as Free Running Multivibrator or Free Running Oscillator. Using just additional three components, we can make the 555 Timer to work in Astable Mode. They are a couple of resistors and a capacitor.





Astable multivibrator is also called as Free Running Multivibrator. It has no stable states and continuously switches between the two states without application of any external trigger. The IC 555 can be made to work as an astable multivibrator with the addition of three external components: two resistors (R1 and R2) and a capacitor (C). The schematic of the IC 555 as an astable multivibrator along with the three external components is shown below.
The pins 2 and 6 are connected and hence there is no need for an external trigger pulse. It will self trigger and act as a free running multivibrator (oscillator). The rest of the connections are as follows: pin 8 is connected to supply voltage (VCC). Pin 3 is the output terminal and hence the output is available at this pin. Pin 4 is the external reset pin. A momentary low on this pin will reset the timer. Hence, when not in use, pin 4 is usually tied to VCC.

Test points	Voltage	Frequency
1) Modulating Signal Output.	1.52 V	805.8 Hz
2) Carrier Pulse O/P.	3.68 V	1.28 kHz.
3) PWM Generator.	3.84.	1.80 kHz.

Conclusion:

In this expt, by using trainer kit of pulse width modulation we performed practical on generation of detection of pulse modulation.

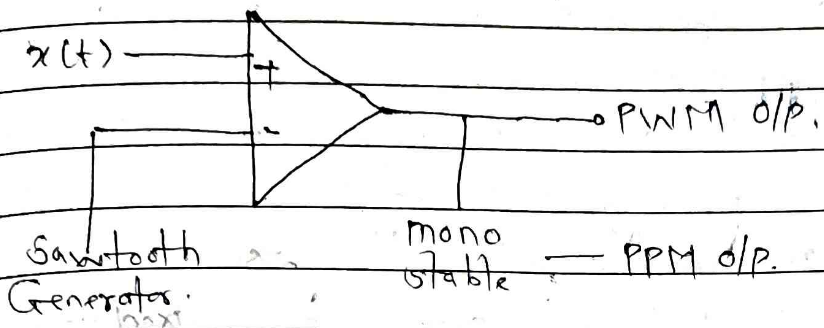
Oral Questions:

1. What is mean by Pulse Modulation?
2. What are the different types of Pulse Modulation?
3. Explain PAM with Waveforms?
4. Explain PWM with waveforms and generation diagram?
5. Compare PAM, PWM, PPM?
6. What is Nyquest Criteria?
7. Explain IC 555
8. Application of PWM
9. Advantages and Disadvantages of PWM
10. Compare Analog Digital and Pulse Modulation

Experiment No. 4

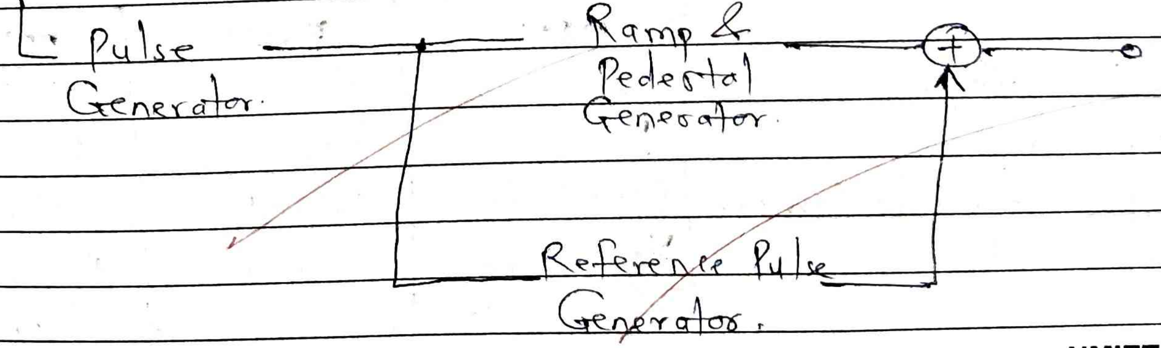
Q.1) Draw & explain the PWM Generation & Reception.

⟹ Generation of PWM Signal ⟹.



- A sawtooth generater a sawtooth signal of freq. Fs, therefore the sawtooth signal is this case is a sampling signal.
- It's applied to the inverting terminal of a Comparator.
- The modulating signal x(t) is applied to the Non-Inverting terminal of the same comparator.
- The comparator O/P will remain high as long as the instantaneous amplitude of x(t) is higher than that of the ramp signal.
- Note that leading edges of a PWM waveform coincide with the falling edges of ramp signal.
- Thus, leading edges of PWM signal are always generated at fixed time constant.

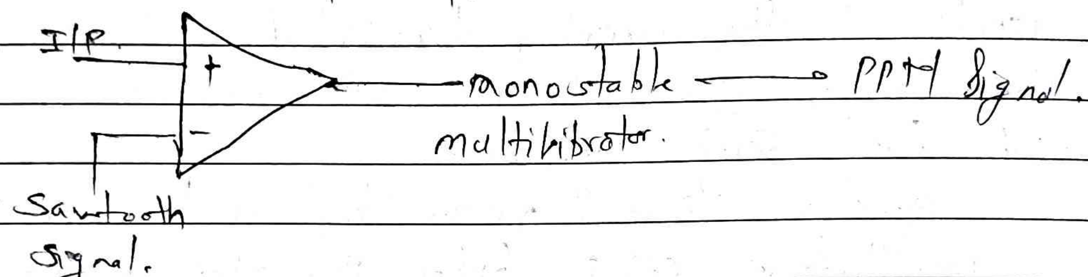
Reception of PWM =



- The PWM signal received at the i/p detect ckt is contaminates with 'n'.
- This signal is applied to pulse generator ckt which regenerates the PWM signal.
- Thus some of the Noise is removed pulse are squared up.
- The regenerated pulse are applied to reference pulse generator.
- It produces ~~are~~ synchronized to the example of the regenerated PWM pulses delayed by a fixed interval.

Q.2 Draw & Explain PPM Generator.

⟹

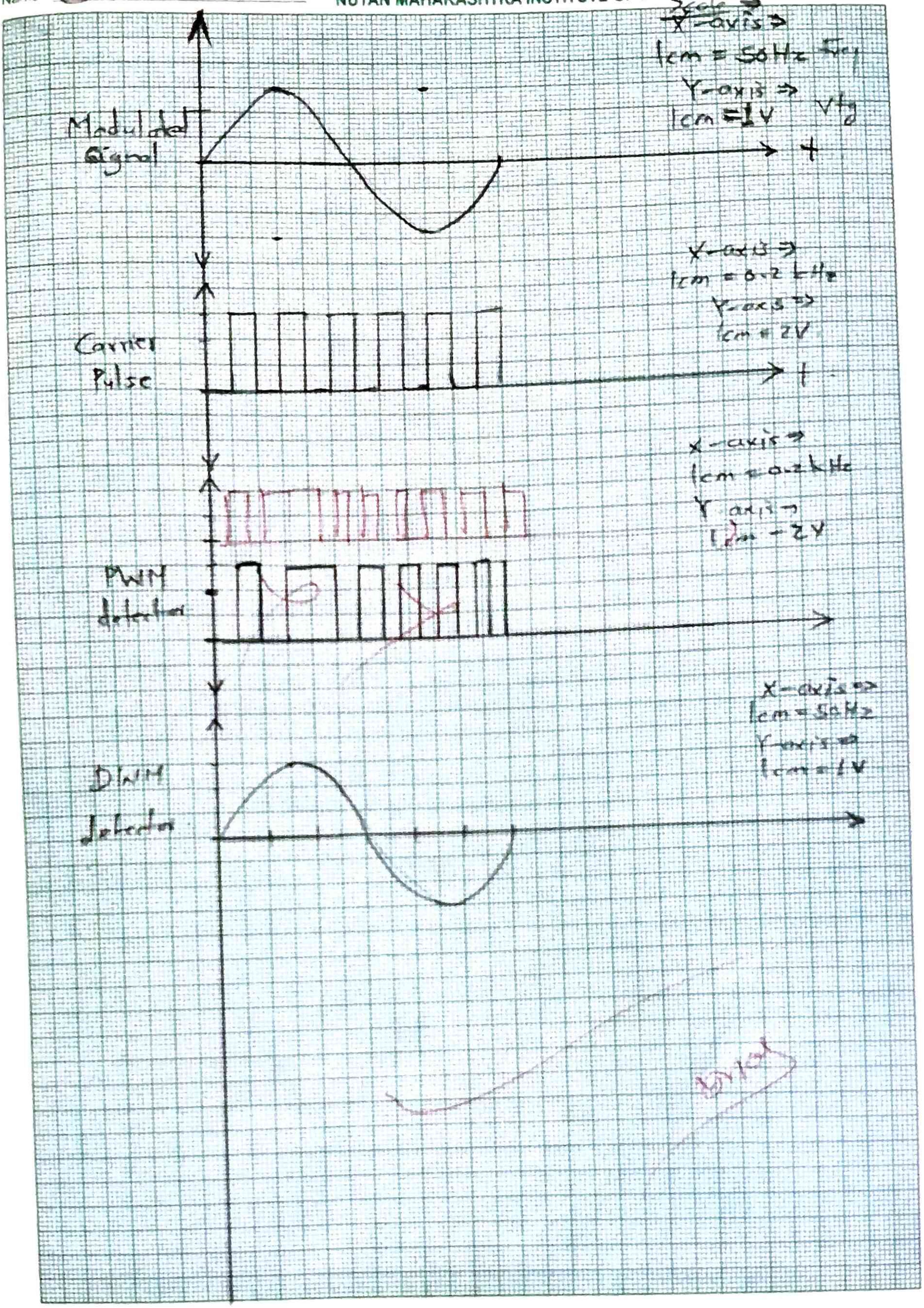


- PPM signal can be generated from PWM. The PWM pulses obtained at the comparator o/p are applied to a ~~multi~~ monostable multivibrator.
- The monostable is a -ve edge triggered.
- Hence, corresponding to each trailing edge of PWM signal the monostable o/p goes high.

Q.3. Compare PAM, PWM & PPM

⟹.

	PAM	PWM	PPM.
Pulse Immunity.	Low	High	High.
Requirement.	Low	High	High.
table parameter Pulse Carrier.	Amplitude	Width	Position.
Transmission Power.	Varies with amp. of Pulse.	Varies with Variation in Width.	Remain Constant.


Scale
X-axis ⇒
1cm = 50Hz Freq
Y-axis ⇒
1cm = 1V Vtg
Modulated Signal
t
X-axis ⇒
1cm = 0.2 kHz
Y-axis ⇒
1cm = 2V
Carrier Pulse
t
X-axis ⇒
1cm = 0.2 kHz
Y-axis ⇒
1cm = 2V
PWM detection
X-axis ⇒
1cm = 50Hz
Y-axis ⇒
1cm = 1V
DWM detector

EXPERIMENT NO. 5

AIM: To Study the Pulse Code Modulation and Demodulation

Objective: Study of Pulse Code Modulation and Demodulation. Drawbacks of PCM.

Apparatus: DM trainer kit, connecting wires, CRO.

Theory:

One method is to sample the analog signal at regular discrete intervals and code the signal amplitude into a digital format. This procedure is commonly called 'PULSE CODE MODULATION' (PCM). Pulse-code modulation or PCM is known as a digital pulse modulation technique. In fact, the pulse-code modulation is quite complex as compared to the analog pulse modulation techniques i.e. PAM, PWM and PPM, in the sense that the message signal is subjected to a great number of operations .

Reasons for Digitizing

During the past decade, digitization of analog signals for transmission and processing has become fairly common. This trend is observed in almost all fields of electronics, but specifically in communication, process control, and data processing. There are many reasons for this.

1. The digital format allows transmission of information over long distances without deterioration, since digital signals, unlike analog signals, can be regenerated with only small probability of error. They are relatively insensitive to noise, crosswalk and distortion.

2. Time division multiplexing of digital information frequently leads to economical use of cables (or channels). Compared with frequency division multiplexing, no complex filters are required in the digital case since all the multiplexing function can be accomplished with digital circuitry.

Converting Analog Signals to Digital

PCM converts analog signals into digital form through a three-step process: sampling, quantization, and encoding.

1. **Sampling:** This involves measuring the amplitude of the analog signal at uniformly spaced intervals. The frequency at which the signal is sampled is crucial, as dictated by the Nyquist Theorem, to accurately reconstruct the original signal.
2. **Quantization:** In this step, each sampled value of the signal is approximated by the nearest value within a set range. This process inherently introduces some level of quantization noise, but the effect can be minimized by increasing the resolution, or the number of bits used in the representation.
3. **Encoding:** Finally, the quantized values are encoded into a binary form to create the digital signal. This binary data represents the PCM signal, ready for processing, storage, or transmission.

This distortion arises because of large dynamic range of the input signal. the rate of rise of input signal x(t) is so high that the staircase signal can not approximate it, the step size 'Δ' becomes too small for staircase signal u(t) to follow the step segment of x(t).Hence, there is a large error between the staircase approximated signal and the original input signal x(t). This error or noise is known as **slope overload distortion** .

Granular or Idle noise occurs when the step size is too large compared to small variation in the input signal.This means that for very small variations in the input signal, the staircase signal is changed by large amount (Δ) because of large step size.when the input signal is almost flat , the staircase signal u(t) keeps on oscillating by ±Δ around the signal.The error between the input and approximated signal is called **granular noise.**

Observation Table:

Sr.No	Test points	Freq	voltage
1	Analog I/p	5.235 Hz.	0 – 5.28 V
2	Pulse generator o/p	10.47 Hz	5.28V
3	PCM O/P (Digital O/P)	548.3 Hz.	2.64 V.

Conclusion:

PCM is a digital modulation in which analogs continuous signal is converted into binary code by using sampling quantization and encoding process.

Questions :

1. Difference between Analog modulation & Digital Modulation?
2. Explain Quantization process?
3. Advantages Disadvantages and application of PCM?

Experiment No.5. (Ques)

Draw & explain PCM transmitter & Receiver diagram.

PCM Transmitter ⇒.

→ LPF → Sample & Hold → Quantizer → Encoder → ||el to Serial Converter → o/p.

↑ (to Sample & Hold and to ||el to Serial Converter) Pulse Generator.

The analog signal $x(t)$ is passed through bandlimiting LPF, which has a cutoff Freq. $f_c = W$ Hz

- This will ensure that $x(t)$ will not have only freq. component freq. component higher than "W".
- This will eliminate the possibility of aliasing.
- The band limited analog signal is then applied to a sample & hold ckt where it is sampled at adequetly high sampling rate
- Output of sample and hold block is a flat topped PAM signal.
- These samples are then subjected to the operation called "Quantization" in the Quantizer.
- Quantization process is the process of approximation as will be explained later on.

PCM Receiver ⇒.

Noise → Regeneration ckt. → Serial to ||el converter → Decoder → LPF → Analog $x(t)$.

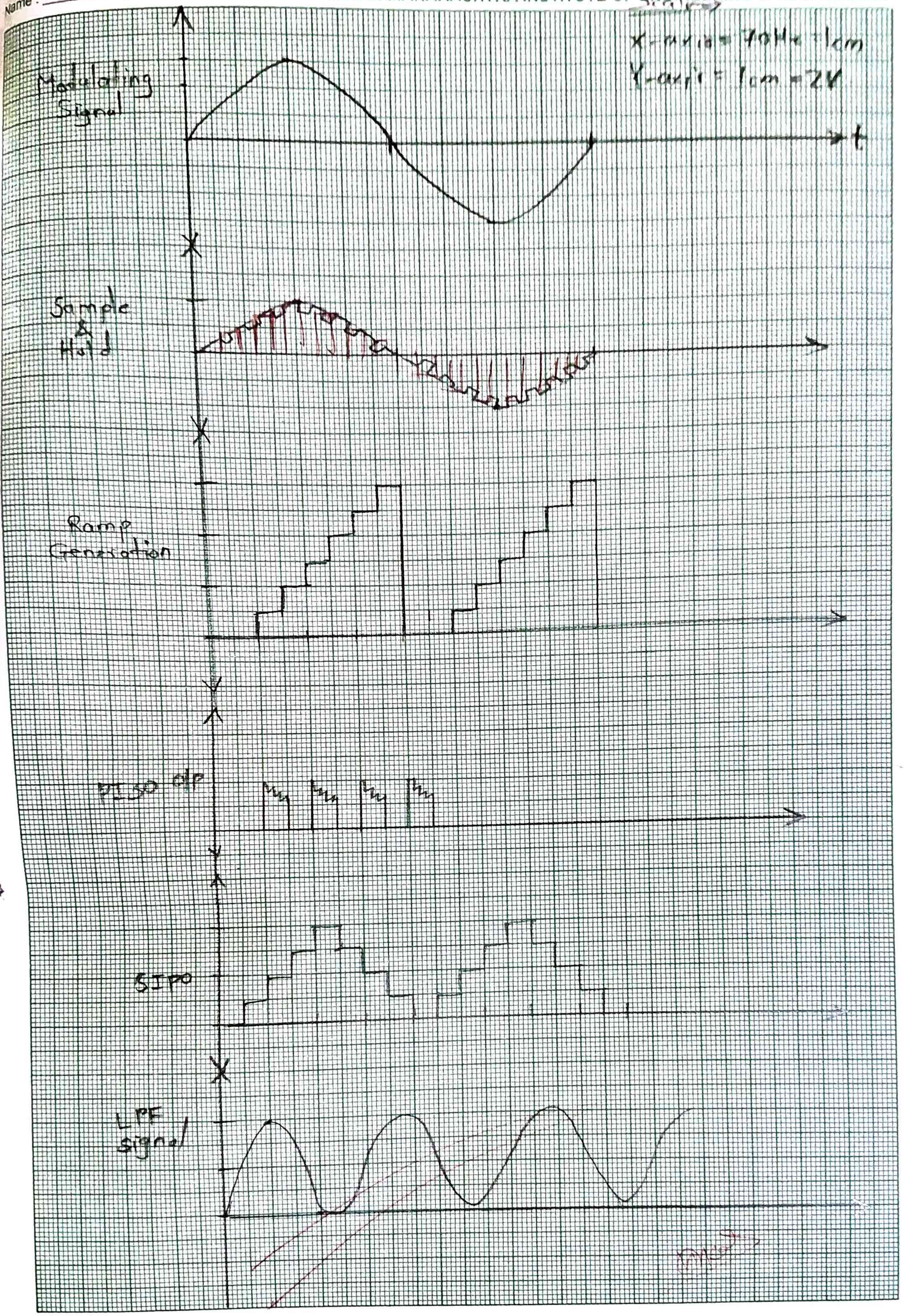
↑ (to Regeneration ckt. and to Serial to ||el converter) Pulse Generator.

PCM signal contaminated with noise is available at the receiver i/p.
- The regeneration ckt at the receiver will separate the pulses from noise and will reconstruct the original PCM signal.
- Thus at the regeration ckt o/p we get a clean PCM signal.

Q.2 Explain Drawback of PCM.

⟹ 1] The Encoding, Decoding and Quantizing circuitary of PCM is complex.

2] PCM requires a large bandwidth as compared to the other system.


X-axis = 70Hz = 1cm
Y-axis = 1cm = 2V
Modulating Signal
t
Sample & Hold
Ramp Generation
PISO o/p
SIPO
LPF signal

EXPERIMENT NO. 6

AIM: To Study of companded PCM.

Objective: Study of Companded PCM and its types.

Apparatus: Companded PCM trainer kit, connecting wires, CRO.

Theory:

Companding

Definition: Companding is a technique of achieving non-uniform quantization. It is a word formed by the combination of words **compression** and exp**anding**. Companding is done in order to improve SNR of weak signals.

If the characteristics of the quantizer is non-linear then it causes the step size to be variable despite being constant then it is known as non-uniform quantization.

In non-uniform quantization, the step size varies according to the signal level.If the signal level is low then step size will be small. So, the step size will be low for weak signal. Thus the quantization noise will also be low.

So, in order to maintain proper signal to quantization noise ratio, the step size must be variable according to the signal level.

Thus in order to achieve non-uniform quantization the process of companding is used. Let us now move further and understand the process of companding.

Model of Companding

The figure below represents the companding model in order to achieve non-uniform companding:

As we can see that the companding model consists of a compressor, a uniform quantizer and an expander.

Companding is formed by merging the compression and expanding. Initially at the transmitting end the signal is compressed and further at the receiving end the compressed signal is expanded in order to have the original signal.

Initially at the transmitting end, the signal is first provided to the compressor. The compressor unit amplifies the low value or weak signal in order to increase the signal level of the applied input signal. While if the input signal is a high level signal or strong signal then compressor attenuates that signal before providing it to the uniform quantizer present in the model.

This is done in order to have an appropriate signal level as the input to the uniform quantizer. We know a high amplitude signal needs more bandwidth and also is more likely to distort. Similarly, some drawbacks are associated with low amplitude signal and thus there exist need for such a unit.

levels and will gradually increase with an increase in the level of the input signal (PDF). In addition, the smaller the quantization interval, the better the signal-to-quantization noise ratio (SQNR). This means companding increases the SQNR at low-level signals while degrading it for higher amplitude ones .

The scenario well suits the demand for telephone systems which primarily transmit human speech wherein low amplitude quieter phonemes occur more frequently when compared to high amplitude louder phonemes (PDF). A direct consequence of this is an improvement in the quality of the audible signal, as we would have accounted for the sensitivity issues posed by the human ear.

Observation Table:

Sr.No	Testpoints	Freq	Voltage
1.	Sine wave	568.2 Hz	2 V
2.	Comparator.	160.0 Hz.	10.3 V
3.	Sample & Hold.	4.32kHz.	7.7 V
4.	D/A Converter.	560.7 Hz	2.5V
5.	LPF.	588.2 Hz	1.28 V
6.	DM.	270 Hz.	2.24 V.

Conclusion:

Companding is a technique of Achieving Non-Uniform quantization companding is done in order to improve SNR of weak signals.

Questions:

Explain Companded PCM?

Explain A law and U law?

Explain Advanges of companded PCM?

Experiment No. 6.

Q.1] Explain use of Companding.

⟹ Companding is non-uniform quantization, it is required to be implemented to improve the signal to quantization noise ratio of weak signals.

- In the uniform quantization once the step size is fixed, the quantization noise power remains constant.
- But the signal power isn't constant it is proportional to the sqⁿ of signal amplitude.

Que.2] Quantization Noise define

⟹ $$\text{Quantization Noise} = \frac{\text{Normalised Signal Power}}{\text{Normalised Noise Power}}$$

Que.3] ⟹ Quantization Process can be classified into 2-types as:

1] Uniform Quantization.

2] Non-Uniform Quantization.

- If step size (s) of quantized signal remains constant throughout the i/p range. is called uniform quantization.

- If step size varies depending on i/p then quantizer is known as the non-uniform quantizer.

Que.4] Explain mathematical repre. of A & U Law.

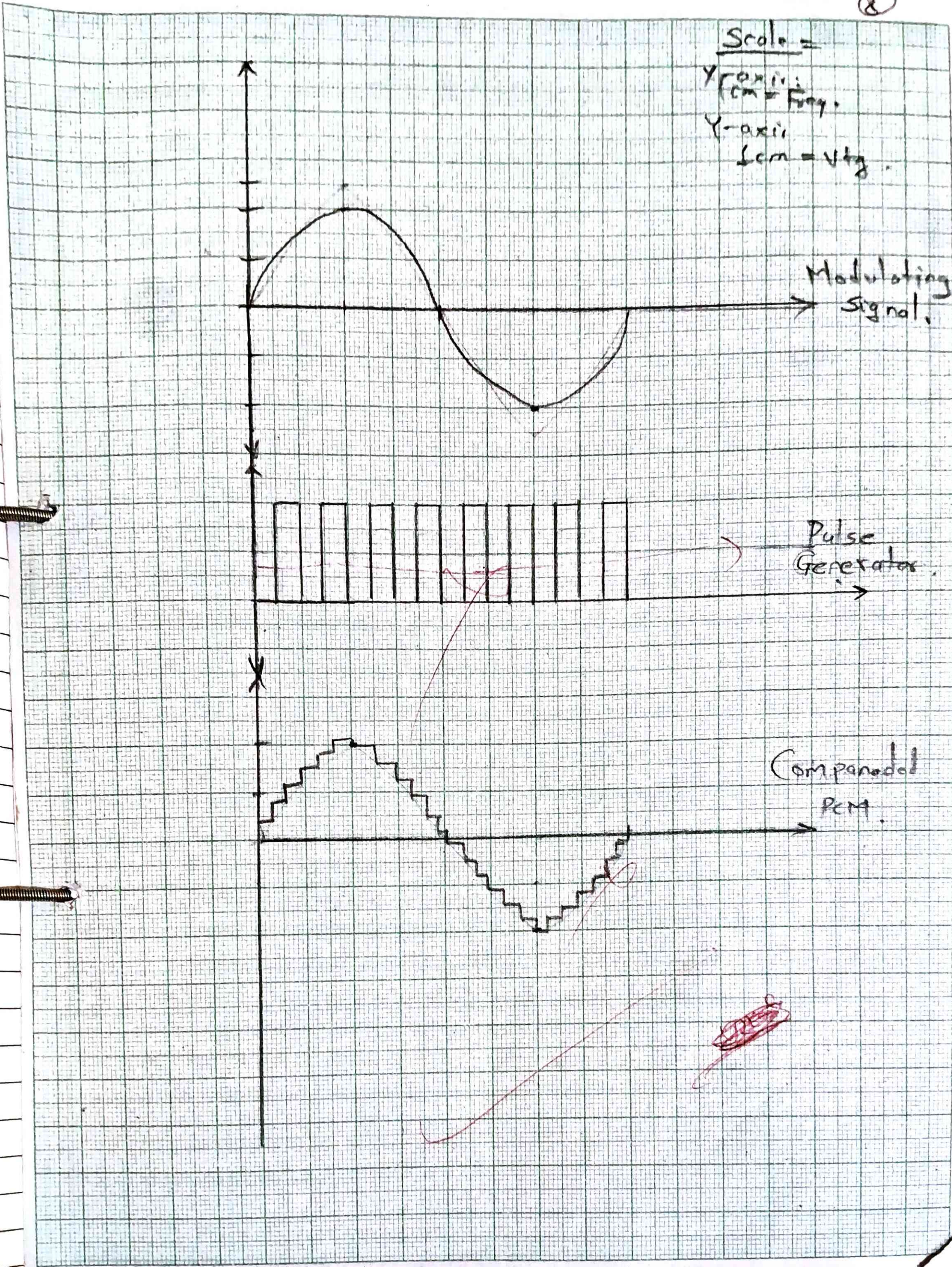
⟹ U Law :

$$Z(x) = (sgn(x)) \cdot \frac{\ln\left(1 + \frac{\mu|x|}{X_{max}}\right)}{\ln(1+\mu)}.$$

A law :

$$\frac{Z(x)}{X_{max}} = \frac{\frac{A|x|}{X_{max}}}{1+\log_e A} \qquad 0 \le \frac{|x|}{X_{max}} \le 1.$$

$$= \frac{1+\log_e\left[\frac{A|x|}{X_{max}}\right]}{1+\log_e A} \qquad \frac{1}{A} \le \frac{|x|}{X_{max}} \le 1.$$


Scale =
Y-axis:
1cm = Freq.
Y-axis
1cm = Vltg.
Modulating Signal.
Pulse Generator.
Companded PCM.

EXPERIMENT NO. 7

AIM: To Study the Delta Modulation and Demodulation

Objective: Study of Delta Modulation and Demodulation. Drawbacks of Delta Modulation: slope overload and Granular Noise.

Apparatus: DM trainer kit, connecting wires, CRO.

Theory:

The sampling rate of a signal should be higher than the Nyquist rate, to achieve better sampling. If this sampling interval in Differential PCM is reduced considerably, the sampleto-sample amplitude difference is very small, as if the difference is 1-bit quantization, then the step-size will be very small i.e., Δ (delta)

- Features of Delta Modulation

Following are some of the features of delta modulation.

- An over-sampled input is taken to make full use of the signal correlation.
- The quantization design is simple.
- The input sequence is much higher than the Nyquist rate.
- The quality is moderate.
- The design of the modulator and the demodulator is simple.
- The stair-case approximation of output waveform.
- The step-size is very small, i.e., Δ (delta).
- The bit rate can be decided by the user.
- This involves simpler implementation.

Delta Modulation is a simplified form of DPCM technique, also viewed as 1-bit DPCM scheme. As the sampling interval is reduced, the signal correlation will be higher.

The Delta Modulator comprises of a 1-bit quantizer and a delay circuit along with two summer circuits. Following is the block diagram of a delta modulator.

The predictor circuit in DPCM is replaced by a simple delay circuit in DM.From the above diagram, we have the notations as –

- x(nTs)x(nTs) = over sampled input

Observation Table:

Sr.No	Testpoints	Freq	Voltage
1.	Sine Wave.	576.8 Hz.	2.28V
2.	Comparator.	165.0 Hz.	10.2V.
3.	Sample & Hold.	4.88 kHz.	7.8V
4.	D/A Converter.	567.5 Hz.	2.66V
5.	LDF	588.5 Hz.	1.28V
6.	DM.	278.0 Hz.	2.44V

Conclusion:

In this experiment, we have learned to the DM process compress the present sample value of the previous sample value.

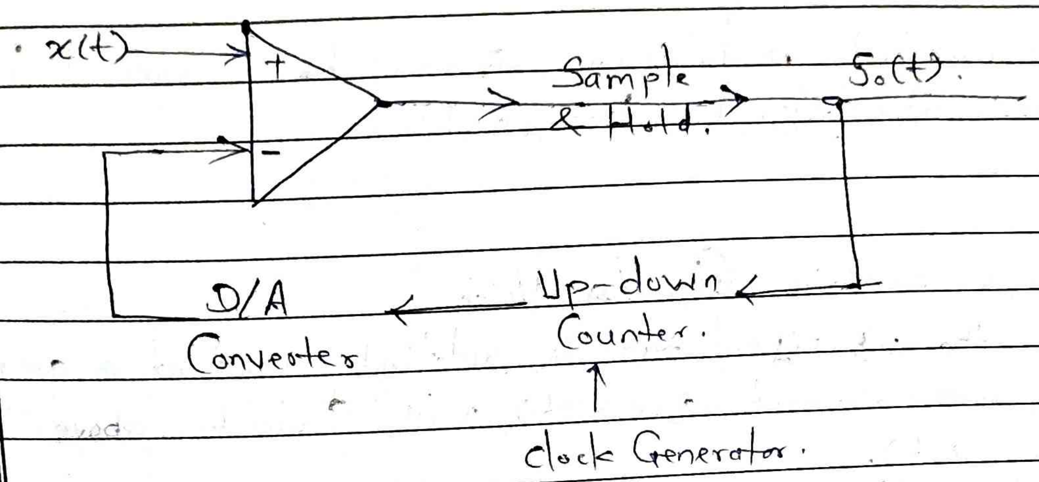
Questions:

1. Difference between PCM & DM?
2. How DM overcome the drawbacks of PCM?
3. Explain block diagram of DM transmitter and Receiver?

Experiment No. 7

I] Draw & explain DM transmitter & receiver.

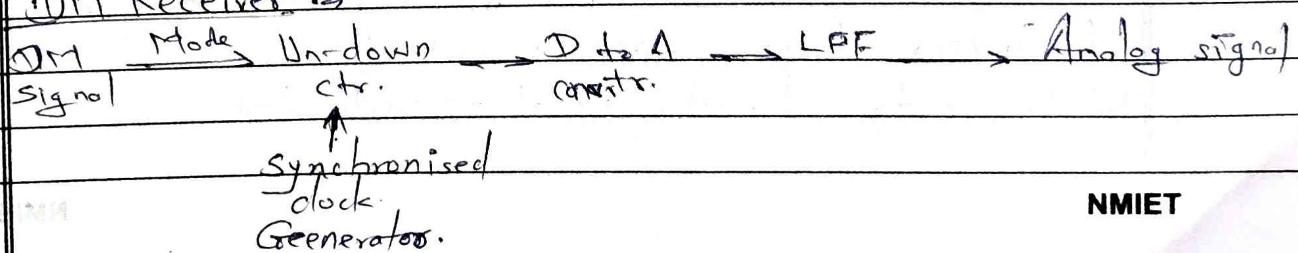
DM Transmitter ⇒



x(t) is the analog i/p signal & x(t) is the quantizer version of x(t) both these signals are applied to a computer.

- The comparator gives high if x(t) > x(t) & it goes low if x(t) < x(t)
- Thus, the comparator o/p is either 1 or 0. The sample & Hold ckt will hold this level (0 or 1) for the entire clock cycle period.
- The o/p of the sample & hold ckt is transmitted as the o/p of DM system.
- Thus, in DM info. which is transmitted is only whether x(t) > x(t).
- Also note that 1 bit / clock cycle is being sent this will reduce the bit rate & hence the B/W DM receive.

DM Receiver ⇒



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- Compare it with the transmitter's block diag. Then will find that it's identical to chain block producing the signal x(t).
- The original modulating sig can be recovered back by passing this signal through LPF.

Q.2 Draw & explain Granular Noise.

⟹ When the i/p signal x(t) is relatively constant in amplitude, the approximated sig. x'(t) will fluctuate above & below x(t).

- The diff. betⁿ. x(t) & x'(t) is called granular noise.
- Granular noise is similar to the quantized noise in PCM system.
- It increases with increase in the step size of the reduc the granular noise, the step-size should be as small as possible
- However this will ~~overload~~ increase slope overload distortion.

Q.3 Draw & explain slope-Overload distortion.

⟹ 1] Occurs in Delta Modulation (DM).

2] Happens when i/p signal changes too fast.

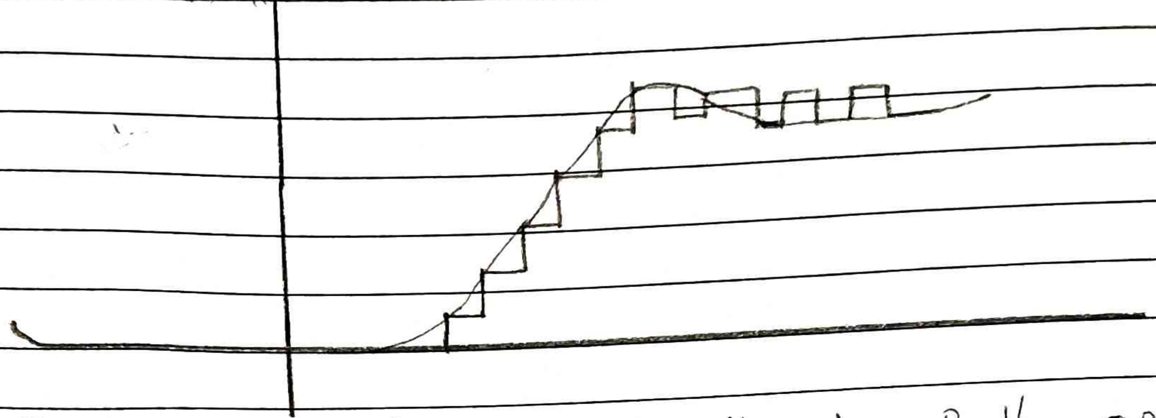
3] Modulator cannot track rapid changes.

4] Step size becomes too small to follow input.

5] Causes large error in output.

6] Leads to distorted signal reconstruction.

* Overload distortion ⇒.



Due to small step-size (δ), the slope of the appr. sig ($x'(t)$) will be small.

The Slope of $x'(t) = \frac{\delta}{T_s} = \delta F_s$.

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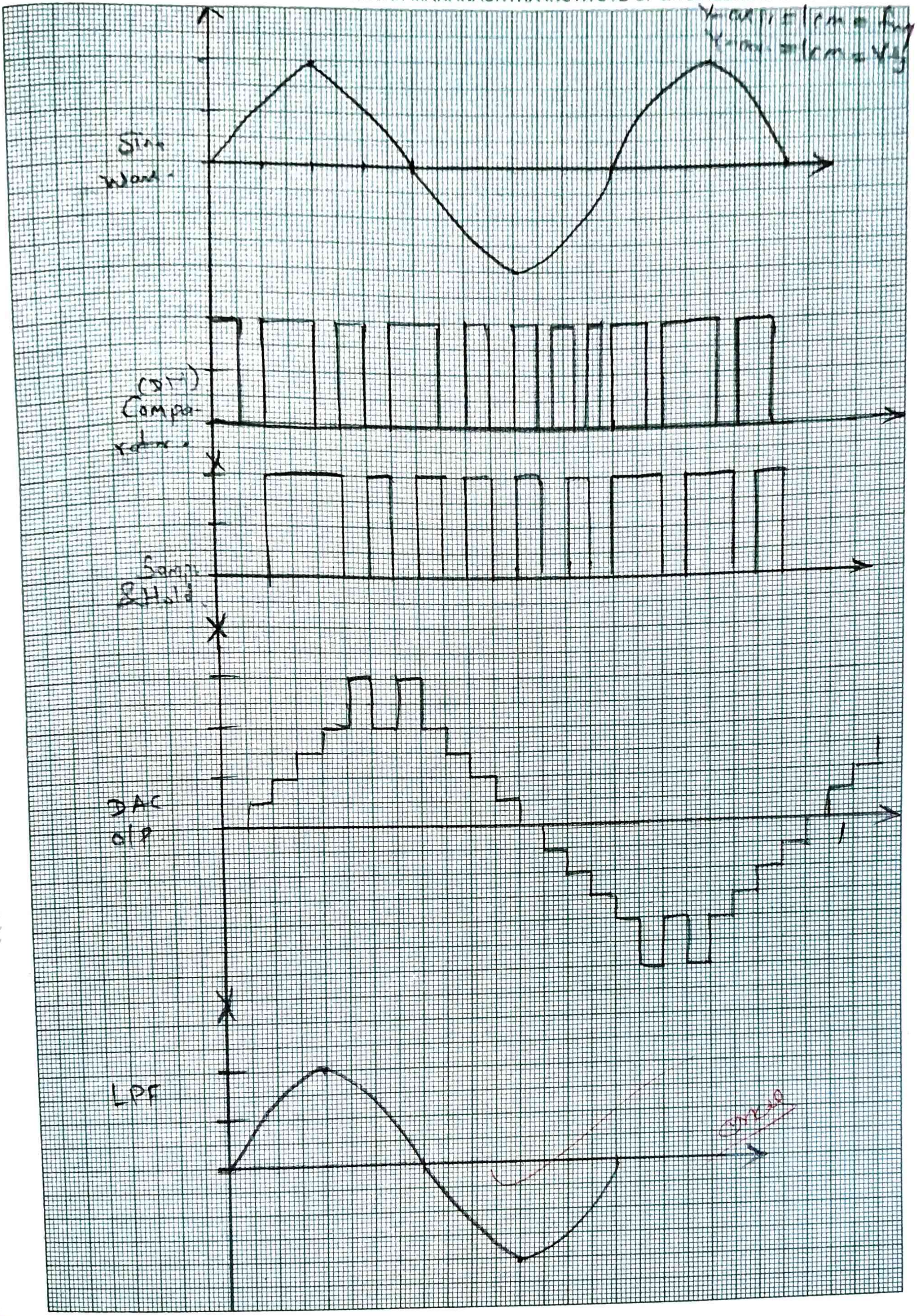
Scale



Experiment No.8

AIM: To study Theory of Adaptive Delta modulation & demodulation

OBJECTIVE:

APPARATUS: ADM Trainer kit, CRO, DSO, Connecting Wires.

THEORY:

(1) Linear Delta Modulation (LDM)

In this type of modulation, an input signal is compared with its staircase-approximated signal by a comparator. The error output signal of the comparator is then sampled once per clock by a sample and hold circuit. The output

of sample and hold is a digital output signal, which is used for transmission. This digital signal is also fed back to an integrator. The integrator (also known as local decoder) generates a positive or a negative voltage step according to the output digital signal. This voltage step is added to the previous value of staircase-approximated signal. Thus this signal tracks the input signal with staircase nature.

A demodulator (decoder) consists of an integrator & a Low Pass Filter (LPF). The integrator is similar to that is used in the encoder. Hence the output of integration is a staircase-approximated signal. This signal is passed through LPF. The error between the original input signal and approximated staircase signal, is of high frequency hence the output of LPF, the recovered signal is a close facsimile of the input signal.

This LDM system have drawback of slope over load and hunting due to fixed size of step. To overcome this drawback, step size should very with the slope of the input signal. The system in which step is changed automatic, is known as Adaptive Delta Modulation (ADM).

(2) Adaptive Delta Modulation (ADM): -

This system is similar to LDM except the step size is variable. The step is increased when the slope of the input signal is more and decreased when the slope of the input signal is less.

The Adaptive Delta Modulation technique helped to overcome the problem of granular noise in delta modulation. This method improves the granular noise and slope overload error found in delta modulation.

The adaptive delta modulation technique overcomes the above mentioned problems by providing variable step size according to the input provided.

The quality of output is hampered in case of delta modulation because of fixed step size. In case of a steep slope of a modulating signal a larger step size is needed and vice versa. Due to this we are unable to get the exact value and hence it is required to make the step size adjustable in order to obtain desired result. This is the main advantage and the theory behind Adaptive Delta Modulation. The working of adaptive delta modulation is as in the block diagram.

D. Connect the comparator section output to the sample and hold block of the trainer Kit.
E. Connect the sample and hold block's output to the step controller section
F. Connect the output of the A-D convertor section output to the second input of the comparator section.
G. Connect the second output of the A-D convertor section to the input of the LPF Section for recovering of Original Signal.
H. Observe Waveform on Each Point.
I. Draw and comment Each waveform

Observation Table:

Sr.No	Testpoints	Freq	Voltage
1.	Sine Wave	2.40 kHz.	1.52 V
2.	Comparator.	2.38 kHz.	5.4 V
3.	Sample & Hold.	2.39 kHz.	5.2 V
4.	DA Converter.	2266.4 Hz.	2.0 V
5.	LPF	272.9 Hz.	1.5 V
6.	ADM.	417.0 Hz.	984 mV.

Conclusion:

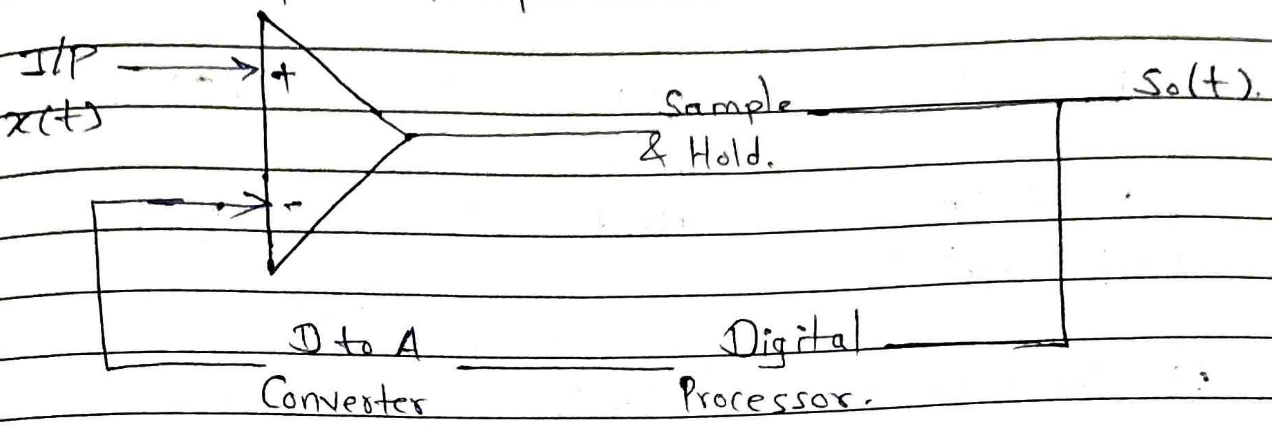
The ADM technique helped to overcome the problem of granular noise in data modulation, this method improves the ~~gr~~ granular noise and slope overload found in DM.

Questions:

Experiment. No. 28.

Q.1] Draw & explain Adaptive Delta Modulation.

⇒



In the ADM system, the step-size isn't constant rather when the slope overload occurs the step size is becomes progressive larger & it will $x'(t)$ catch with $x(t)$ more rapidely.

- In response to the kth clock pulse trailing edge, the processor generates a step which is equal to magnitude to the step generated in response to the previous.
- If the direction of both the steps is same, then the processor will increase the magnitude of the present step by 'δ'.

$S_o(t) = 1$, if $x(t) > x'(t)$

$S_o(t) = -1$, if $x(t) < x'(t)$.

Q2] Explain 2-Drawbacks over by ADM ?

⇒ For a relatively constant magnitude i/p signal $x(t)$, the ADM will produce a high granular noise.

Q.3 Compare DM & ADM.

Parameter.	DM.	ADM
1 No. of bits per sample.	N=1.	N=1.
2 Step-Size.	Fixed.	Variable
3 Complexity	Simple	Simple.
4 Noise Immunity	Very Good	Very Good.
5 Use of Repeaters.	Possible.	Possible.

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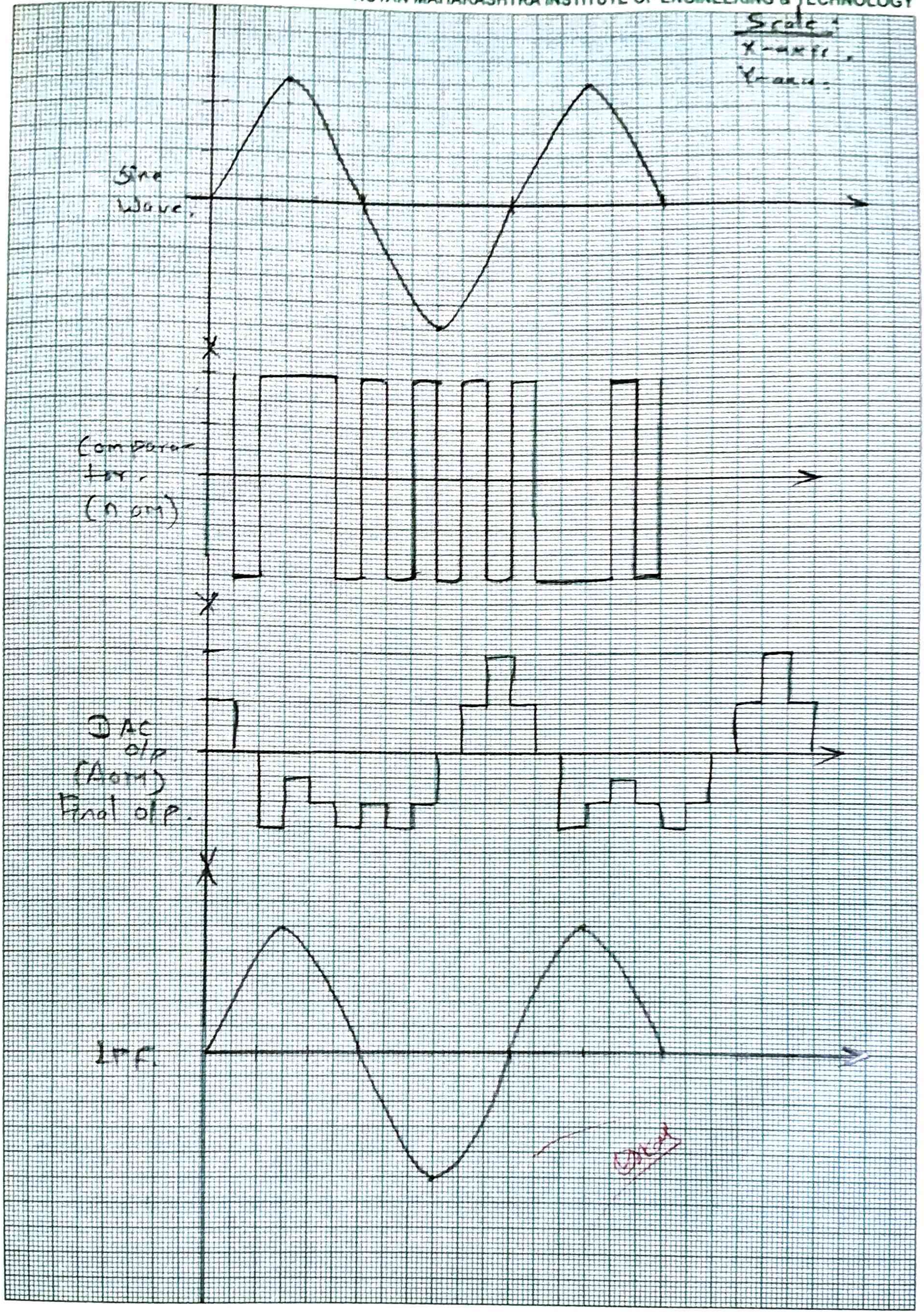


Experiment No.9

AIM: Study of Spectral analysis of line codes.

OBJECTIVE:

1. To study concept of line coding
2. Characteristics of line coding
3. Comparison of different line codes.

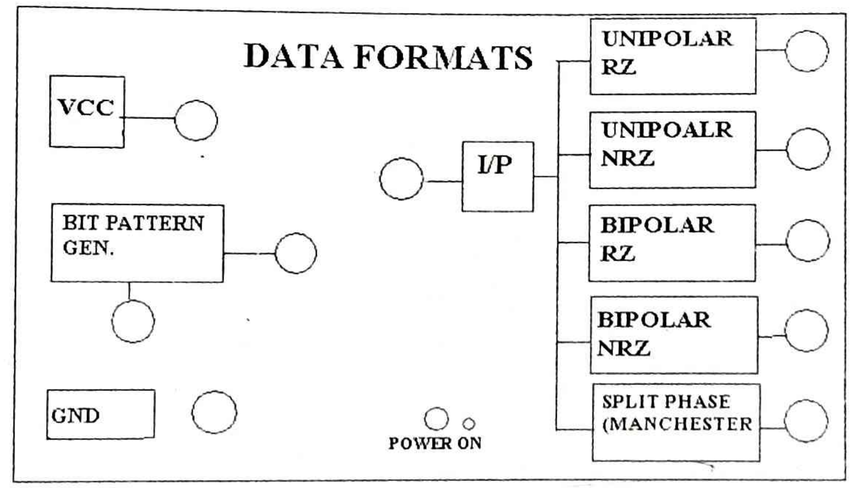
APPARATUS: Line coding kit, CRO, Spectrum Analyzer.

THEORY:-

The symbols `0' and `1' in digital system can be represented in various formats with different levels and waveforms. The selection of particular format for communication depends on the system bandwidth, system's ability to pass DC level information, error checking facility, ease of clock regeneration & Synchronization at receiver, system complexity and cost etc.

Line coding is the process of converting digital data to digital signals. We assume that the data, in the form of of text, numbers, graphical images audio or video are stored in computer memory as sequence of bits. Line coding converts a sequence of bits to a digital signal. At the sender, digital data are encoded into a digital signal. At receiver the digital data are recreated by decoding the digital signal.

Digital data can be transmitted by various transmission or line codes such as on-off, polar, bipolar and so on.. Each has its own Advantages and disadvantages.



<u>Unipolar RZ Formats:</u> The return to zero (RZ) unipolar format is as shown in fig.1. In this format each "0" is represented by an off pulse (ak = 0) and each "1" by an on pulse with amplitude ak = A and a duration of Tb/2, followed by a return to zero level. Therefore this is called as return to zero

Observation Table:

Sr.No	Test points	Frequency(Hz)Bit Period	Voltage(V)
1	Data Generation (1 0 1 1 0 1 1 0)	900.9 Hz.	5V
2	Bipolar NRZ	678.0 Hz	5V
3	Bipolar NRZ decode o/p.	902.9 Hz	5V
4	Bipolar RZ	1.351 kHz	5V
5	Bipolar Am	900.9 Hz	5V
6	Polar NRZ	1.476 kHz	5V
7	Polar RZ.	900.9 Hz.	5V

GRAPHS:

Plot the observed line codes of a given sequence.

Conclusion:

We have learned various line coding techniques properties and characteristics of line coding.

Questions:

1. What are the properties of line coding? Compare RZ and NRZ line coding formats on the basis of above properties along with their merits and demerits?
2. Explain power spectral of various line coding?
3. Draw the following line codes for 101101000, Unipolar RZ, AMI, Polar RZ, Split phase Manchester

22/4/25

Experiment No. 9

Que. 1] What are the properties of line coding? Compare RZ & NRZ line coding formats on the basis of above properties along with their merits & demerits?

⟹ Properties of Line Coding ⇒

1] High Efficiency ⇒ As the code adds redundancy the code efficiently should be as high as possible.

2] Self Synchronization ⇒ To ensure synchronization at the receiver, the line signal should undergo a sufficient no. of zero crossings that means, the transmitted should always undergo transitions.

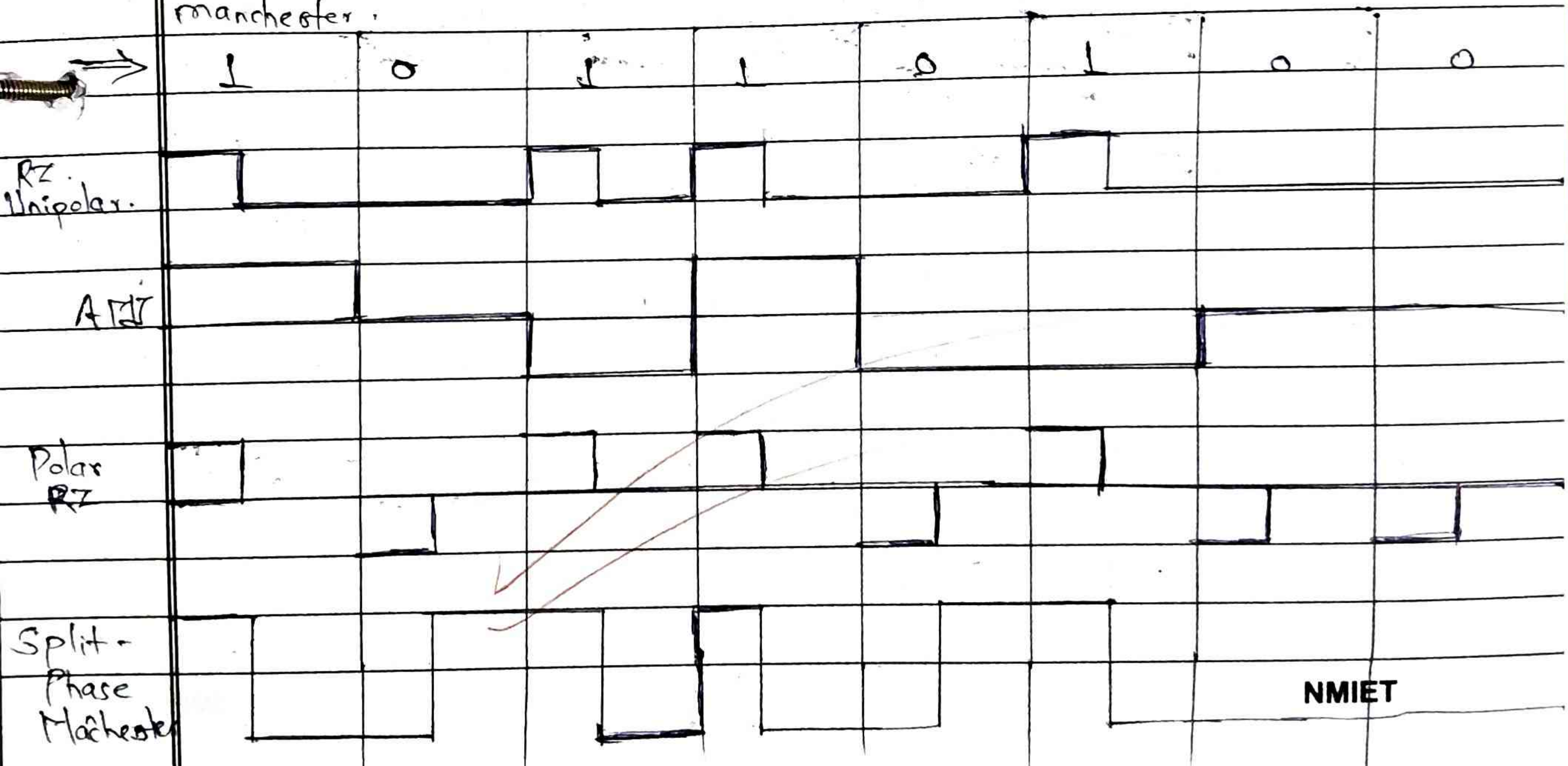
3] No Crosstalk ⇒ Cross-talk betn channel should be minimized

4] Self Clocking Capability ⇒ Any digital system needs symbol or bit synchronization.

5] Error Detection ⇒ Some of the line codes such as 'duobinary' are capable of detecting the data errors without introducing additional error detection bits into the data sequence.

Que. 3] Draw the code for 10110100, unipolar RZ, AMI, Polar RZ, split phase manchester.

⟹



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Que:2 Explain power spectral various line coding?

⟹ 1) Power Spectra of NRZ Unipolar ⇒

- Power spectral density & freq. are normalised
- Most of the power of the NRZ unipolar Format is concentrated betn dc ($F=0$) & the bit rate of i/p data ($F_2 = 1/T_0$)

2) Power Spectra of NRZ Polar ⇒

- Also like unipolar NRZ format, most of the power lies in the main lobe of the sinc shaped curve.

3) Power spectra of Bipolar Format ⇒

- It can be seen that almost all the power lies in the bandwidth equal to $1/T_0$.

4) Manchester Format

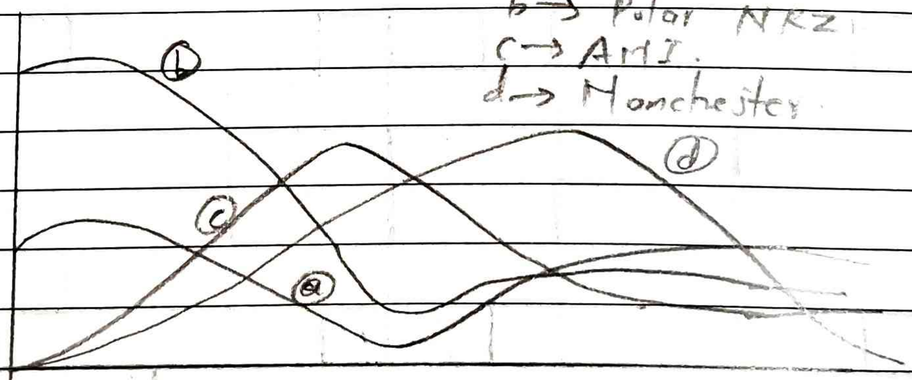
- Here most of power lies inside the B/W equal to $2/T_0$ which is twice the B/W of Unipolar, Polar & bipolar NRZ Formats.

a → Unipolar NRZ

b → Polar NRZ

c → AMI

d → Manchester



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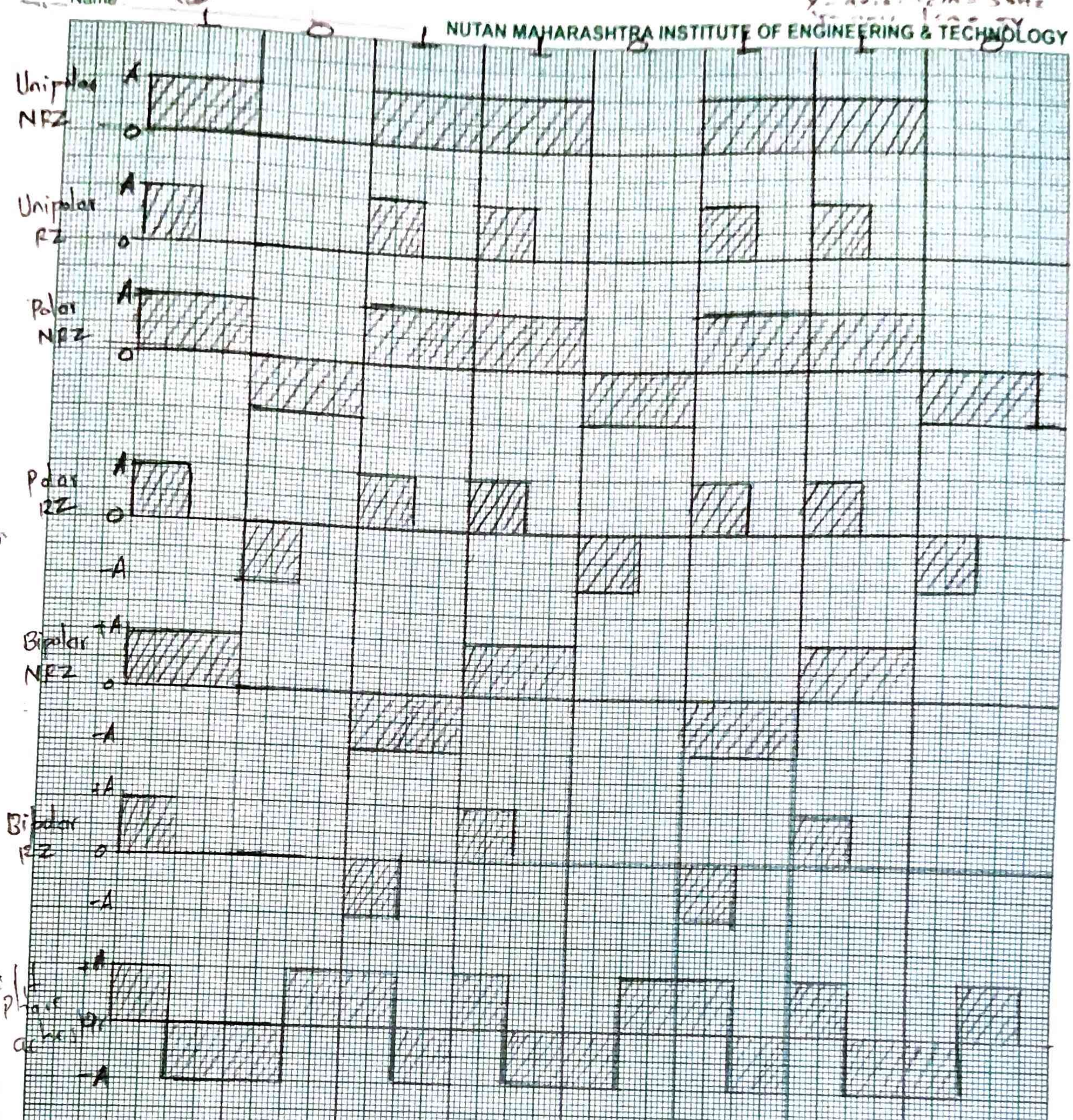


EXPERIMENT NO. 10

AIM: To Verify Sampling Theorem using simulation.

Objectiive: Simulate the sampling conditions 1. Fs>2Fm , 2. Fs=2Fm, 3. Fs < 2Fm.

Apparatus: MATLAB Software.

Theory:

Sampling is defined as, "The process of measuring the instantaneous values of continuous-time signal in a discrete form."

Sampling Theorem statement:

Sampling theorem states that "continues form of a time-variant signal can be represented in the discrete form of a signal with help of samples and the sampled (discrete) signal can be recovered to original form when the sampling signal frequency Fs having the greater frequency value than or equal to the input signal.

Sampling Rate

To discretize the signals, the gap between the samples should be fixed. That gap can be termed as a sampling period Ts.

$$\text{Sampling Frequency} = 1/Ts = fs$$

Where,

- Ts is the sampling time
- fs is the sampling frequency or the sampling rate

Sampling frequency is the reciprocal of the sampling period. This sampling frequency, can be simply called as Sampling rate. The sampling rate denotes the number of samples taken per second, or for a finite set of values.

For an analog signal to be reconstructed from the digitized signal, the sampling rate should be highly considered. The rate of sampling should be such that the data in the message signal should neither be lost nor it should get over-lapped. Hence, a rate was fixed for this, called as Nyquist rate.

Nyquist Rate

Suppose that a signal is band-limited with no frequency components higher than W Hertz. That means, W is the highest frequency. For such a signal, for effective reproduction of the original signal, the sampling rate should be twice the highest frequency.

Which means,

$$fs = 2W$$

Where,

- fs is the sampling rate

Algorithm ⇒

1) Start.

2) I/p the desired frequency for which sampling thm is to verified.

3) Generate on analog signal $x(t)$ of frequency for comparison.

4) Generate overdamped, nyquist & under sampled discrete time signals.

5) Plot the waveforms and hence prove sampling theorem.

Flow Chart ⇒

(Start)

↓

define time axis $t = -10$ to 10 with step 0.01. Set $T = 4$, Calculate $Fm = \frac{1}{T}$. Generate CT signal $x(t) = \cos(2\pi Fmt)$

↓

Plot CT signal $x(t)$, subplot (2, 2, 1)

↓

Define sampling points $n1 = -4$ to 4, set $Fs1 = 1.6\,fm$
Sample $x1(n) = \cos(2\pi Fm/Fs1 * n1)$

↓

Plot sampled signal $x1$ ($Fs < 2Fm$) (subplot(2,2,2).)

↓

Define Sampling points $n2 = -5$ to 5; set $fs2 = 2fm$
Sample $x2(n) = \cos(2\pi fm/fs2 * n2)$

↓

plot Sampled signal $x2$ ($fs = 2fm$) (subplot(2,2,3))

↓

Define Sampling points $n3 = -20$ to 20, set $fs3 = 8fm$
Sample $X3(n) = \cos(2\pi Fm/Fs3 * n3)$

↓

Plot Sampled Signal $x3$ ($Fs >> 2Fm$) subplot (2,2,4)

↓

(End).

observe from the above pattern that the over-lapping of information is done, which leads to mixing up and loss of information. This unwanted phenomenon of over-lapping is called as Aliasing.

Aliasing

Aliasing can be referred to as "the phenomenon of a high-frequency component in the spectrum of a signal, taking on the identity of a low-frequency component in the spectrum of its sampled version."

The corrective measures taken to reduce the effect of Aliasing are –

In the transmitter section of PCM, a low pass anti-aliasing filter is employed, before the sampler, to eliminate the high frequency components, which are unwanted.

The signal which is sampled after filtering, is sampled at a rate slightly higher than the Nyquist rate. This choice of having the sampling rate higher than Nyquist rate, also helps in the easier design of the reconstruction filter at the receiver.

Solution on Aliasing:

Solution 1: Anti-Aliasing Analog Filter

1. All physically realizable signals are not completely band limited
2. If there is a significant amount of energy in frequencies above half the sampling frequency (fs /2), aliasing will occur
3. Aliasing can be prevented by first passing the analog signal through an called a pre filter) before sampling is performed
4. The anti-aliasing filter is simply a LPF with cutoff frequency equal to half the sample rate

Solution 2: Over Sampling and Filtering in the Digital Domain

1. The signal is passed through a low performance (less costly) analog low the bandwidth.
2. Sample the resulting signal at a high sampling frequency.
3. The digital samples are then processed by a high performance digital filter and down sample the resulting signal.

Flowchart:
Algorithm:
Program:
Input:
Output:

attached.

Conclusion:

In this experiment, we have verified the sampling theorem using MATLAB s/w.

Expt. 10.

Q.1] What does sampling mean. Name various sampling technique?

⟹ Sampling extracts discrete values of a continuous signal at regular time interval.

Ideal impulse sampling uses infinitely narrow pulse, natural sampling uses rectangular pulses of finite width.

Flat-top scrambling holds the sampled-value const. for a period.

Que2] What do you know abt impulse scrambling mention its disadvantages!

⟹ Impulses scrambling in the freq. domain results in a periodic repition of the original singnal's spectrum.

Recovery of the original signal ideally requires a perfect lowe pass filter after impulse scrambling.

A practical disadvantages is the difficulty in generating ideal impulses.

Q.3] Explain subplot stem & plot in matlab.

⟹ Subplot is essential for visualizing multiple plots within a single figure for comparison or analysis.

Stem is particularly useful for displaying discrete time-signal or the individual value of a sampled signal.

Plot is for visualizing CT signal & sampled data where a trend or connectn betn points is important.

Q.4] Explain anti-aliasing filter?

⟹ • The cutoff freq. of the antialiasing filter is typically set to the nyquist frequency.

• Without its freq. above the Nyquist rate would fold back into the lower freq. spectrum causing distortn.

EXPERIMENT NO. 11

AIM: Simulation program for PCM system & DM system.

Objectiive: To Simulation program to calculate Signal to noise ratio for PCM system & DM system

Apparatus: MATLAB Software.

Theory:

Quantization Process:

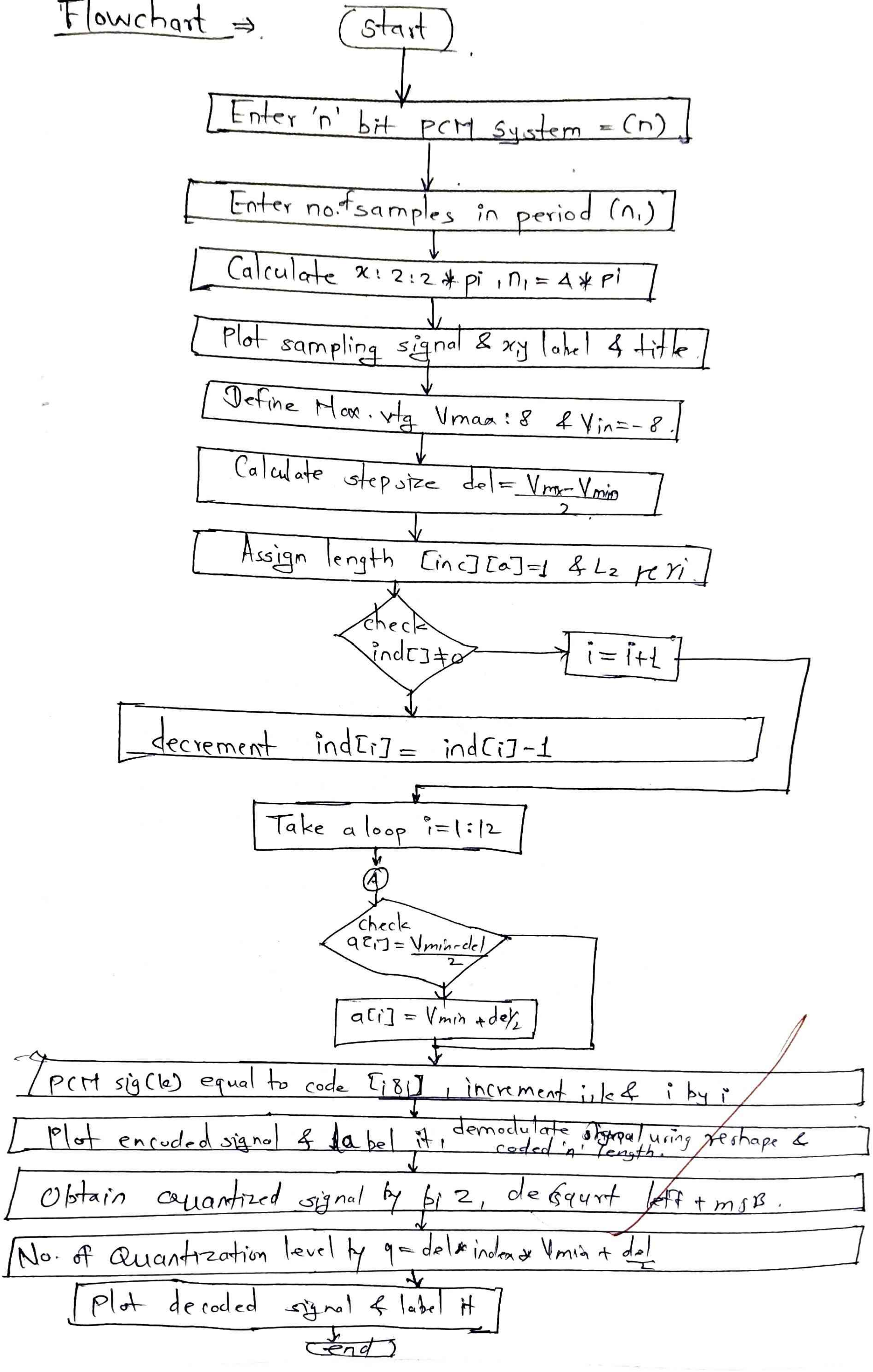
- Quantization is process of approximation or rounding off.
- The sampled signal in PCM transmitted is applied to the quantizer block.
- Quantizer converts the sampled signal into an approximate quantized signal, which consists of only a finite number of predecided voltage levels.
- Each sampled value at the input of the quantizer is approximated or rounded off to the nearest standard predecided voltage level.
- This standard levels are known as the "quantization levels" refer figure, to understand the process of quantization.

The quantization Process takes place as follows:

- The input signal X (t) is assumed to have a peak-to-peak swing of VL to VH volts. This entire voltage range has been divided into "Q" equal intervals of step size "S".
- "S" is called as the step size and its value is given as, S = VH-VL/Q.
- In figure the value of Q=8.
- At the center of these steps, the quantization levels q0, q1, q3…q7 are located.
- Xq (t) represents the quantized version of X (t). We obtain Xq (t) at the output of the quantizer. When X (t) is in the range Δ0, then corresponding to each value of X (t), the quantizer output will be equal to "q0". Similarly for all the values of X (t) on the range Δ1, the quantizer output is constant equal to "q1". Thus in each range from Δ0 to Δ7, the signal X (t) is rounded off to the nearest quantization level and quantized signal is produced.
- The quantized signal Xq (t) is thus approximation of X (t). The difference between them is called quantization error or quantization noise. This error should be as small as possible. To minimize the quantization error we need to reduce the step size "S" by increasing the number of quantization levels Q.

Why Quantization is required?

Flowchart ⇒


start
Enter 'n' bit PCM System = (n)
Enter no. of samples in period (n1)
Calculate x: 2:2 * pi , n1 = 4 * pi
Plot sampling signal & xy label & title
Define Max. vtg Vmax: 8 & Vmin = -8
Calculate stepsize del = Vmax - Vmin / 2
Assign length [ind] [a] = 1 & L2 per i
check ind[] ≠ 0
i = i+1
decrement ind[i] = ind[i] - 1
Take a loop i = 1:12
A
check q[i] = Vmin - del/2
a[i] = Vmin + del/2
PCM sig(k) equal to code [i 8 1] , increment i, k & i by i
Plot encoded signal & label it, demodulate signal using reshape & coded 'n' length.
Obtain quantized signal by bi 2, de [illegible] left + msB.
No. of Quantization level by q = del * index + Vmin + del/2
Plot decoded signal & label it
end

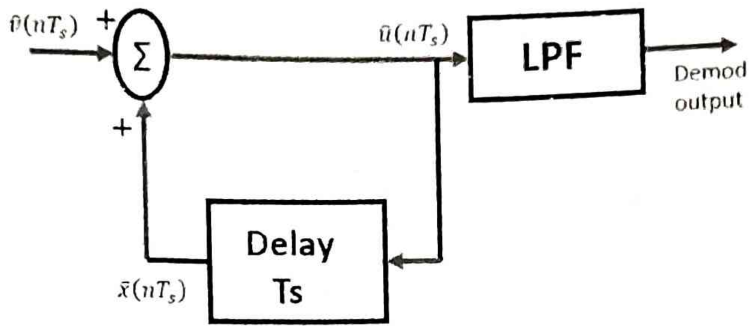
Algorithm ⇒

1] Start.
2] Enter val. of 'n' bit PCS system.
3] Enter No. of Samples.
4] Sampling Operation:
   (a) $x = 0.2 \& pi \; (n = 4 * pi)$
   (b) $s = 8 \cdot \sin(x)$
   (c) subplot & plot sampled signal.
5] Quantization Process:
   1> Init $V_{max} = 8$.
   2> $V_{min} = -V_{max}$
   3> Quantizer = (s, part, Code).
   4> Check Condition (i=1 : (i))
   5> if $(ac) = V_{min} - (\frac{del}{2})$ &
      $a(i) = V_{min} + (\frac{del}{2})$
6] Encoding Power:
   a> Code a de2bi(in1, left-msb).
   b> $j = j+1$ & $k = k+1$.
   c> Plot the signal.
7] Demodulation of PCM ⇒
   (1) Quant = Reshape(code, n length(coded/n)).
   (2) $q = del * index + V_{min} + (\frac{del}{2})$.
   (3) stop.

Delta Demodulator

The delta demodulator comprises of a low pass filter, a summer, and a delay circuit. The predictor circuit is eliminated here and hence no assumed input is given to the demodulator.

Following is the diagram for delta demodulator.



A binary sequence will be given as an input to the demodulator. The stair-case approximated output is given to the LPF. Low pass filter is used for many reasons, but the prominent reason is noise elimination for out-of-band signals. The step-size error that may occur at the transmitter is called **granular noise**, which is eliminated here. If there is no noise present, then the modulator output equals the demodulator input.

Write following data for PCM and DM:

1. **Flowchart:**
2. **Algorithm:**
3. **Program:**
4. **Input:**
5. **Output (Printouts):**
6. **Result:**

attached.

Conclusion:

Analog signal is converted into digital signal by process of quantization and the drawback of PCM are overcame by DM like large no. of bits and large transmission bwd.

17/4/25

Experiment No. 11.

Que. 1] What are advantages of digital communication.

⟹ 1] Immune to Noise & Interference.

2] Regeneration of Coded signal possible along path.

3] Highly secure.

4] Possible to use uniform format for diff. baseband signals.

5] Possible to store signal & process it further.

Que. 2] Disadvantages of Digital Communication.

⟹ 1] Complex System.

2] Synchronization is very necessary.

3] Required b/w increased due to signal technology.

4] It may be costly.

5] Not Compatible to older analog transmission systems.

Que. 3] Diff. betn PCM & DM, ADM, DPCM?

⟹

Parameter	PCM	DM	ADM	DPCM.
1) Bits per Sample.	N = 4, 8, 16, 32, 64 ...	N = 1.	N = 1.	N > 1 but N < L for PCM
2) Step Size.	Depends on No. of levels.	Fixed	Variable.	Fixed.
3] ~~Error~~ Complexity	~~Quantization~~ Complex	Simple.	Simple.	Simple.
4) Error.	Quantization Error.	slope Overload, Granular Noise.	Granular Noise	Slope Overload, Granular Noise

Q:4] Application of DM & PCM.

⟹ DM ⟹

1] Voice Modulation in telephony.
2] Military Comm. Systems.
3] Satellite and Space comm.
4] Modem & Radio Comm.

PCM →

1] Digital telephony.
2] CD & DVD audio systems.
3] TV Broadcasting
4] Satellite comm.
5] Data Storage & transmission.

Experiment No.12

AIM: To Demonstrate Scrambling and descrambling operation either using hardware or any simulation tool.

OBJECTIVE: Demonstrate Scrambling and descrambling operation either using hardware or any simulation tool.

APPARATUS: MATLAB Software

THEORY: Scrambling and Descrambling: In telecommunications and recording, a scrambler (also referred to as a randomizer) is a device that manipulates a data stream before transmitting. The manipulations are reversed by a descrambler at the receiving side. Scrambling is widely used in satellite, radio relay communications and PSTN modems. Scrambling is a technique that does not increase the number of bits and does provide synchronization. Problem with technique like Bipolar AMI(Alternate Mark Inversion) is that continuous sequence of zero's create synchronization problems one solution to this is Scrambling Purposes of scrambling

A scrambler (or randomizer) can be either:

1. An algorithm that converts an input string into a seemingly random output string of the same length (e.g., by pseudo-randomly selecting bits to invert), thus avoiding long sequences of bits of the same value; in this context, a randomizer is also referred to as a scrambler. 2. An analog or digital source of unpredictable (i.e., high entropy), unbiased, and usually independent (i.e., random) output bits. A "truly" random generator may be used to feed a (more practical) deterministic pseudo-random random number generator, which extends the random seed value.

There are two main reasons why scrambling is used:

• To enable accurate timing recovery on receiver equipment without resorting to redundant line coding. It facilitates the work of a timing recovery circuit an automatic gain control and other adaptive circuits of the receiver (eliminating long sequences consisting of '0' or '1' only).

• For energy dispersal on the carrier, reducing inter-carrier signal interference. It eliminates the dependence of a signal's power spectrum upon the actual transmitted data, making it more dispersed to meet maximum power spectral density requirements (because if the power is concentrated in a narrow frequency band, it can interfere with adjacent channels due to the intermodulation (also known as cross-modulation) caused by non-linearities of the receiving tract).

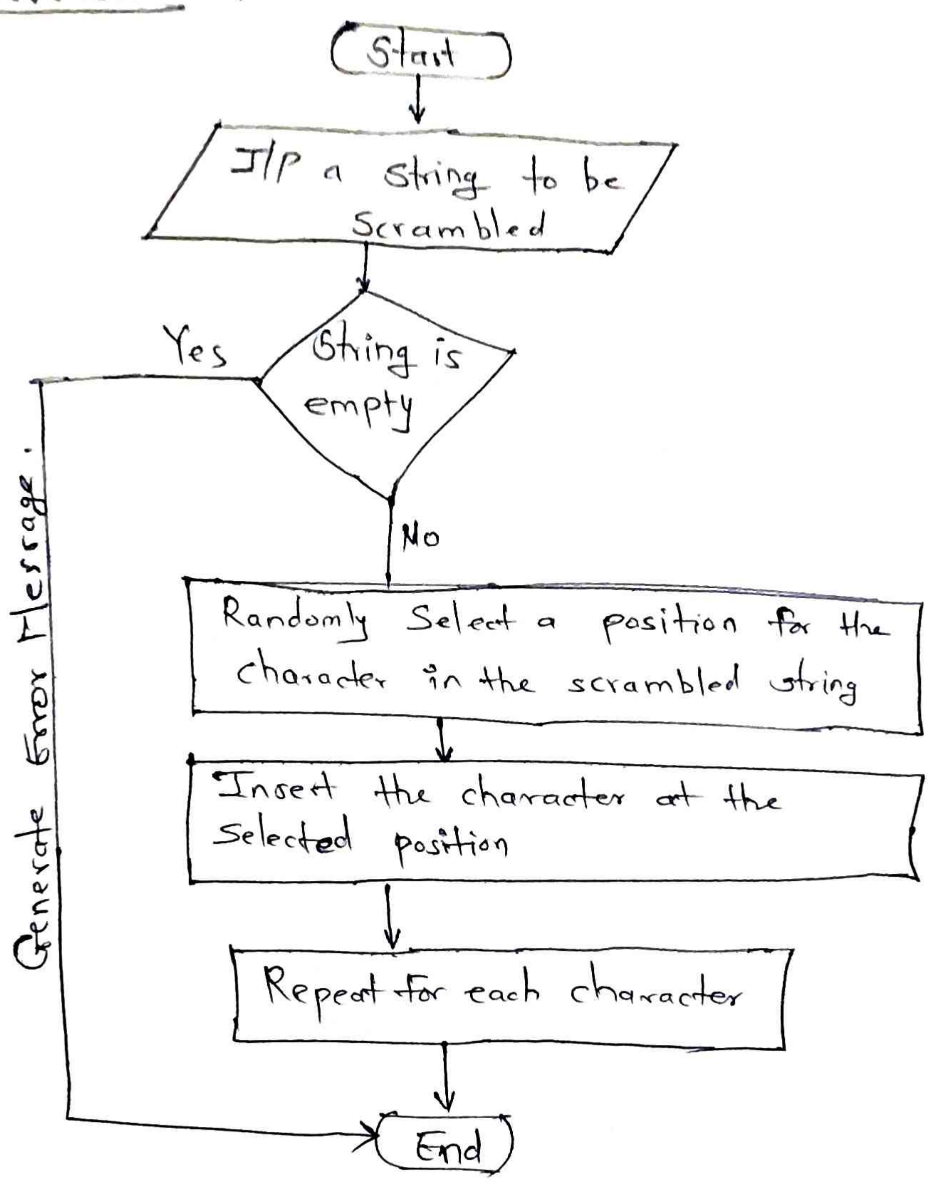
Benefits or advantages of Scrambling

Following are the benefits or advantages of Scrambling:

➡It does not increase data rate unlike block coding technique.

➡It eliminates long string of 0s to provide more transitions in the data. This helps receiver for synchronization to recover the original bit pattern.

Flowchart ⇒.



Algorithm:-

1] Start
2] Input a string to scramble.
3] Check if string is empty.
4] If yes; then Generate Error and end program.
5] If No;
6] then; Randomly select a position for the character in the scrambled string.
7] Insert the character at the selected positn.
8] Repeat for each character.
9] End.

s=bitset(s,15,msb); t=bitxor(bitget(scrambler_out(j),9-i),msb); descrambler_in(j)=bitset(descrambler_in(j),9-i,t); end end descrambler_out=descrambler_in

Write following data for Scrambling and Descrambling:

1. **Flowchart:**
2. **Algorithm:**
3. **Program:**
4. **Input:**
5. **Output (Printouts):**
6. **Result:**

attached.

Conclusion:

Demonstrated Scrambling & Descrambling operation using MATLAB S/W.

27/3/25

Experiment No.12.

Que.1] What is the purpose of Scrambling?

⇒ 1] To randomise the data for efficient transmission.

2] Helps in clock recovery and reduces long sequences of 0s ~~or~~ 1s.

3] Prevents synchronization issue.

4] It may help to randomise the data.

5] For the efficient transmission the randomizing the data.

6] This is the purpose of Scrambling.

Que.2] Is Scrambling known as Encryption?

⇒ • No, Scrambling is not encryption.

• Scrambling is for signal integrity; encryption is for security.

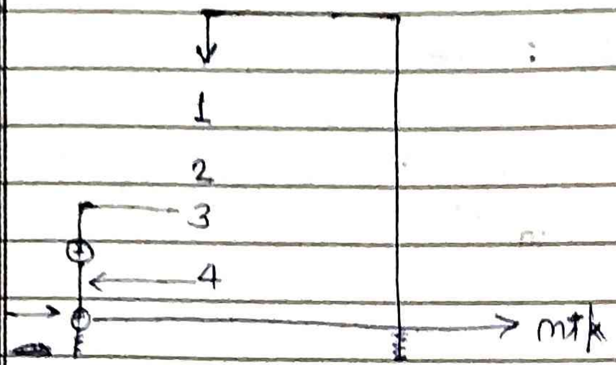
Q.3] ⇒ Define Scrambling & Descrambling

⇒ • Scrambling ⇒
It's a technique that alters data pattern using a predefined algorithm to make it more suitable for transmission.

• Descrambling ⇒
It's the process of restoring the data (original) from the scrambled signal using the same algorithm.

Que 7 Design a Scrambler with seq. given to. 1011000000001. is applied to input of scrambler determine the scrambler o/p.

⟹



Determine the output sequence.

Output $m'k = mk \oplus m''k$ —— ①

but; $m''k = m'k-4 \oplus m'k-3$ —— ②

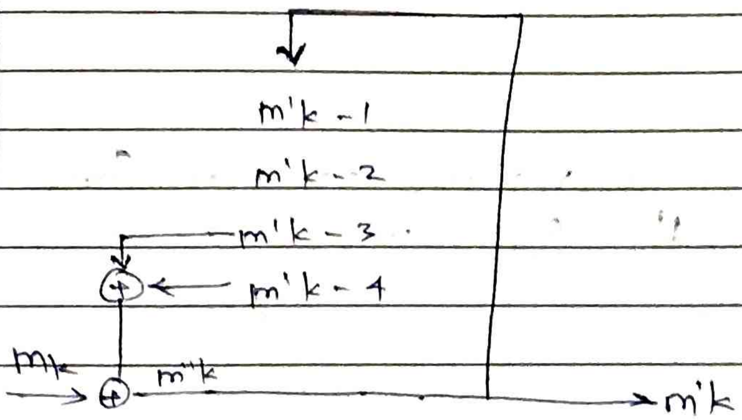
∴ The o/p ⇒

$m'k = mk \oplus m'k-4 \oplus m'k-3$ —— ③

Initially assume that the i/p are in the same condition.

$m'k-4 = m'k-3 = 0$

$m''k = 0$



∴ $m''k = m'k-3 \oplus m'k$.