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LABORATORY MANUAL

SUBJECT: PRINCIPLES OF COMMUNICATION SYSTEMS

[SUBJECT CODE: 204193]

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List of Experiments

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 modulation and angle modulation. Angle modulation can be further characterized as either frequency modulation or phase modulation.

Modulation degree and sideband amplitude

Amplitude modulation of a sine or cosine carrier results in a variation of the carrier amplitude that is proportional to the amplitude of the modulating signal. In the time domain (amplitude versus time), the amplitude modulation of one sinusoidal carrier by another sinusoid resembles figure 1a. The mathematical expression for this complex wave shows that it is the sum of three sinusoids of different frequencies. One of these sinusoids has the same frequency and amplitude as the unmodulated carrier. The second sinusoid is at a frequency equal to the sum of the carrier frequency and the modulation frequency; this component is the upper sideband. The third sinusoid is at a frequency equal to the carrier frequency minus the modulation frequency; this component is the lower sideband. The two-sideband components have equal amplitudes, which are proportional to the amplitude of the modulating signal. Figure 1a shows the carrier and sideband components of the amplitude-modulated wave of figure 1b as they appear in the frequency domain (amplitude versus frequency).

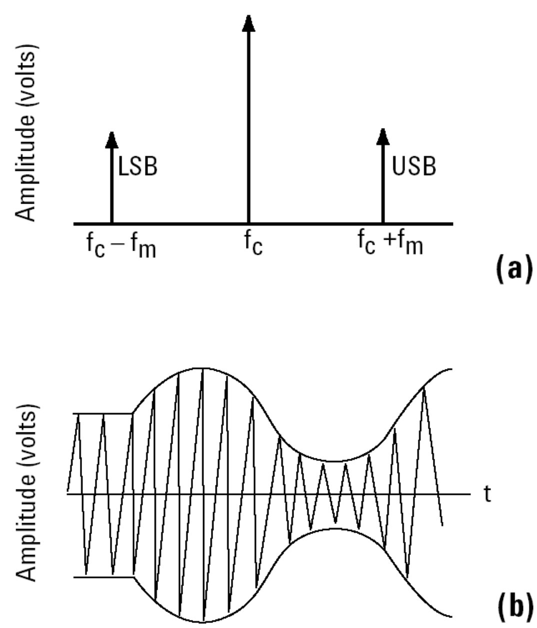


Figure 1. (a) Frequency domain (spectrum analyzer) display of an amplitude-modulated carrier.
(b) Time domain (oscilloscope) display of an amplitude-modulated carrier

A measure of the degree of modulation is m, the modulation index. This is usually expressed as a percentage called the percent modulation. In the time domain, the degree of modulation for sinusoidal modulation is calculated as follows, using the variables shown in figure 2a:

$$m = \frac{E_{max} - E_c}{E_c}$$

Since the modulation is symmetrical,

$$E_{max} - E_c = E_c - E_{min}$$

And

$$\frac{E_{max} + E_{min}}{2} = E_c$$

From this, it is easy to show that:

$$m = \frac{E_{max} - E_{min}}{E_{max} + E_{min}}$$

for sinusoidal modulation.

When all three components of the modulated signal are in phase, they add together linearly and form the maximum signal amplitude Emax, shown in figure 2.

$$E_{max} = E_c + E_{USB} + E_{LSB}$$

$$m = \frac{E_{max} - E_c}{E_c} = \frac{E_{USB} + E_{LSB}}{E_c}$$

and, since $E_{USB} = E_{LSB} = E_{SB}$, then:

$$m = \frac{2E_{SB}}{E_c}$$

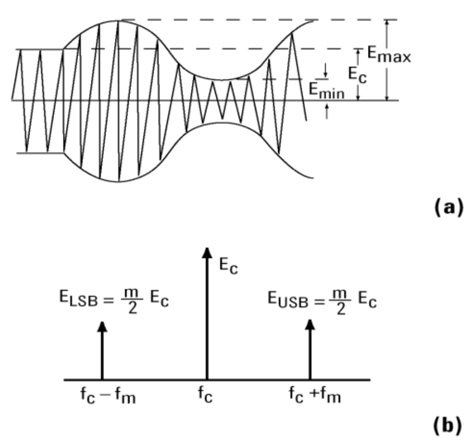


Figure 2(a) (b). Calculation of degree of amplitude modulation from time domain and frequency domain displays

For 100% modulation (m = 1.0), the amplitude of each sideband will be one-half of the carrier amplitude (voltage). Thus, each sideband will be 6 dB less than the carrier, or one-fourth the power of the carrier. Since the carrier component does not change with amplitude modulation, the total power in the 100% modulated wave is 50% higher than in the unmodulated carrier.

Block Diagram:

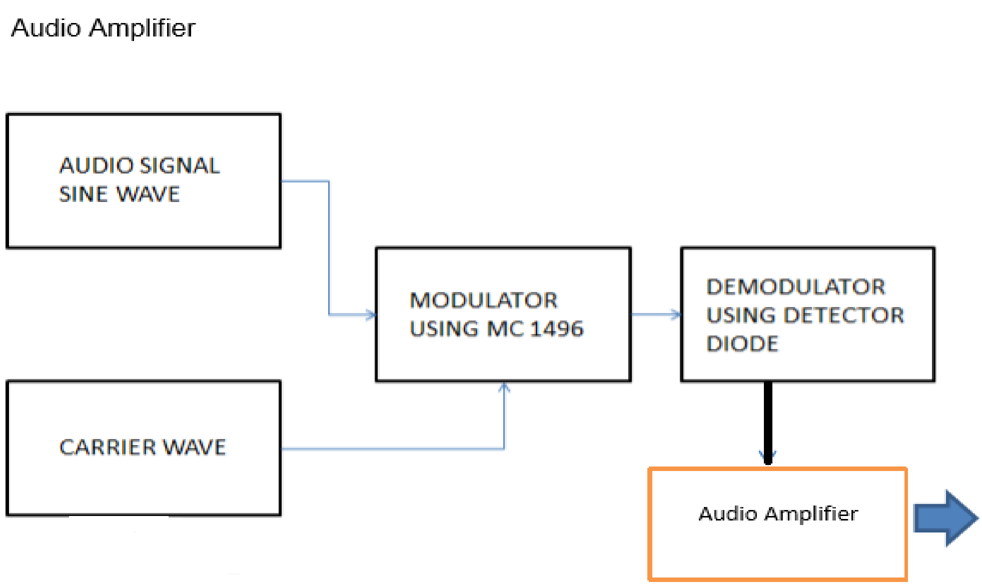


Fig. Amplitude Modulator and Demodulator

Procedure:

To generate AM Double Side band Full Carrier (DSB-FC) signal

1.Connect patch cord between Sin O/P and Mod I/P of balanced modulator And Now Connect patch cord between carrier O/P and Carrier I/P of Balanced Modulator.

2. Connect CRO Channel-1 At Sine Output terminal of Sine Wave Generator Adjust frequency to 700Hz and Amplitude 3 Vpp.

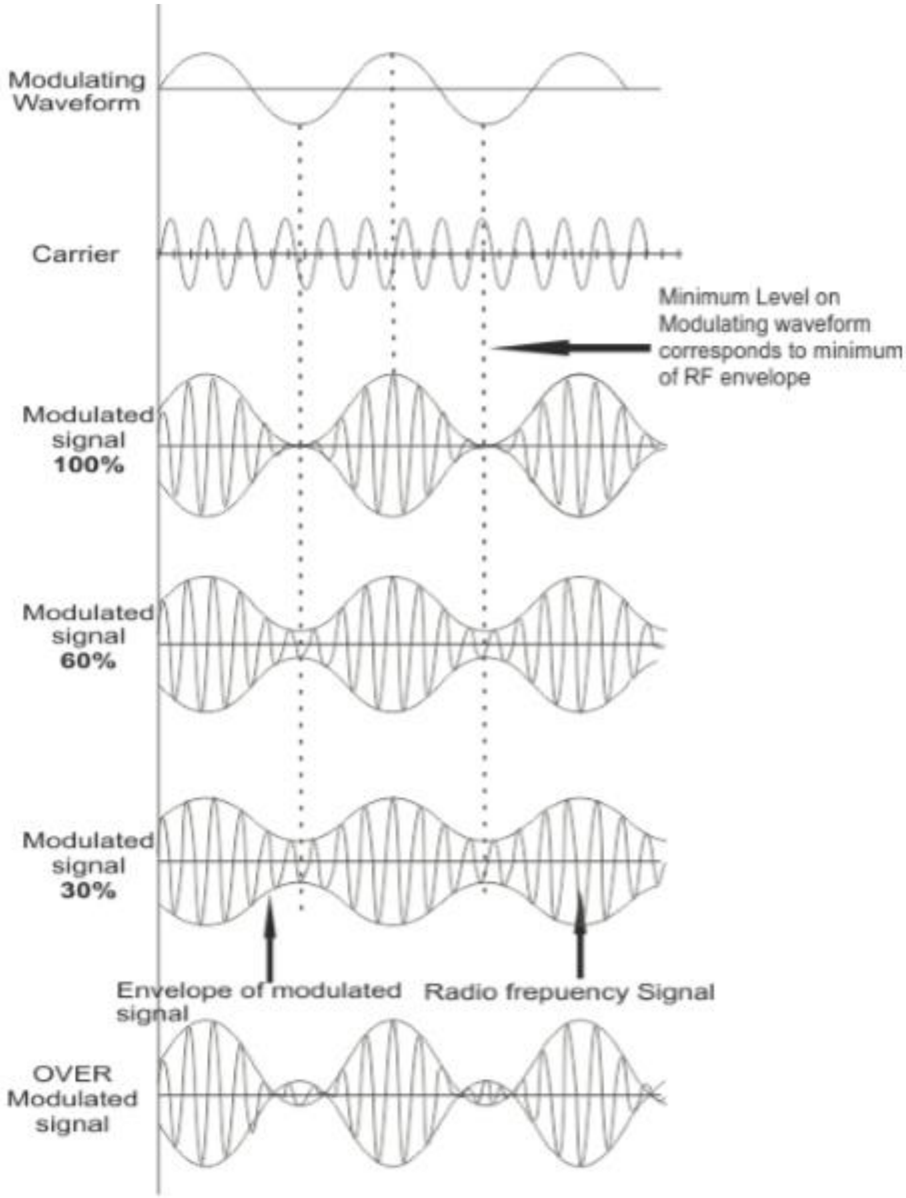
3. Connect CRO Channel-2 at Carrier Output terminal of Carrier Generator Adjust output frequency of carrier generator to 100khz /200 Khz and Amplitude to 3 Vpp.

4. Now Connect CRO Channel-2 AM MOD O/P terminal of balanced Modulator. And Modulated Wave will be observed.

5. Connect patch cord between Modulator O/P and Detector, Observe final output. .

6. Same Procedure for DSB MODULATION (keep Select switch on DSB)

Different Types of Modulation based on Modulation Index:m=1,m>1,m<1

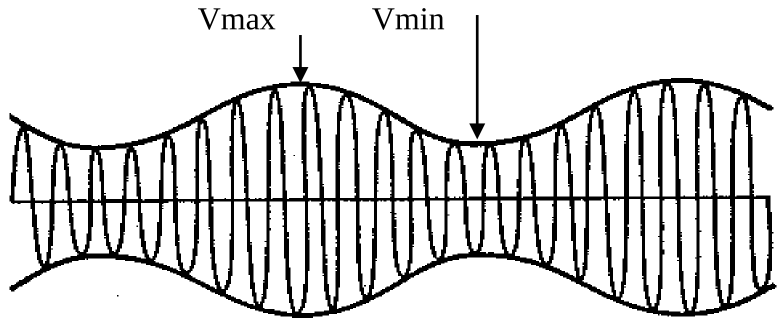


[To calculate modulation index by graphical method]()

1. In the AM modulated waveform T3 measure B and V amplitudes on CRO.

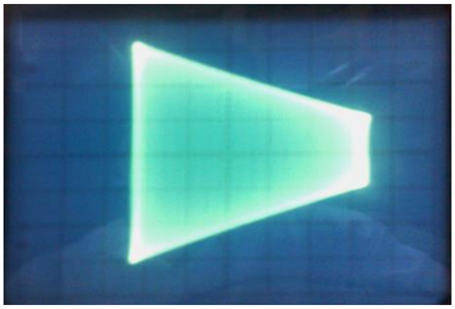
   Then calculate modulation Index by following formula:

$$M = \frac{B}{V} \times 100 = \frac{V_{max} - V_{min}}{V_{max} + V_{min}} = \frac{5.5 - 1.5}{5.5 + 1.5} = \frac{4}{7} = 0.57 \times 100 = 57\%$$



2. **Modulation index by Trepezoidal method.**

With keeping modulating signal and Y1 and RF signal at Y2 (same setup as previous), put horizontal scale at X-Y position. Observe trapezoidal waveform on CRO as under.



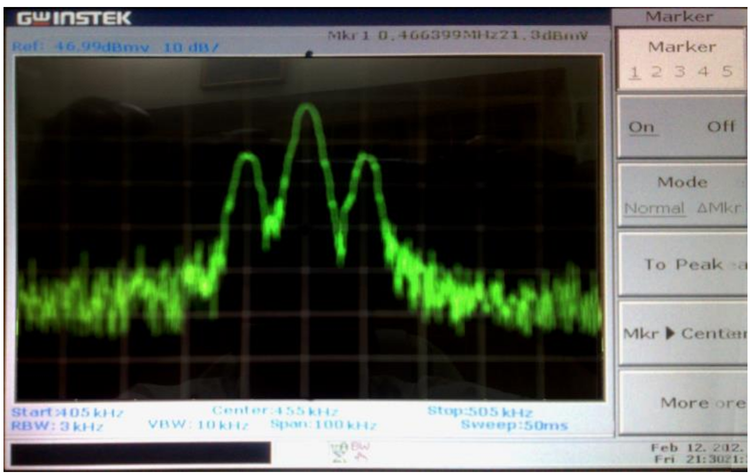
Here left side amplitude of trapezoidal waveform is Vmax and right side amplitude is Vmin.

To calculate power of AM Wave for different modulating signal using Spectrum analyser

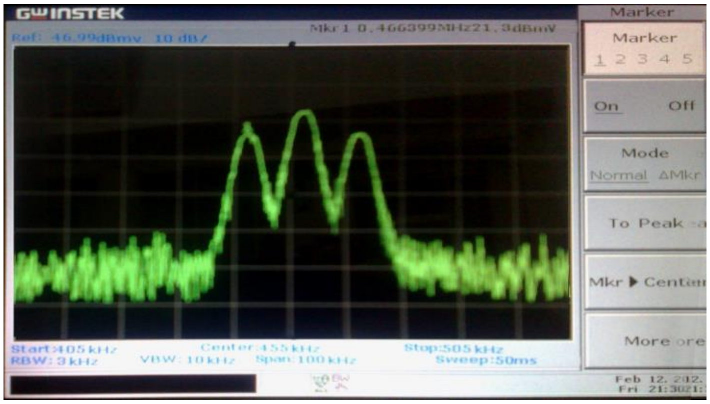
Spectrum Display and Measurement

1. Connect following links as per connection diagram .

2. Connect AM modulated output to the input of Spectrum analyzer. Observe AM spectrum as under.Here modulation index is 50%.



3. Reduce modulation index and observe AM spectrum as under.



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		
2	Carrier signal		
3	Modulated signal		
4	Demodulated Signal		

Graphical Methods

	Test points	Graphical Display	Modulation Index
Sr. No.		Voltage(V)	
1	Vmax (Maximum Modulating voltage)		
2	Vmin (Minimum Modulating voltage)		

Trapezoidal Display

	Test points	Trapezoidal Display	Modulation Index
Sr. No.		Voltage(V)	
1	L1		
2	L2		

Sr. No.	Parameter	Graphical Display	Trapezoidal display
1	Modulation Index		

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		
2	USB		
3	LSB		

Conclusion:

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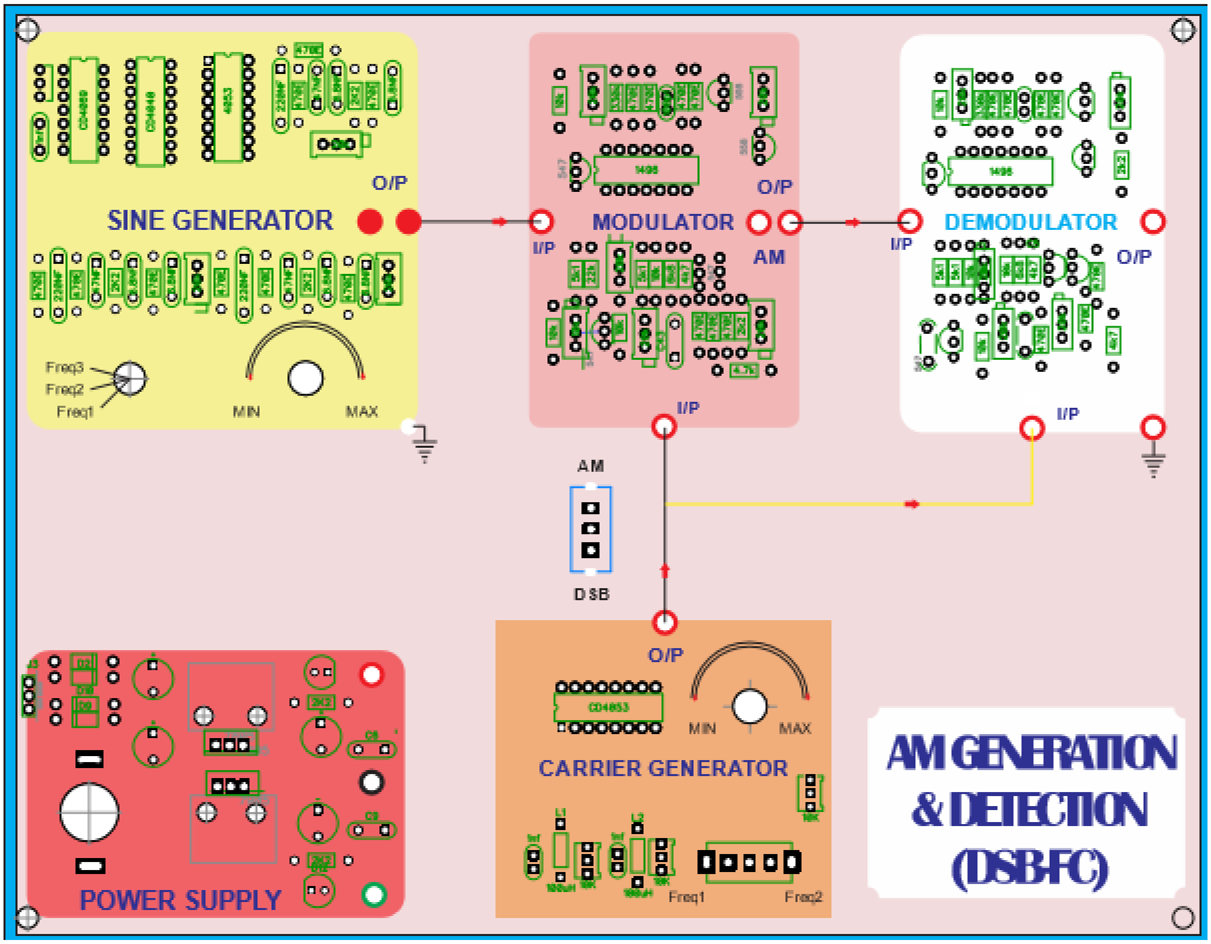
Questions:
1. Define Modulation? Need of Modulation?
2. Write mathematical expression for AM?
3. Application of AM?
4. Difference between DSB-FC,DSB-SC,SSB,VSB?
5. Advantages and disadvantages of AM?
6. Draw waveforms and spectrum of AM on graph paper?

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

Circuit Diagram



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:

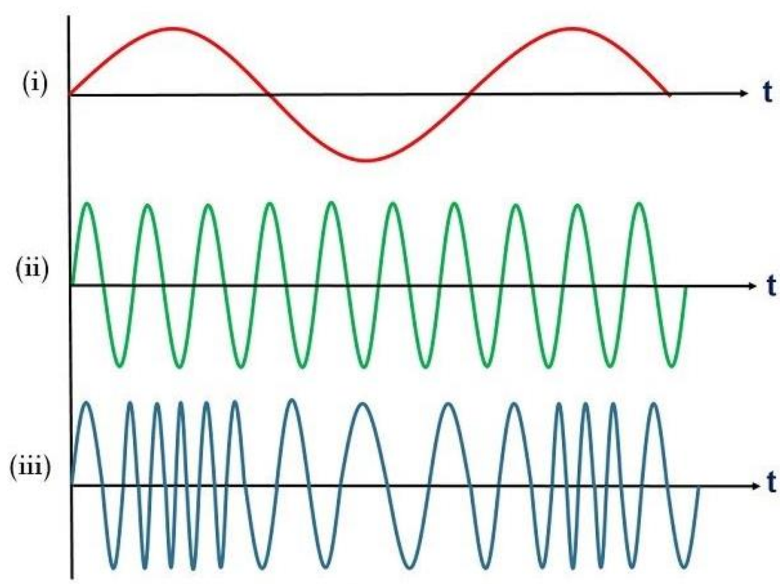
1. To understand generation of frequency modulated Signal.
2. To understand demodulation of FM.
3. Calculation of Modulation Index & BW of FM.

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

$$m(t) = V_m \cos(2\pi f_m t)$$

and the carrier wave is given by

$$c(t) = V_c \sin(\omega_c t + \phi)$$

It is noteworthy that in FM, the frequency changes according to the modulating voltage.

Hence, instantaneous frequency is given as,

$$f_i(t) = f_c + k_f \,.\, m(t)$$

Substituting the value of the modulating signal in the expression shown above,

Substituting the value of the modulating signal in the expression shown above,

$$f_i(t) = f_c + k_f \,.\, V_m \cos(2\pi f_m t)$$

$$f_i(t) = f_c + \Delta f \cos(2\pi f_m t)$$

$$: \Delta f = k_f \,.\, V_m$$

Here, Δf denotes the frequency deviation.

The **extent of the occurred change in frequency** of the carrier wave is termed as the **deviation**. It is noteworthy that this deviation is made according to the voltage of the modulating signal and does not depend on the modulating frequency, which can be easily understood by the expression of frequency deviation.

Thus, we can conclude that the maximum frequency of an amplitude modulated wave is given as,

$$f_{max} = f_c + \Delta f$$

Modulation index

It is defined as the ratio of frequency deviation to the modulating signal frequency. Thus, is given by the expression,

$$m_f = \frac{\Delta f}{f_m}$$

When the frequency deviation is constant, then due to inverse relation, with the increase in modulating frequency, modulation index will decrease.

CIRCUIT DETAILS OF FM GENERATION & DETECTION

The FM Generation & Detection System consists of following sections.

1. Modulating Audio Signal Generator
2. Amplitude Limiter/Mixer
3. VCO FM Modulator section
4. FM demodulation by PLL
5. Power supply.

Procedure:

1. Observe SINE Wave in sine Generator section
2. Observe carrier Wave in modulator section.
3. Connect sine wave section to the modulator section.
4. Varie the modulating signal (amplitude & Frequency), & Observe the FM wave form.
5. Connect Modulator Section to Demodulator (PLL) section. Observe Wave Form.
6. Connect demodulator section to the Low pass filter.
7. Observe the final O/P on CRO.

Spectrum Display and Measurement

1. Connect links as per connection diagram
2. Adjust amplitude of sine wave to 0 Vpp and audio Frequency to 100 KHz.
3. Connect FM transmitted output to the input of Spectrum analyser. Observe FM spectrum as under.

3 Now vary amplitude of sine wave to 2 Vpp and observe FM spectrum.

4. Vary amplitude of input modulating signal and observe change in spectrum.

To see the effect of Eigen values on carrier power using Spectrum analyser

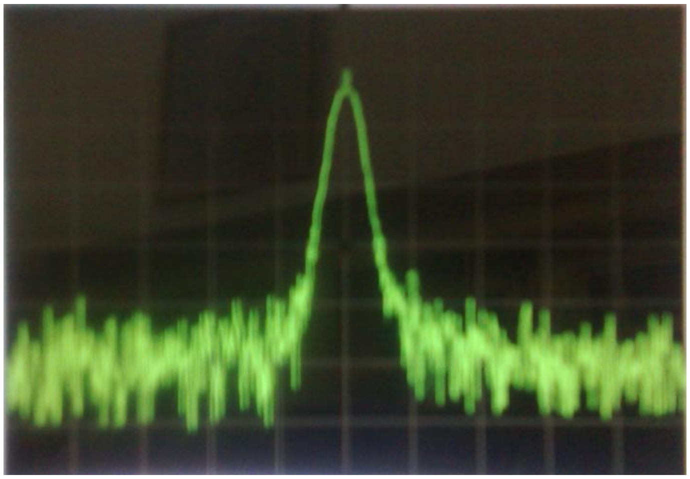
Spectrum Setup

1. Resolution Bandwidth : 3 KHz or less available particular model of Spectrum analyser
2. Center Frequency : 455 KHz
3. Span : 100KHz
   Hence Start of Span : 405 KHz Automatically decided by Span and Centre frequency
   End of Span : 505 KHz
4. Step : 5 KHz
5. Markers : ON
   Marker 1 : 435KHz
   Marker 2 : 445KHz

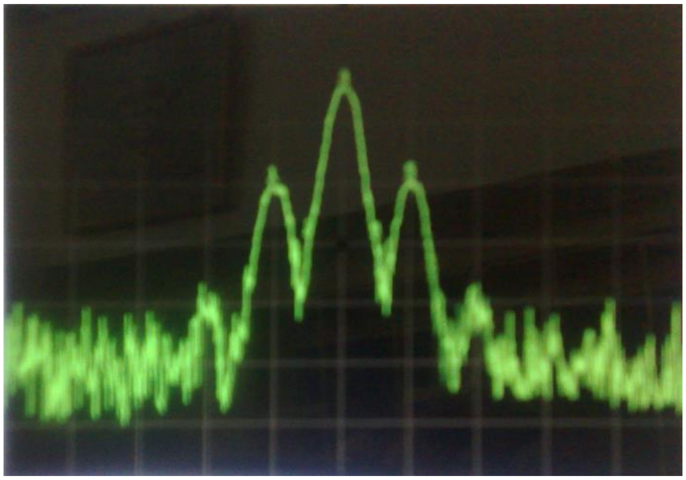
Marker 3	:	455KHz
Marker 4	:	465KHz
Marker 5	:	475KHz
6. Amplitude Unit	:	dbmv
7. System Save Setup	:	Position 6

Spectrum Display and Measurement

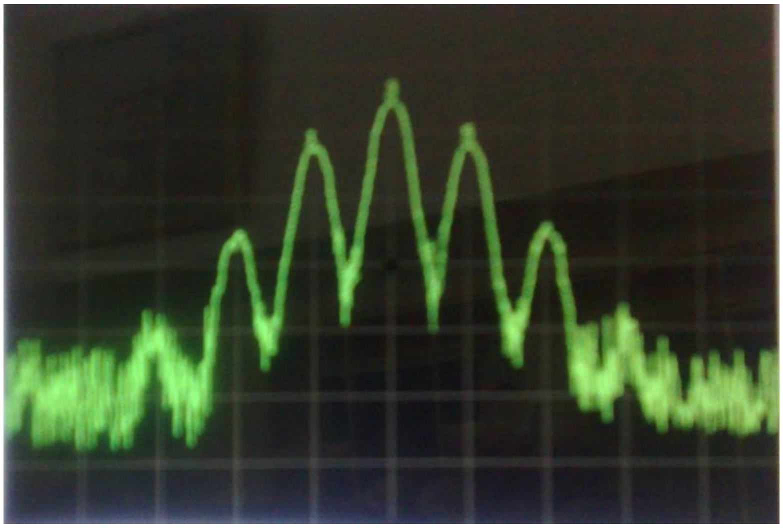
1. Connect following links as per connection diagram CN3.
2. Adjust amplitude of sine wave to 0.0 and audio Frequency to 10 KHz.
3. Connect FM modulated output of VCO modulator to the input of Spectrum analyser.
4. Observe AM spectrum as under. This is only FM carrier.



5. Now change to amplitude of sine wave to 0.3V and observe spectrum.

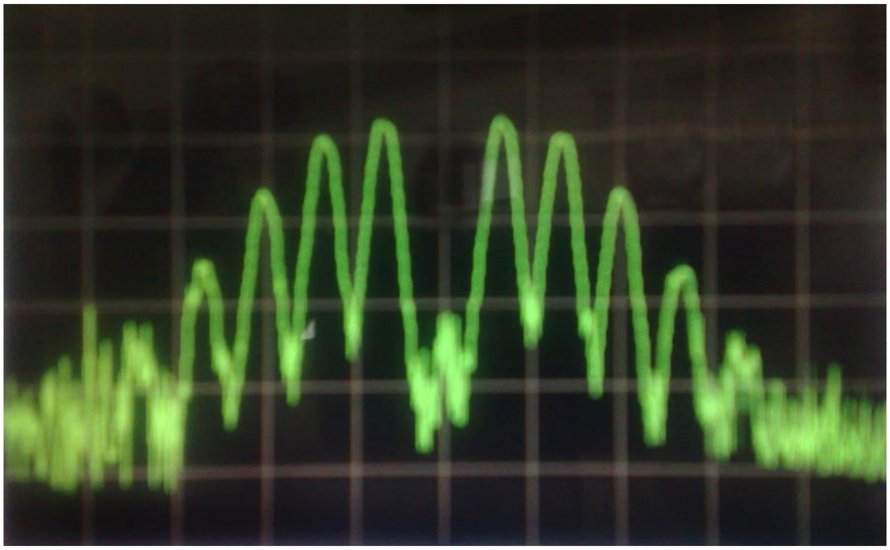


6. Now change to amplitude of sine wave to 0.5V and observe spectrum.

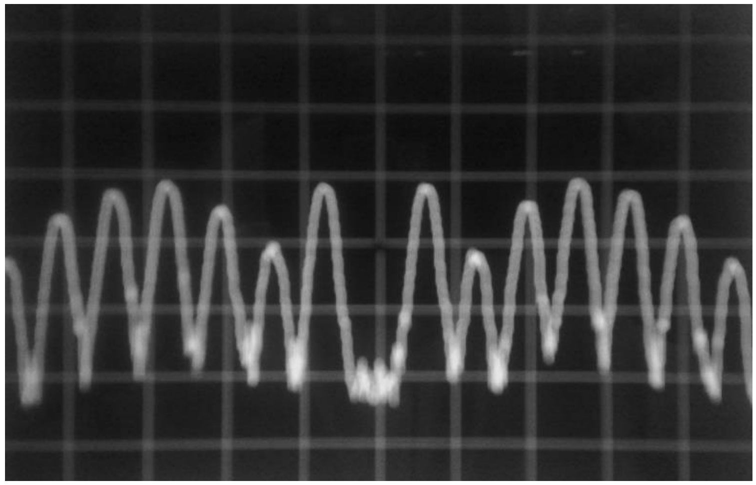


EIGEN VALUE PRACTICAL

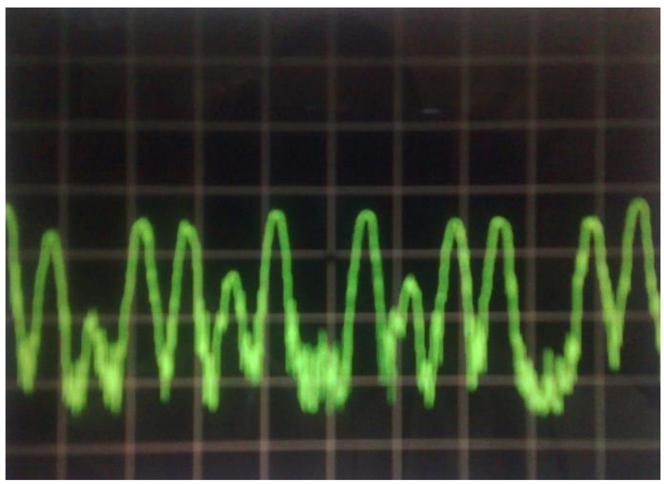
1. The Eigen values are the values of fm modulation index when carrier amplitude in FM spectrum becomes
   zero. To find Eigen value change amplitude of modulating signal such that FM carrier amplitude in spectrum become zero. Measure that amplitude of modulating signal and then calculate modulation index as per Eq. (1) given in exp. 3.
2. Measured amplitude of modulating signal for first Zero as for spectrum shown below is 0.65V.



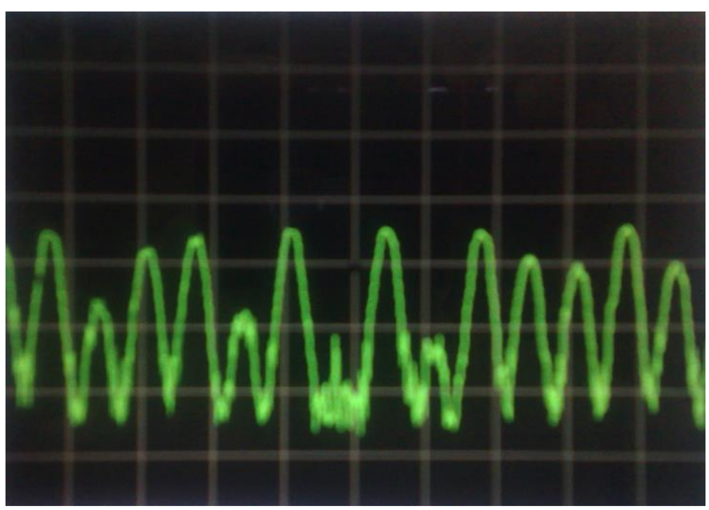
3. Measured amplitude of modulating signal for second Zero as for spectrum shown below is 1.50V.



4. Measured amplitude of modulating signal for third Zero as for spectrum shown below is 2.35V.



5. Measured amplitude of modulating signal for 4th Zero as for spectrum shown below is 3.20V.



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**		
2	**Carrier Signal**		
3	**Demodulator Output**		
4	**LPF Output**		
5			

Sr. No.	Modulating Voltage		Frequency	Power
1		Carrier		
2		Carrier		
		USB1		
		LSB1		
3		Carrier		
		USB1		
		LSB1		
		USB2		
		LSB2		
4		Carrier		
		USB1		
		LSB1		
		USB2		
		LSB2		
		USB3		
		LSB3		

Conclusion:

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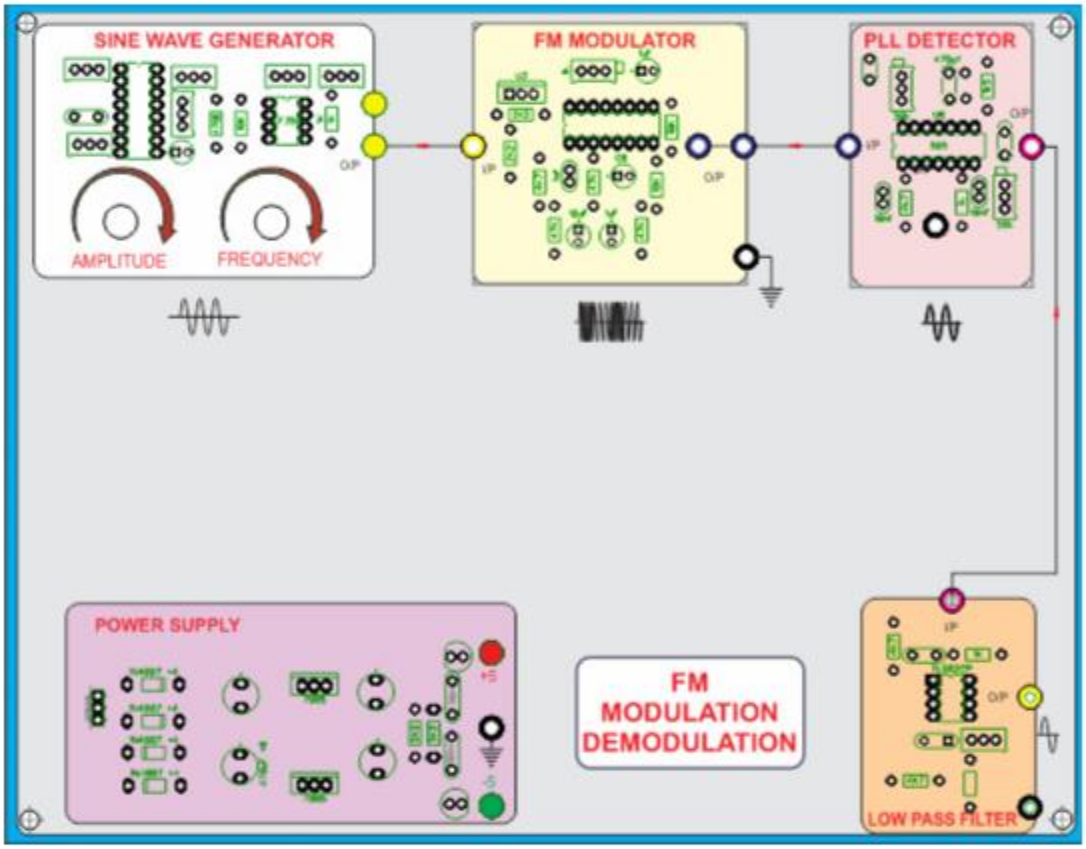
Questions:

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?

FM MODULATION & DEMODULATION



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:

$$Y = \frac{\sin X}{X}$$

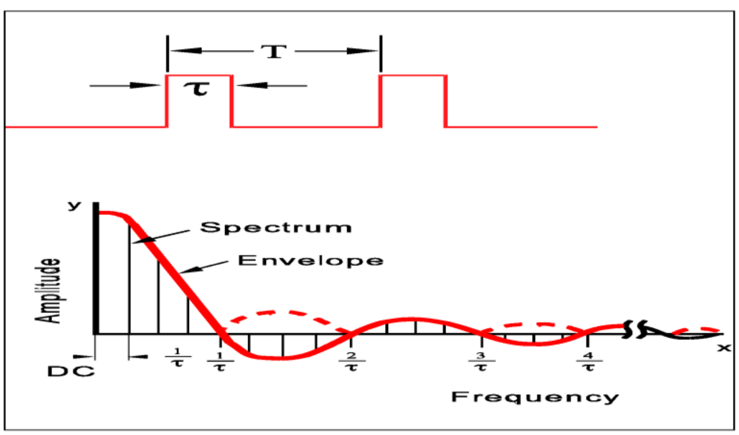
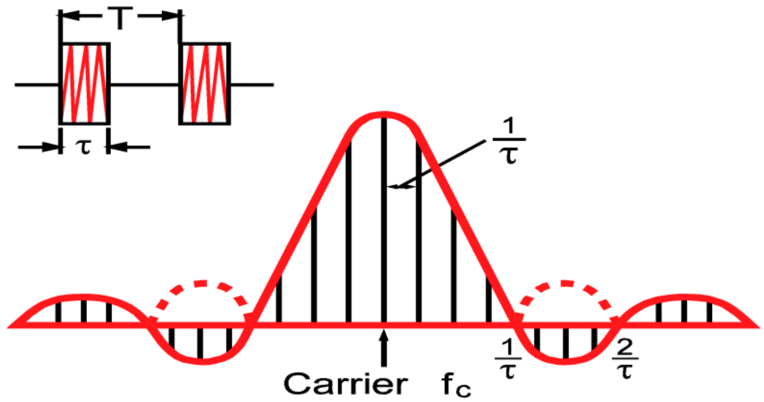


Figure - 1

Figure-2 shows the spectral plot resulting from rectangular amplitude pulse modulation of a carrier. The individual lines represent the modulation product of the carrier and the modulation pulse repetition frequency with its harmonics. Thus, the lines will be spaced in frequency by whatever the pulse repetition frequency might happen to be.



We know from single tone AM how the sidebands are produced above and below the carrier frequency. The idea is the same for a pulse, except that the pulse is made up of many tones, thereby producing multiple sidebands, which are commonly referred to as spectral lines on the analyzer display. In fact, there will be twice as many sidebands (or spectral lines) as there are harmonics contained in the modulating pulse.

The mainlobe (in the center) and the sidelobes are shown as groups of spectral lines extending above and below the baseline. For perfectly rectangular pulses and other functions whose derivatives are not continued at some point, the number of sidelobes is infinite.

The mainlobe contains the carrier frequency and is represented by the longest spectral line in the center. The amplitude of the spectral lines forming the lobes varies as a function of frequency.

Notice in Figure 36 how the spectral lines extend below the baseline as well as above. This corresponds to the harmonics in the modulating pulse having a phase relationship of 180% with respect to the fundamental of the modulating waveform. Since the spectrum analyzer can only detect amplitude and not phase, it will invert the negative-going lines and display all amplitudes above the baseline.

Because a pulsed RF signal has unique properties, care must be taken to interpret the display on a spectrum analyzer correctly. The response that the spectrum analyzer (or any swept receiver) can have to a periodically pulsed RF signal can be of two kinds, resulting in displays, which are similar, but of completely different significance. One response is called a line spectrum and the other is a pulse spectrum. We must keep in mind that these are both responses to the same periodically pulsed RF input signal and that line and pulse spectrum refer only to the response displayed on the spectrum analyzer.

Line Spectrum

A line spectrum occurs when the spectrum analyzer IF bandwidth (B) is narrow compared to the frequency spacing of the input signal components. Since the individual spectral components are spaced at the pulse repetition frequency (PRF) of the pulsed RF, we can say: B < PRF

In this case all individual frequency components can be resolved since only one is within the bandwidth at a time as shown in Figure-3. The display is a frequency domain display of the actual Fourier components of the input signal. Each component behaves as a CW signal and the display has the normal true frequency domain characteristics.

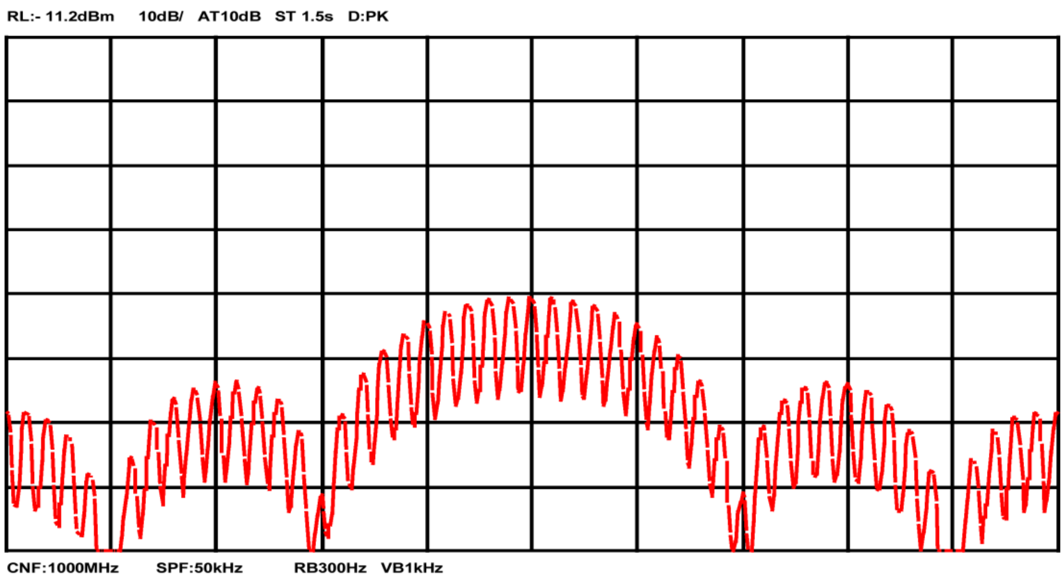


Figure - 3

Pulse Response

If we increase the IF bandwidth in our example to 1 kHz, we get the display shown in Figure-4. Notice that the analyzer has lost the ability to resolve the spectral lines since B = PRF. The lines now displayed are generated in the time domain by the single pulses of the signal. We also see that the displayed amplitude of the spectrum envelope has increased. This is due to the fact that the IF filter is now sampling a broader section of the spectrum, thus collecting the power of several spectral lines.

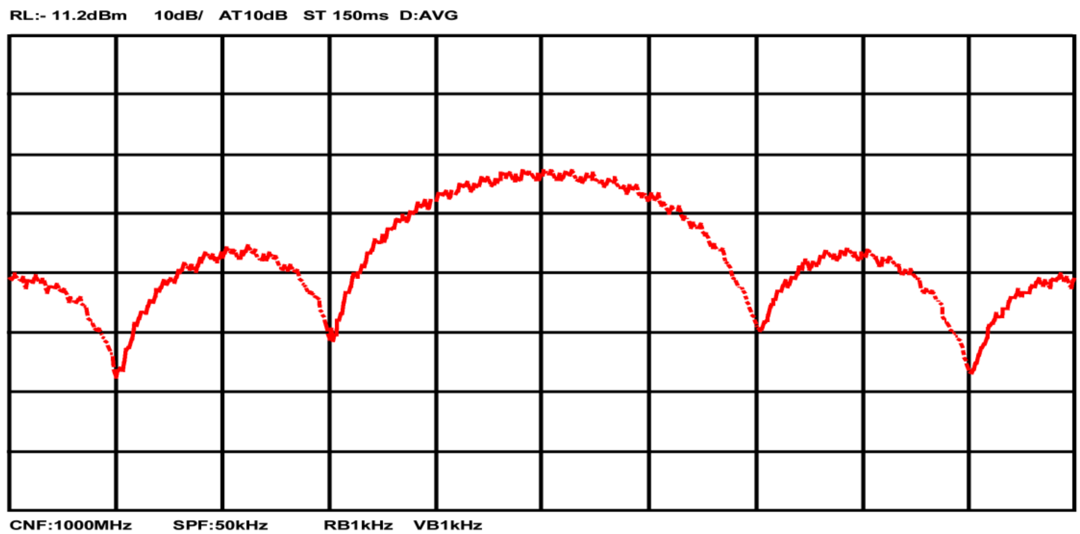


Figure-4

A pulse repetition rate equal to the resolution bandwidth is the demarcation line between a true Fourier-series spectrum, where each line is a response representing the energy contained in that harmonic and a pulse of the Fourier-transform response.

Pulse Spectrum

A pulse spectrum occurs when the bandwidth B of the spectrum analyzer is equal to or greater than the PRF. The spectrum analyzer in this case cannot resolve the actual individual Fourier frequency domain components, since several lines are within its bandwidth. However, if the bandwidth is narrow compared to the spectrum envelope, then the envelope can be resolved. The resultant display is not a true frequency domain display, but a combination of time and frequency domains. It is a time domain display of the pulse lines, since each line is displayed when a pulse occurs, regardless of the frequency within the pulse spectrum to which the analyzer is tuned at that moment. It is a frequency domain display of the spectrum envelope.

CIRCUIT DESCRIPTION OF SAMPLING THEOREM

The sampling theorem circuit consists of following stages/sections.

1. Modulating Audio Signal Generator
2. Sampling Pulse Generator
3. Sampling section
4. Low pass filter demodulator section.
5. Power supply.

The summary of different type of samplings is shown below: -

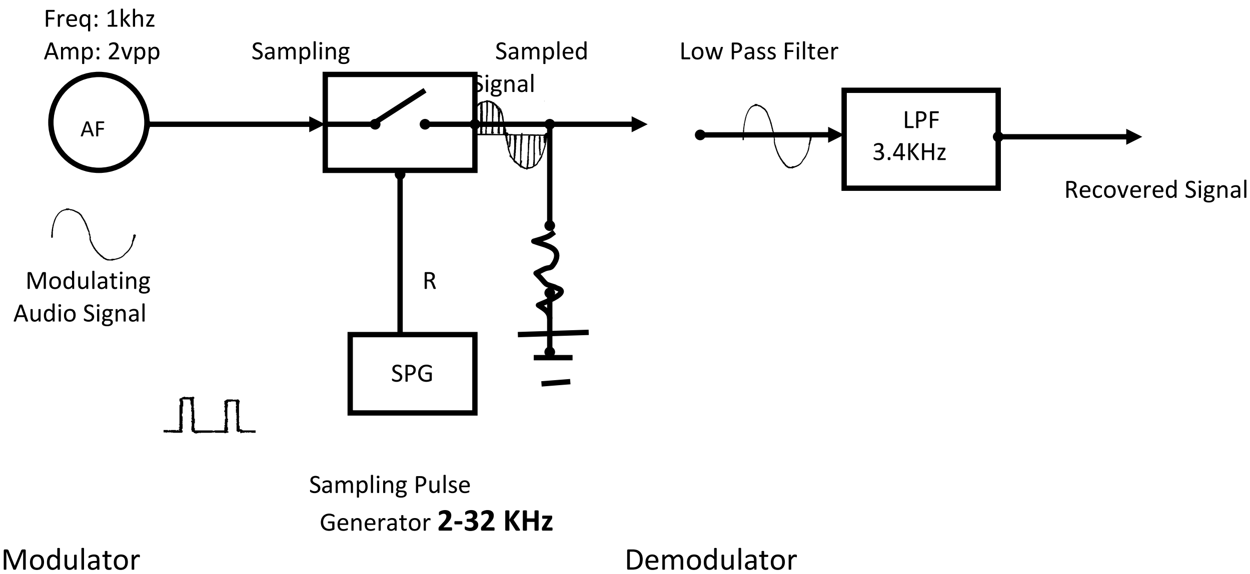
Types	Jumper J1	Jumper J2	Jumper J3
1. Natural Sampling	Closed	Open	Open
2. Sample/hold	Open	Closed	Open
3. Flat Top Sampling	Open	Closed	Closed

(4) Demodulator section:

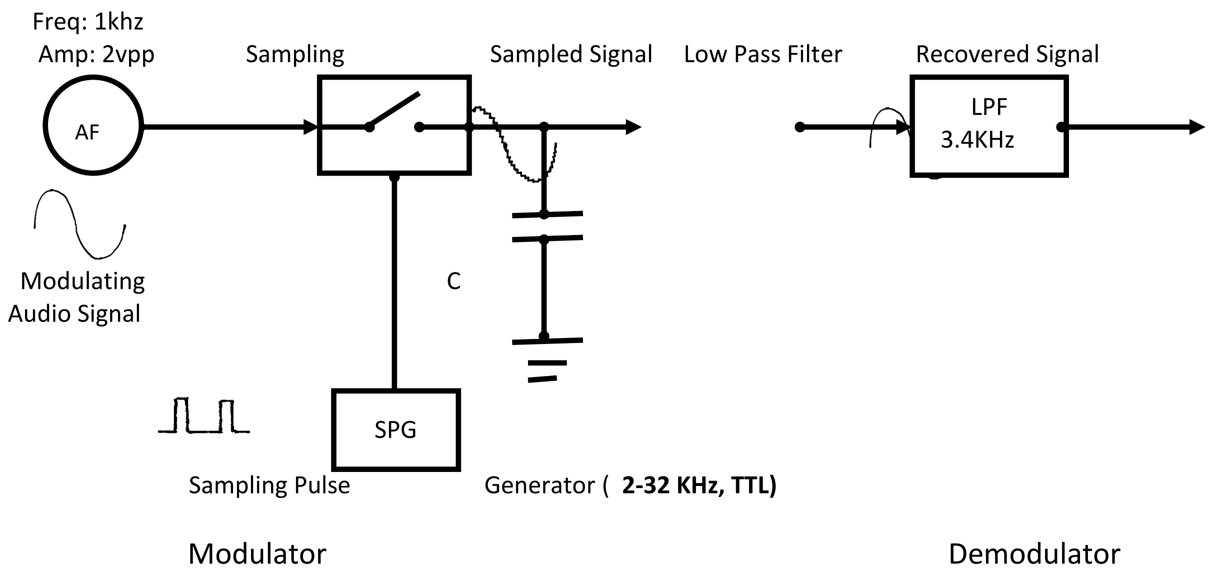
This section is based on low pass filter. It is based on two 741 IC and R-C circuits. It passes only low frequencies up to 3.4 KHz and reduces all other frequencies. Thus it removes high frequency Quantization noise of sampled signals and original modulating signal is recovered.

Block diagram:

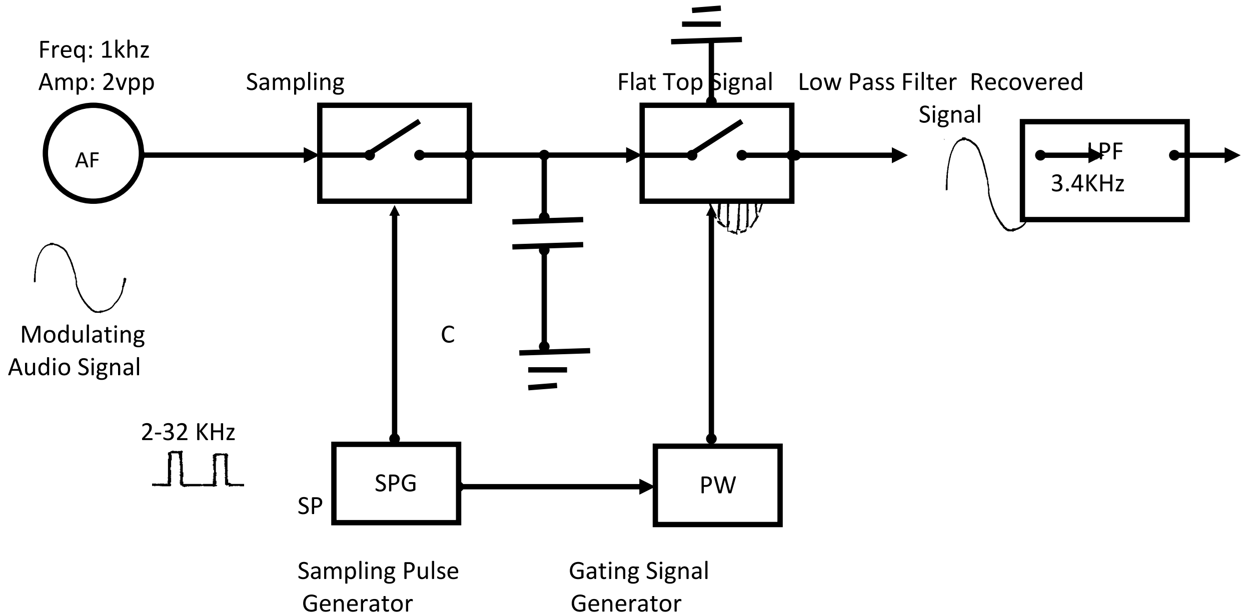
(1) NATURAL SAMPLING:



(2) SAMPLE AND HOLD:



(3) FLAT TOP SAMPLING: -



Procedure:

To study Natural Sampling PAM Technique, to reconstruct original signal using Interpolation filter, To study effect of filter cutoff frequency and to study effect of variable sampling rate

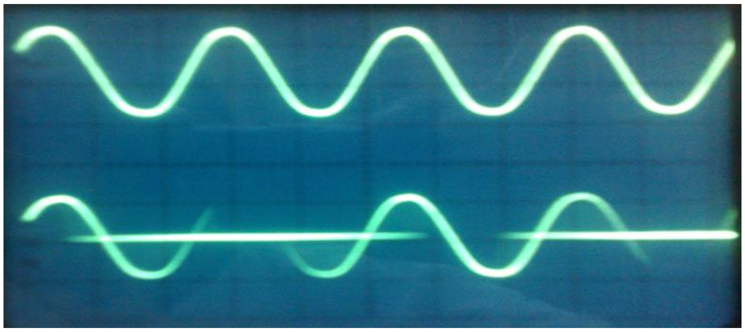
1. Connect following links as per Connection diagram CN1
2. Connect CRO channel-1 at **Sine O/P** terminal of Audio Generator. Adjust amplitude of sine wave to 4 Vpp and audio Frequency to 5 KHz.
3. Connect CRO channel-2 at **SAMP CLK** terminal of Sampling Pulse generator. Set frequency to 30KHz.
4. Connect CRO Channel 2 at PAM O/P terminal of modulator. Trigger CRO by channel-1. The Natural sampling PAM signal will be observed.
5. Then connect CRO Channel-2 at demodulated output DEMOD O/P of Interpolation Low pass filter. Observe recovered sine wave signal.
6. Vary frequency of sine wave modulating signal and observe its effect on PAM output as well as on recovered signal. With increasing frequency recovered output of filter will reduce as signal is beyond its cutoff frequency.
7. Vary frequency of sampling pulse signal and observe its effect on PAM output as well as on recovered signal. With increasing sampling frequency, the error in recovered signal decreases. The error in recovered signal increases with decrease in sampling pulse frequency.

TO VERIFY NYQUIST'S SAMPLING THEOREM (Time Domain)

Keep modulating sine wave frequency to 5 KHz and amplitude 4Vpp. Vary sampling frequency slowly from 32KHz to 10KHz by observing original signal and recovered demodulated signal. Measure the sampling frequency for which original signal and recovered demodulated signal are nearly same i.e. error is less. It will be more than 10 KHz, which proves Nyquist's Sampling Theorem.

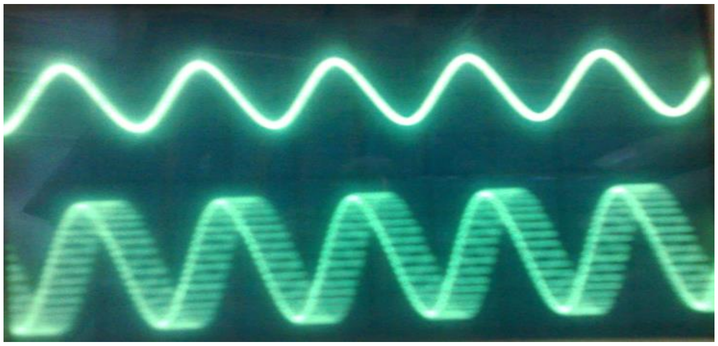
ALIASING EFFECT

1. Connect following links as per connection diagram CN1. (Same setup as experiment No. 1).
2. Keep modulating sine wave frequency to 20 KHz and amplitude 2Vpp.
3. Now vary sampling frequency slowly to 20 KHz. Observe the output at PAM O/P terminal. The slowly varying waveform will be seen. This is known as “Aliasing effect”



To study Flattop Sampling PAM Technique

1. Connect following links as per Connection diagram
2. Connect CRO channel-1 at Sine O/P terminal of Audio Generator.
3. Vary Pulse width pot of sampling pulse generator slowly and observe Flattop sampled PAM signal at particular setting as under.



Observation Table:

Sr. No.	Test points	Frequency(Hz)	Voltage(V)
1	Sine wave generator output		
2	Sampling Pulse generator output		
3	Output of sampler(Natural)		
4	Output of sampler(Flat top)		
5	Output of demodulator (LPF)		

Conclusion:

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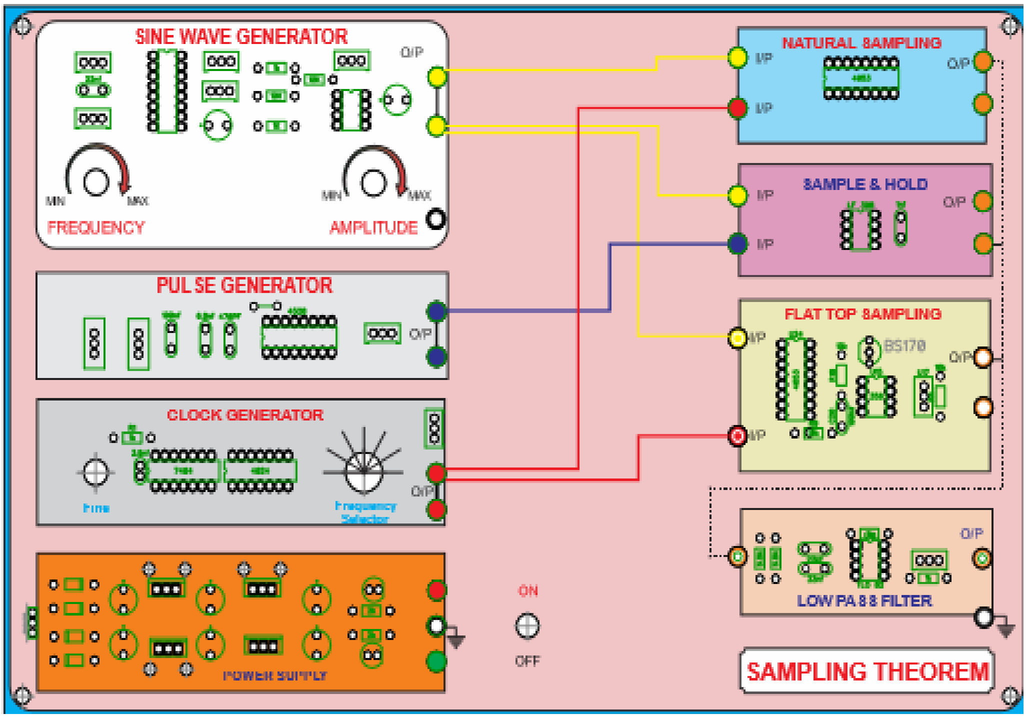
Oral Questions:

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?

VERIFICATION OF SAMPLING THEOREM



Connection diagram

EXPERIMENT NO.4

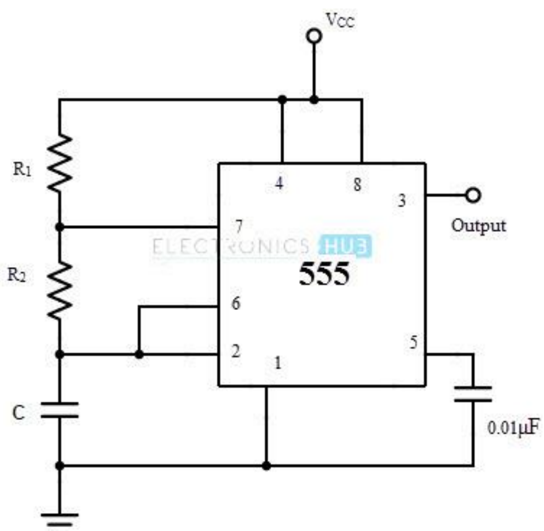
AIM: Generation and Detection of PWM using IC 555

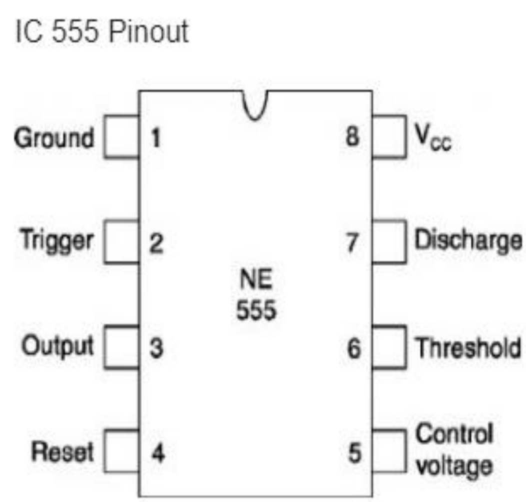
OBJECTIVE: Study the PWM Generation & Detection using IC 555.

APPARATUS/SW: Trainer kit of PWM, DSO.

THEORY:

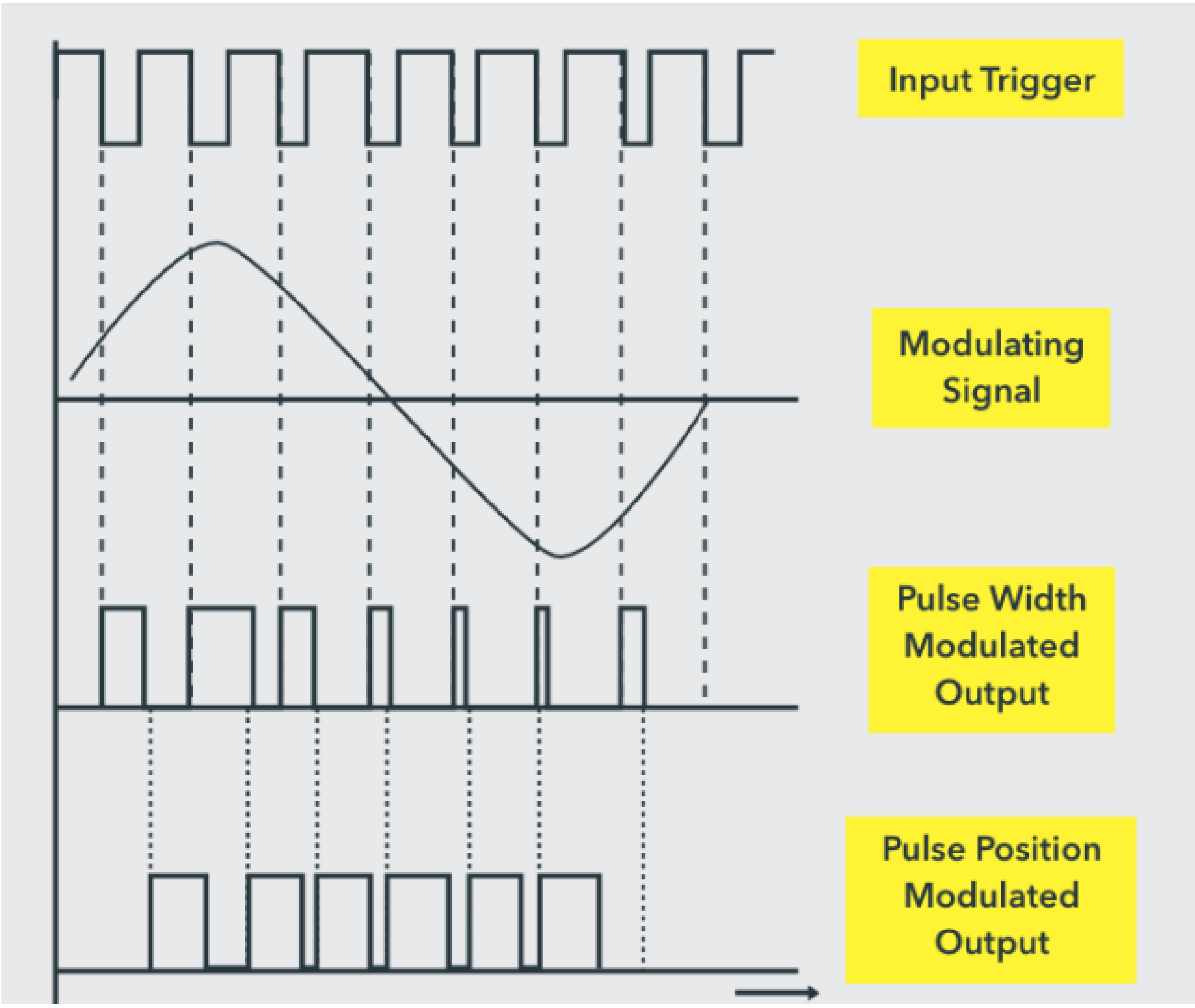
PWM (Pulse Width Modulation) is one of the modulation techniques in which the width of the carrier wave varies with the amplitude of the message signal. In this technique the pulse is used as a carrier signal and the message signal can be any analog signal. Hence as the width is changing it can be used 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.

The control voltage applied at pin 5 will change the threshold voltage level. But for normal use, pin 5 is connected to ground via a capacitor (usually 0.01μF), so the external noise from the terminal is filtered out. Pin 1 is ground terminal. The timing circuit that determines the width of the output pulse is made up of R1, R2 and C.



Procedure:

1. The circuit wiring is done as shown in connection diagram.
2. The amplitude and the time duration of the modulating signal are observed using CRO or DSO.
3. A modulating signal is given to the PWM Generator and observe the output.
4. Finally, the PWM output is observed and the amplitude and time duration of pulse with respect to modulating signal are noted down.
5. PWM signal is applied to the PWM Detector and amplifier circuit for demodulation process.
6. After demodulation, the original signal is recovered.

Observation Table:

Test points	Voltage	Frequency

Conclusion:

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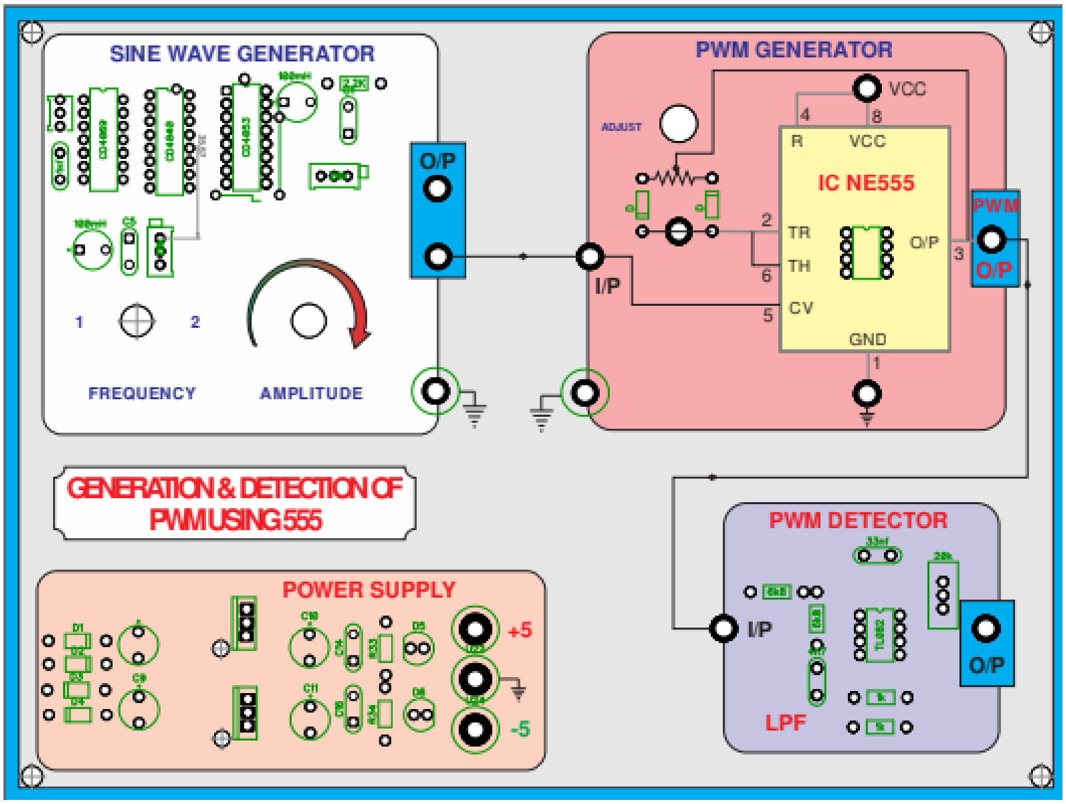
Oral Questions:

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

PWM using IC 555

CONNECTION DIAGRAM



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.

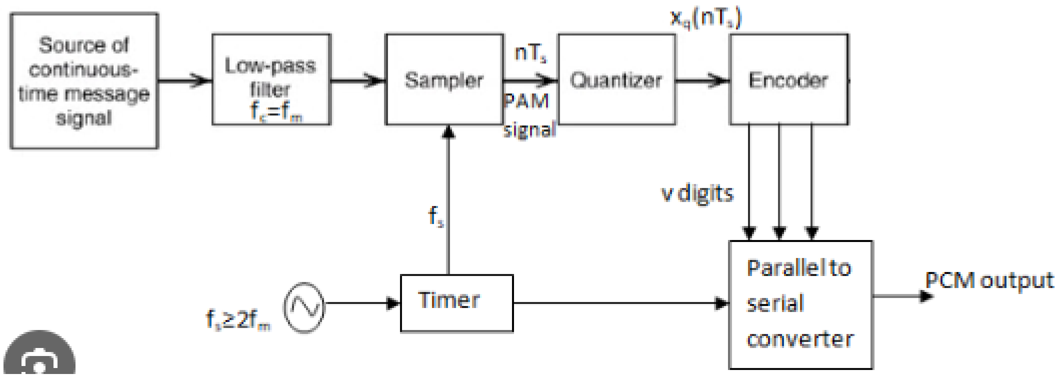
2. Time division multiplexing of digital information frequently leads to economical use of cables (or channels). Compared with frequency division multiplexing, no complex filters are required in the digital case since all the multiplexing function can be accomplished with digital circuitry.

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.

PCM Transmitter:



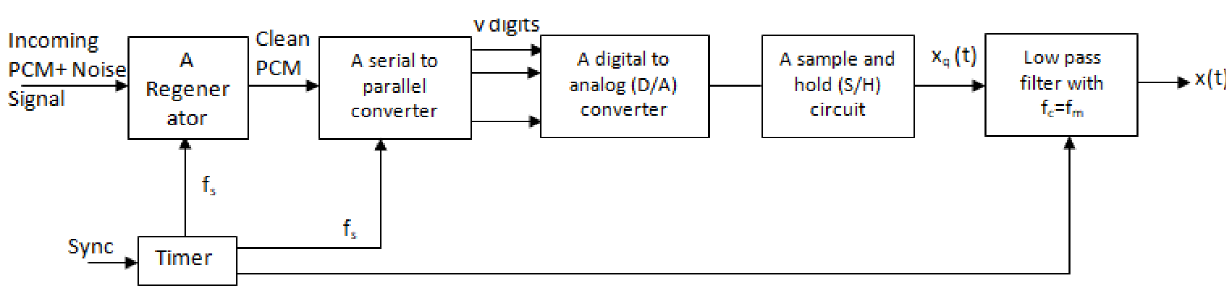
In PCM transmitter, the signal x(t) is first passed through the low-pass filter of cut-off frequency f_m Hz. This low-pass filter blocks all the frequency components above f_m Hz. This means that now the signal x(t) is bandlimited to f_m Hz .The sample and hold circuit then samples this signal at the rate of f_s.

Sampling frequency f_s is selected sufficiently above nyquist rate to avoid aliasing i.e.,

$f_s \geq 2f_m$ In fig.2, the output os sample and hold circuit is denoted by $x(nT_s)$. This signal $x(nT_s)$ is discrete in time and continuous in amplitude. A q-level quantizer compares input $x(nT_s)$ with its fixed digital levels.

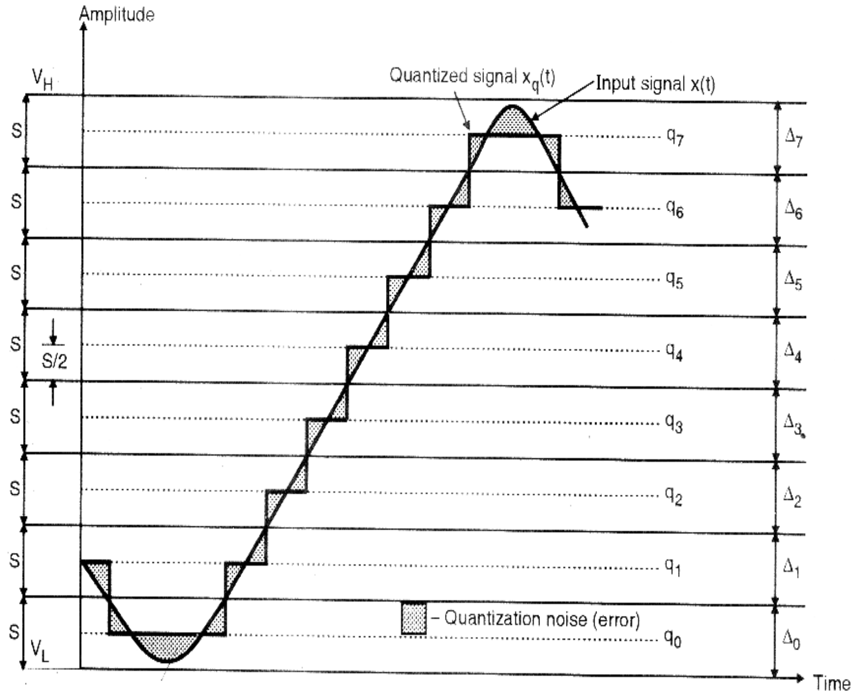
It then assigns any one of the digital level to $x(nT_s)$ which results in minimum distortion or error. This error is called quantization error. Thus output of quantizer is a digital level called $x_q(nT_s)$.Now the quantized signal level $x_q(nT_s)$ is given to binary encoder. This encoder converts input signal to 'v' digits binary word. This encoder is also known as digitizer. In addition to these, there is an oscillator which generates the clocks for sample and hold circuit and parallel to serial converter. In PCM, sample and hold, quantizer and encoder combinely form an analog to digital converter (ADC).

PCM Receiver:



The regenerator at the start of PCM receiver reshapes the pulse and removes the noise. This signal is then converted to parallel digital words for each sample. Now, the digital word is converted to its analog value denoted as $x_q(t)$ with the help of a sample and hold circuit. This signal, at the output of sample and hold circuit is allowed to pass through a low-pass reconstruction filter to get the original message signal x(t) .

Quantization:

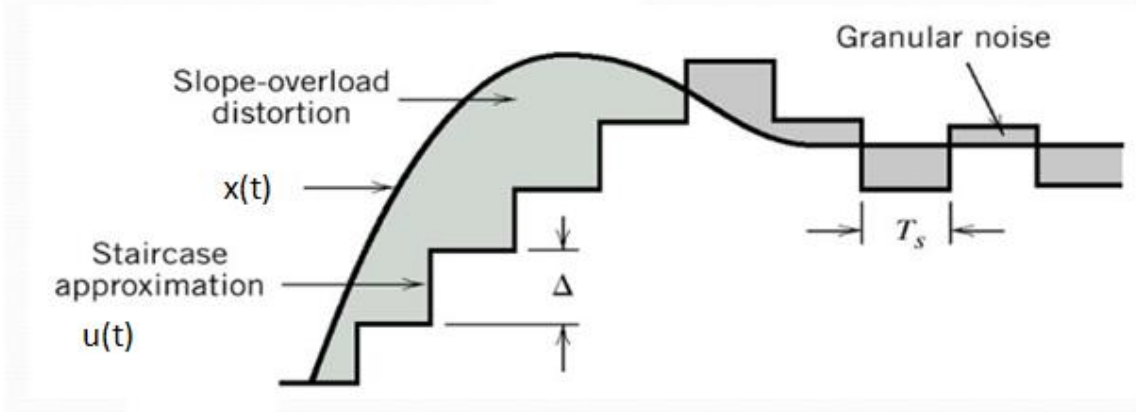


Process of quantization

Drawbacks of LDM

1. Slope Overload: Whenever the incoming signal slope exceeds the maximum slope of the staircase approximated signal, the LDM system cannot track the input signal. Then the output digital signal is a constant chain of '1' or '0'. This situation is known as 'slope overload' and is shown in Fig.3.2.

2. Hunting :- Also whenever the input signal X (t) is a DC signal or a signal, which changes by the amount less than the fixed step size 'S', the LDM system cannot track it but produces a pulse train of alternating '1' and '0' in the signal Q (t). The signal Y (t) oscillates about X (t) with a frequency equal to one-half of the input signal frequency. This situation is known as 'Hunting'.



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 $\pm\Delta$ 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		
2	Pulse generator o/p		
3	PCM O/P (Digital O/P)		

Conclusion:

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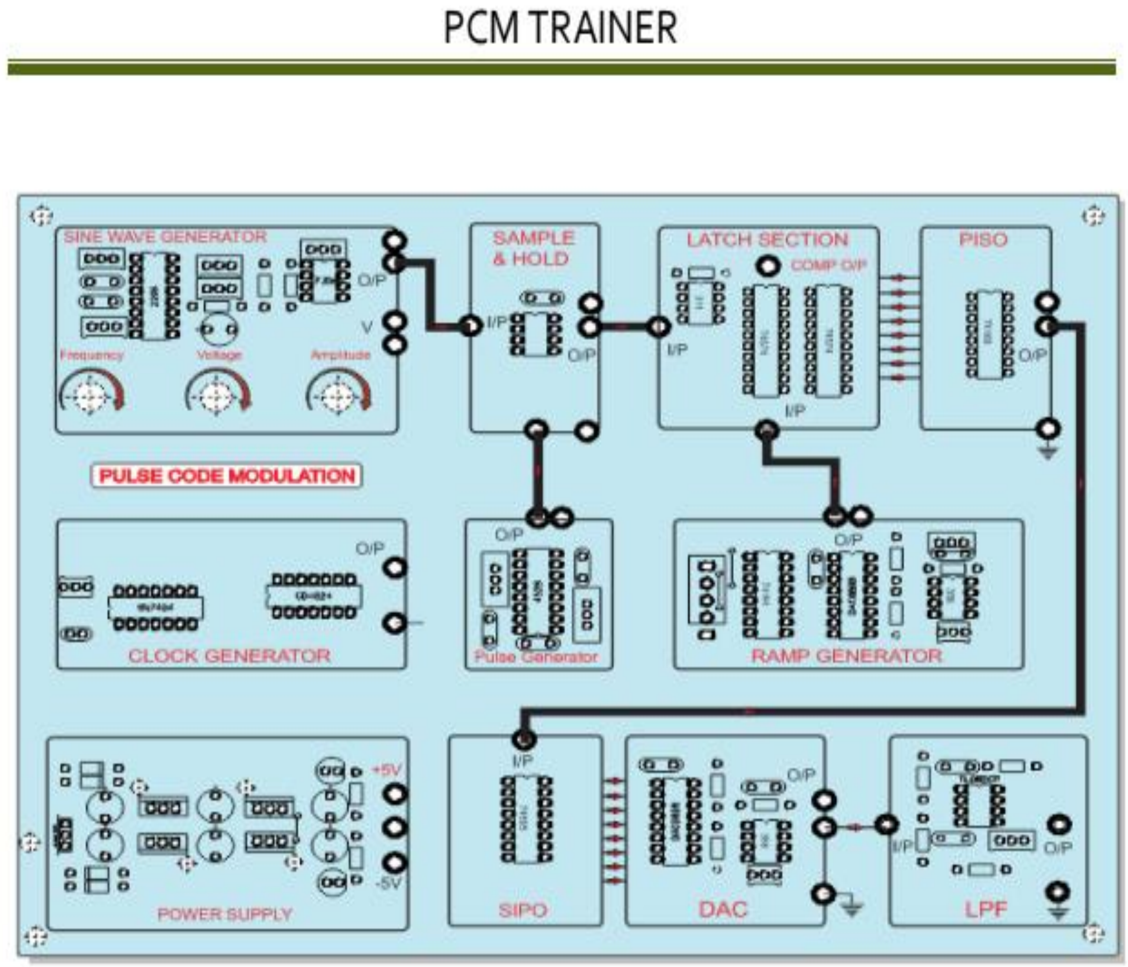
Questions :

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

PCM:


PCM TRAINER
SINE WAVE GENERATOR
O/P
V
Frequency
Voltage
Amplitude
SAMPLE & HOLD
I/P
O/P
LATCH SECTION
COMP O/P
I/P
PISO
O/P
PULSE CODE MODULATION
CLOCK GENERATOR
O/P
Pulse Generator
RAMP GENERATOR
POWER SUPPLY
+5V
-5V
SIPO
DAC
LPF

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 **comp**ression 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.

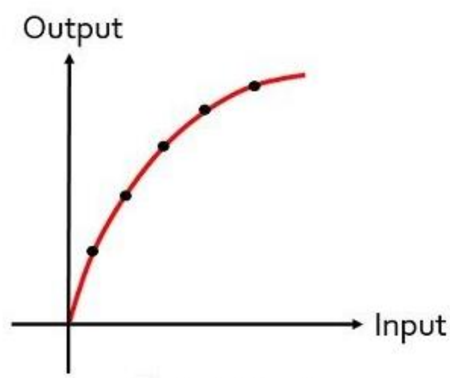
This is done in order to have an appropriate signal level as the input to the uniform quantizer. We know a high amplitude signal needs more bandwidth and also is more likely to distort. Similarly, some drawbacks are associated with low amplitude signal and thus there exist need for such a unit.

The operation performed by this block is known as compression thus the unit is called compressor.
The output of the compressor is provided to uniform quantizer where the quantization of the applied signal is performed.
At the receiver end, the output of the uniform quantizer is fed to the expander.
It performs the reverse of the process executed by the compressor. This unit when receives a low value signal then it attenuates it. While if a strong signal is achieved then the expander amplifies it.
This is done in order to achieve the originally transmitted signal at the output.

Characteristic of Compander

Companding is composed of compression and expanding. So, here in this session we will separately discuss the compressor and expander characteristic.

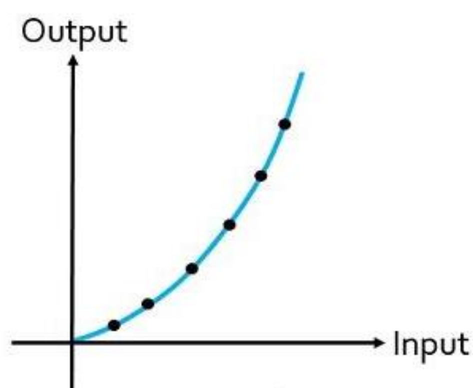
Compressor characteristic: The figure below shows the graphical representation of characteristic of the compressor:



Compressor Characteristic

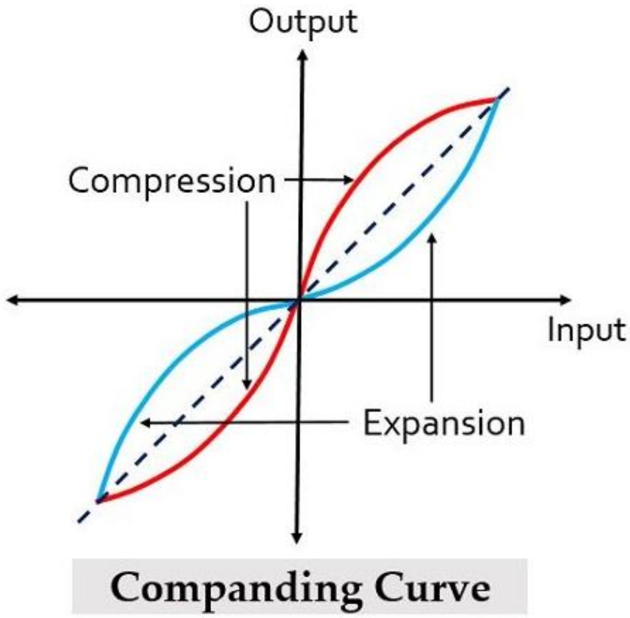
The graph clearly represents that the compressor provides high gain to weak signal and low gain to high input signal.

Expander characteristic: Here the figure shows the characteristic of expander:



Expander Characteristic

As we have already discussed that expander performs reverse operation of the compander. So, it is clear from the above figure that artificially boosted signals is attenuated to have the originally transmitted signal.
The figure below represents the companding curve for PCM system:



The compressor and expander performs inverse operations thus in the above figure the dotted line represents the linear characteristic of the compander indicating that the originally transmitted signal is recovered at the receiver.

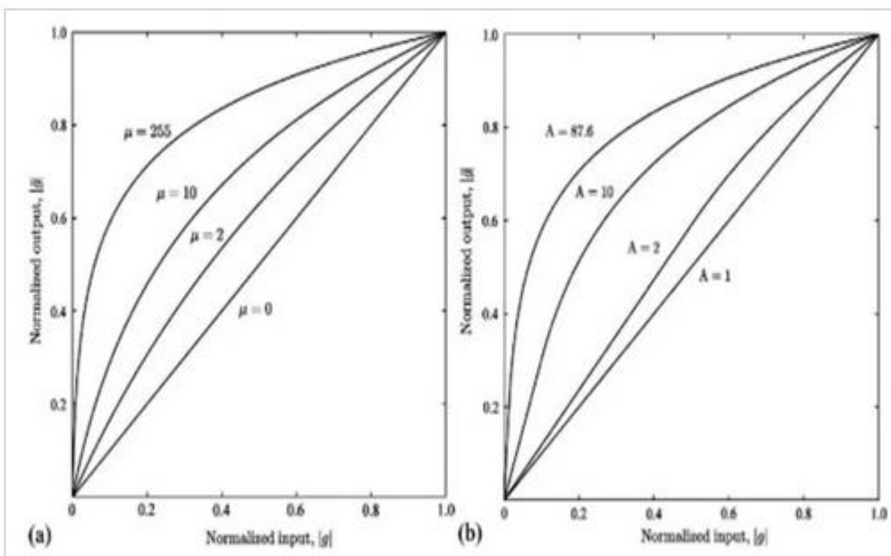


Figure 5: Compression curves for different values of compression parameters in (a) μ law (b) A law.

In companding, the rate of compression depends on the values of compression parameters μ and A of the equations presented in the section "Logarithmic Companding Curves". Greater the value of compression parameters, higher is the rate of compression

Higher compression rates imply greater non-linearity in quantization, which means there is a better representation of lower amplitude signals (using more number of bits) compared to higher amplitude ones. This indicates that, in a companded signal, the quantization error will be at its minimum at low 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

Conclusion:

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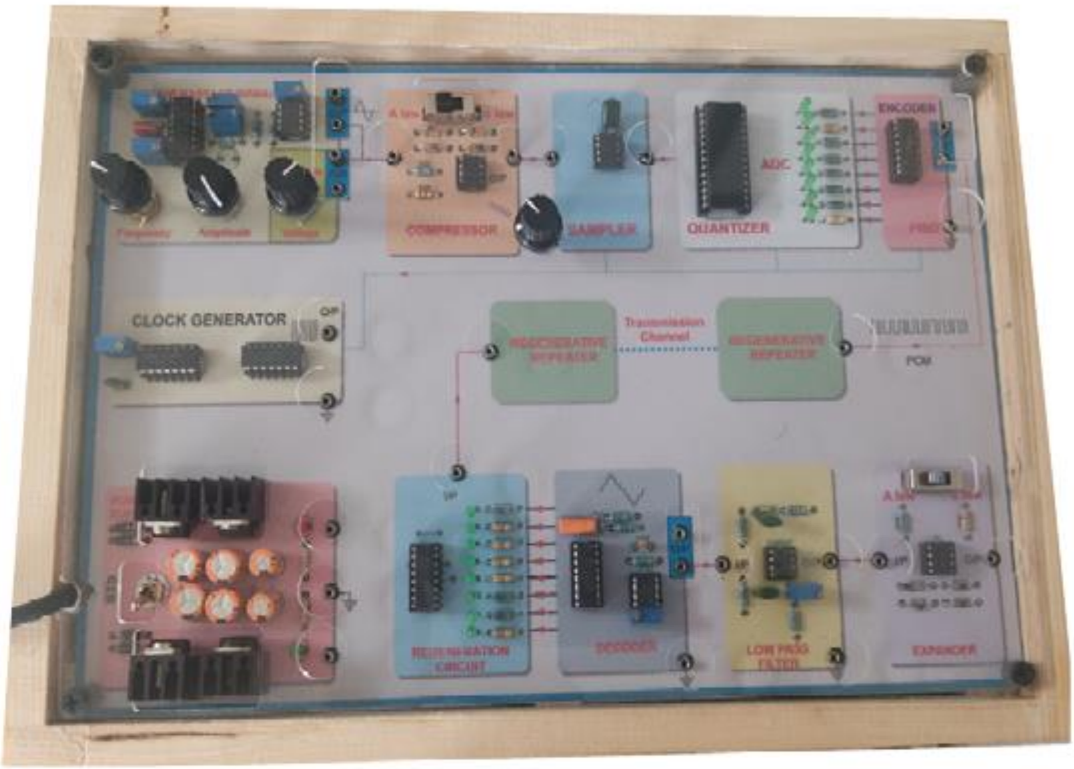
Questions:

Explain Companded PCM?

Explain A law and U law?

Explain Advanges of companded PCM?

COMPANDED PCM TRAINER



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.

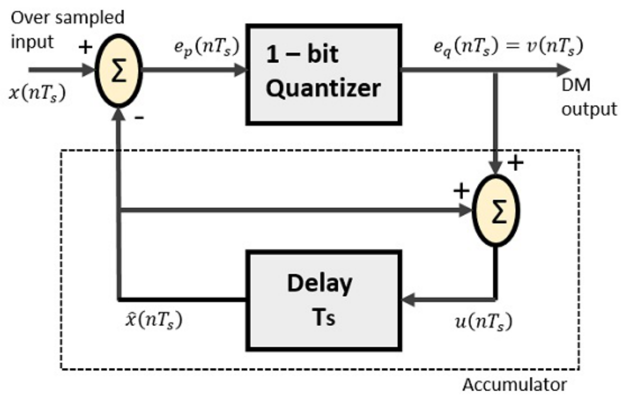
Delta Modulation is a simplified form of DPCM technique, also viewed as 1-bit DPCM scheme. As the sampling interval is reduced, the signal correlation will be higher.

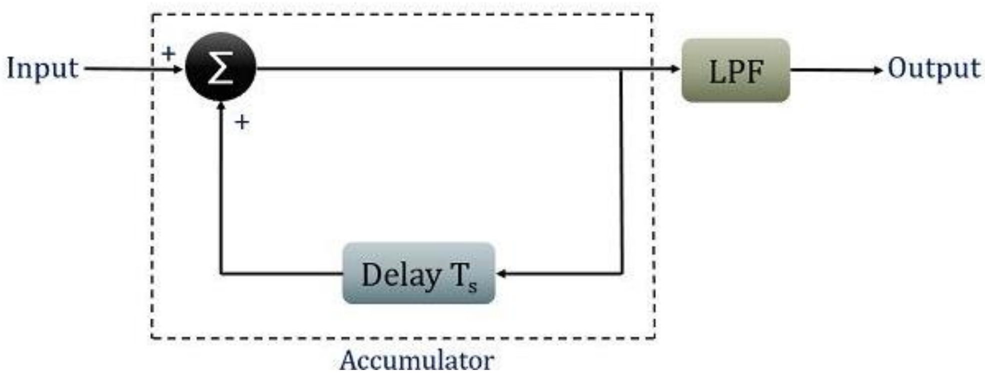
The Delta Modulator comprises of a 1-bit quantizer and a delay circuit along with two summer circuits. Following is the block diagram of a delta modulator.

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

- ep(nTs)ep(nTs) = summer output and quantizer input
- eq(nTs)eq(nTs) = quantizer output = v(nTs)v(nTs)
- xˆ(nTs)x^(nTs) = output of delay circuit
- u(nTs)u(nTs) = input of delay circuit





Advantages of DM Over DPCM

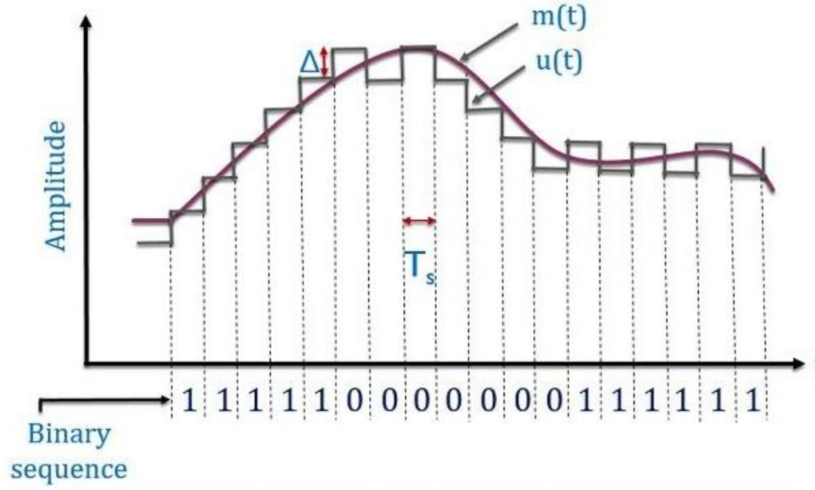
- 1-bit quantizer
- Very easy design of the modulator and the demodulator

However, there exists some noise in DM.

- Slope Over load distortion (when Δ is small)
- Granular noise (when Δ is large)

The gain of the voltage controlled amplifier is adjusted by the output signal from the sampler. The amplifier gain determines the step-size and both are proportional.

ADM quantizes the difference between the value of the current sample and the predicted value of the next sample. It uses a variable step height to predict the next values, for the faithful reproduction of the fast varying values.



Advantages of delta modulation

- Due to transmission of 1 bit per sample, it permits low channel bandwidth as well as signaling rate.
- ADC is not required. Thus permits easy generation and detection.

Disadvantages of delta modulation

- Delta modulation leads to drawbacks such as slope overload distortion and granular noise.

Applications of delta modulation

It is widely used in radio communication devices and digital voice storage and voice information transmission where signal quality is less important.

Procedure: 1. First of all turn on the trainer kit.
2. On the trainer board at the upper left corner there is sine wave generator section for the generating the analogue sine wave.
3. Connect the output of the sine wave generator section to the comparator section having to input.
4. Second input connect to the ground initially
5. Observe the output at the section output using Oscilloscope
6. Connect comparator section output to the input of the sample and hold block then observe the signal should be look like the square wave and inverted to the input here we can get the DM output.
7. Connect the DM output to the UP/DOWN Counter Section And observe the output on the output pins of the A-D convertor Section.
8. Connect the Second Pin of comparator section input probe to the output of the A-D convertor Block
9. Then Repeat the observation at A-D convertor.
10. Now you can see the Synthesize output using the Delta Modulation Technique.
11. Now give the other output Pin signal to the LPF Section so that we Can see the received signal to the across the output of the LPF section.
12. Draw the output observation at observation table.

Observation Table:

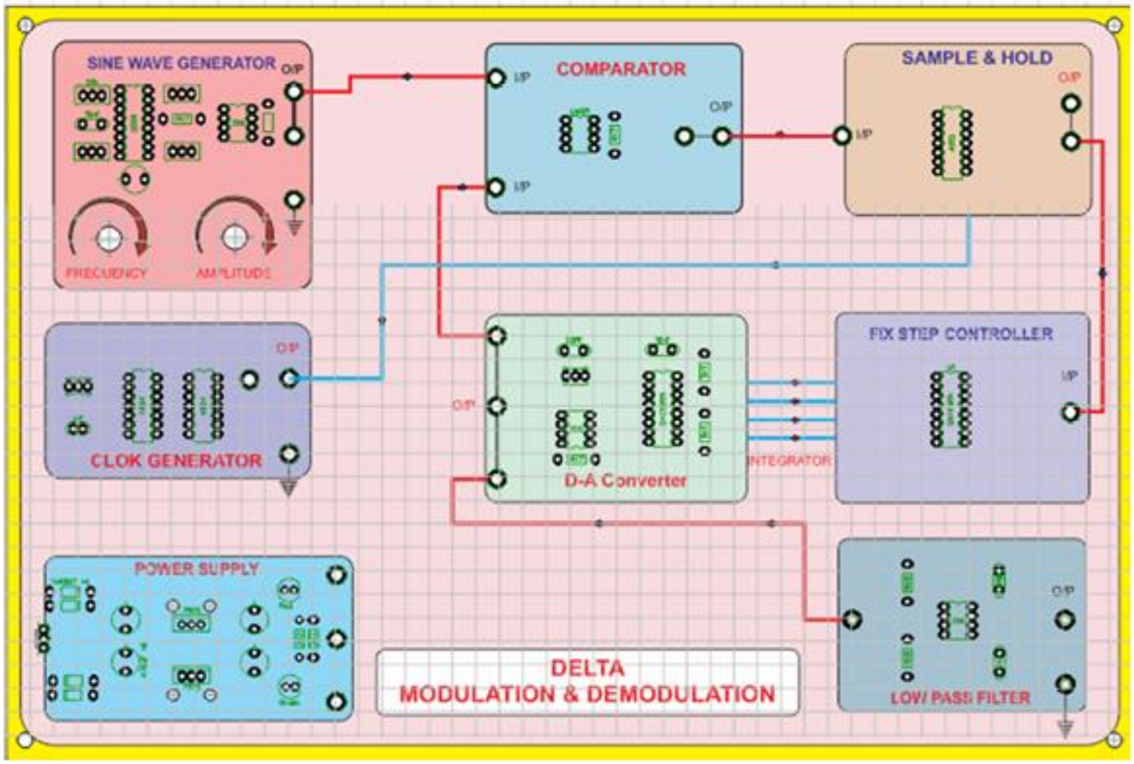
Sr.No	Testpoints	Freq	Voltage

Conclusion:

Questions:

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

DELTA MODULATION



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).

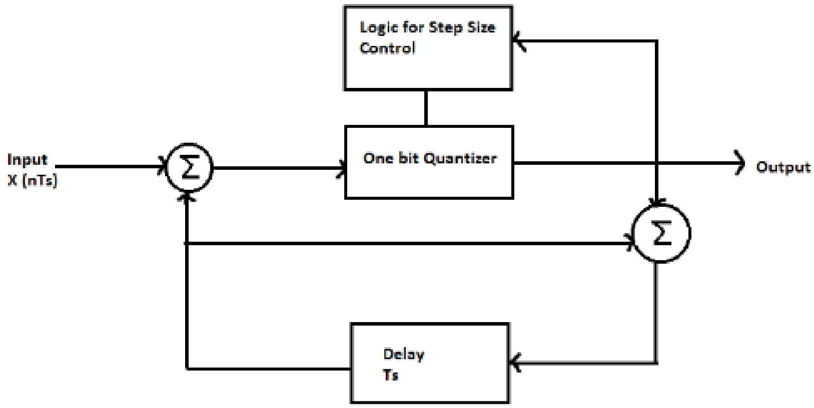
(2) Adaptive Delta Modulation (ADM): -

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.

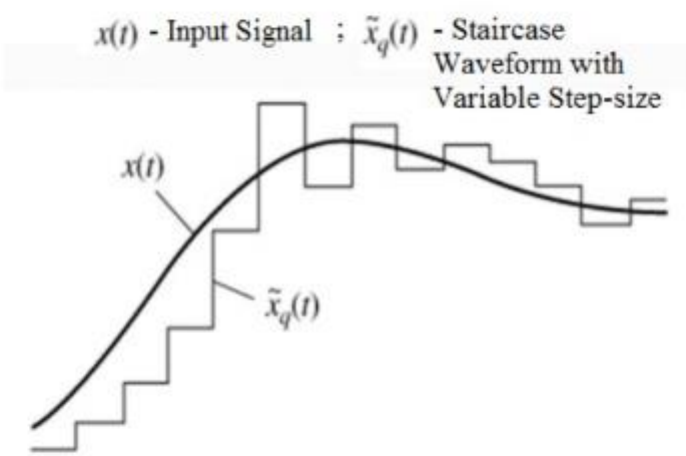
The quality of output is hampered in case of delta modulation because of fixed step size. In case of a steep slope of a modulating signal a larger step size is needed and vice versa. Due to this we are unable to get the exact value and hence it is required to make the step size adjustable in order to 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.



Explanation of concept

As we have already discussed, the step size of the staircase signal is not fixed and can be changed according to the input signal. In this case we calculate and estimate the present value and previous value difference . The error generated is then quantized depending on value, if sample value is smaller the quantized value will be high or else it will be low.

The logic step size unit is responsible for deciding the output based on quantizer output. Depending upon the quantizer output we can decide the step size. The step size for the next sample will be double when the value of the quantizer output is high. We can reduce the step size if the quantizer output is low.



Comparison between Delta Modulation and Adaptive Delta Modulation:

The comparison of Delta modulation and Adaptive delta modulation are

The main advantage of ADM is that the we cannot fix step size unlike in DM.

The ADM is an improvised version of DM which overcomes granular noise and slope overload error.

The DM dynamic range is not that wide as in case of ADM.

The DM does not utilize the bandwidth effectively as in case of ADM.

Procedure

A. Plug the power cable of the trainer KIT into power plug

B. Set the sine wave generator frequency to 500Hz for the start

C. Connect the output of the sine wave generator block to the comparator section input.

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

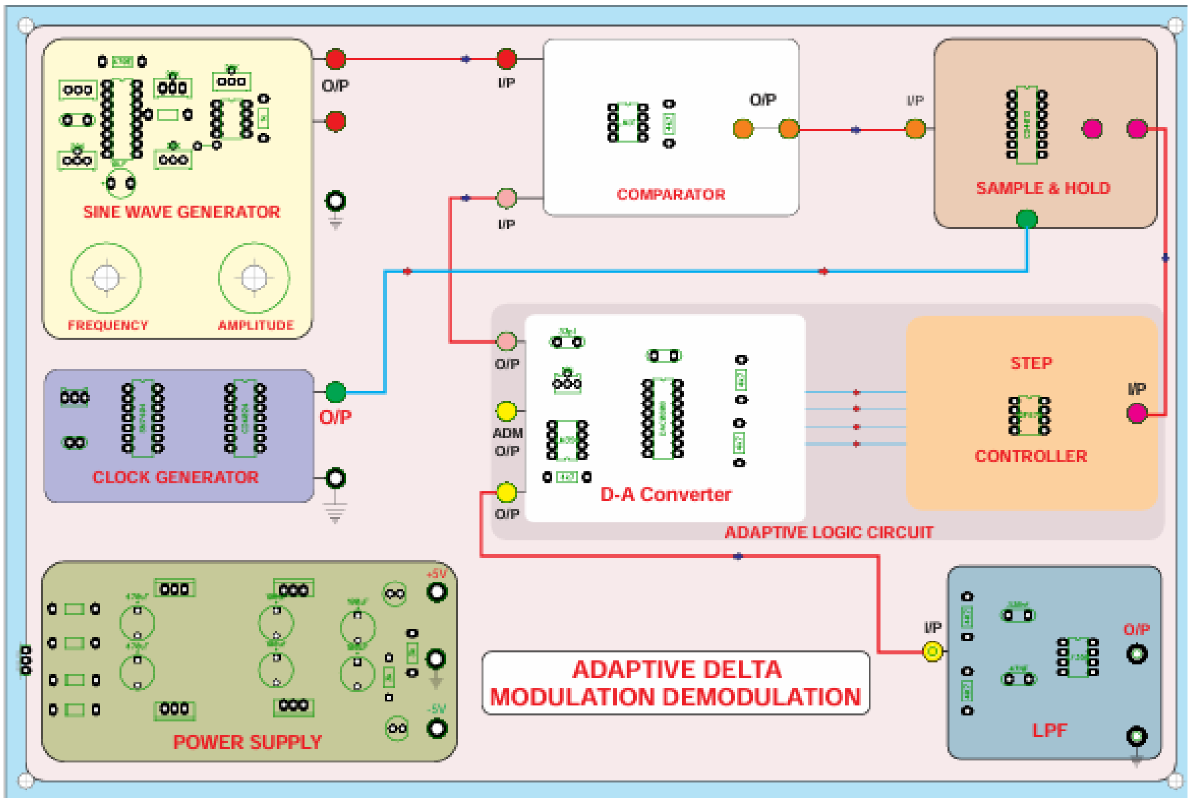
Observation Table:

Sr.No	Testpoints	Freq	Voltage

Conclusion:

Questions:

ADAPTIVE DELTA MODULATION



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.

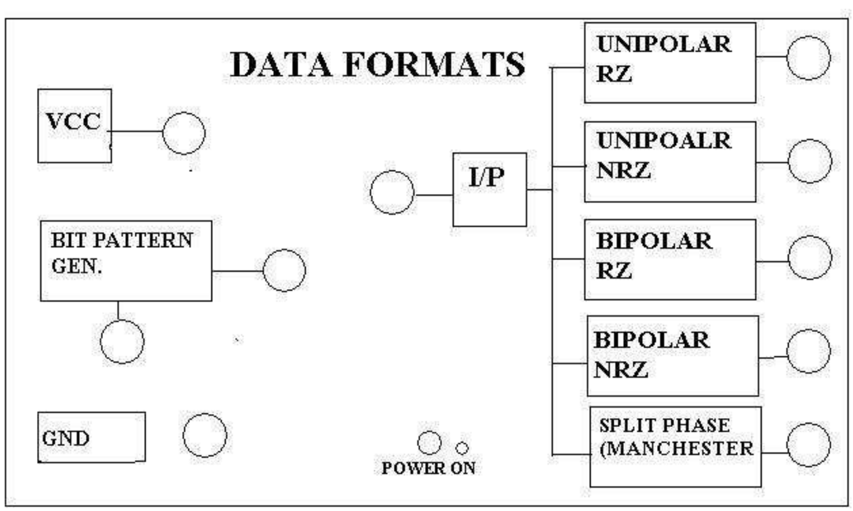
APPARATUS: Line coding kit, CRO, Spectrum Analyzer.

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.



Unipolar RZ Formats: 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

(RZ) format. As the voltage level is either +A or zero, this is a unipolar format. (Unipolar means only one polarity). Due to the unipolar nature, the unipolar RZ format has a nonzero DC value. The DC value does not contain any information.

(2) NON – RETURN – TO ZERO (MARK): NRZ (M)

The NRZ (M) code is very much similar to the NRZ (L) code. Here if logic 1 is to be transmitted. The new level is inverse of the previous level i.e. change in level occurs. If a data `0' is to be transmitted the level remains unchanged. Thus in the case of NRZ (M) waveform the present level is related to the previous levels. Thus, no longer the absolute value of signal is necessary instead it is the change in the level for which we look now.

(3) RETURN TO ZERO (RZ) FORMAT

The RZ code provides a partial solution to overcome the receiver clock regeneration problem with NRZ (L) code. It is similar to NRZ (L) code, except that the information is contained in the first half of the bit interval while the level during the second half of each period is always 0 volts

(4) BIPHASE (MANCHESTER) CODING

The encoding rules for Biphase (Manchester) code are as follows a data `0' is encoded as a low level during first half of the bit time and a high level during the second half. A data `1' is encoded as a high level during first half of the bit time and a low level during the second half.

Thus string of 1's or 0's as well as any mixture of them will not pass any synchronization problem at receiver.

(5) BIPHASE (MARK)

The Biphase (Mark) is yet another form of Biphase formats. In this coding also, the data is coded as two levels in each bit time. Here, the sequence of transmitted levels (low succeeded by high) or (high succeeded by low) depends on the order of sequence in previous bit time and the present data. The encoding laws followed by this format are

If a data `0' is to be transmitted, the sequence of the transmitted levels will remain same as for the previous bit interval. If a data `1' is to be transmitted, the sequence of the transmitted levels will reverse i.e. phase reversal will occur.

Thus,	LOGIC `0'	BIT PATTERN STAYS THE SAME
	LOGIC `1'	PHASE REVERSAL

(6)ALTERNATE – MARK INVERSION (AMI)

AMI being a three level code uses three levels namely, a positive voltage level, a negative voltage level and a bias level of 0 volts.

Like RB waveform, the AMI always returns to the bias level during second half of the bit time interval during the first half the transmitted level can be a positive level a negative level or a bias level, according to following coding rules.

A data `0' is always represented by the bias level

A data `1' may be represented by either a positive level or negative level, the level being chose opposite to what it was used to represent the previous data `1'.

Thus we have alternating positive level and negative level.

This justifies its name alternate Mark Inversion Mark is a telegrapher's word for logic `1'.

In NRZ (L) Code the pulse simply goes to the required level for one bit clock. In Rz, the pulse goes to the required level for first half and then returns to zero during the second half. In AMI, first logic `1' goes to a positive voltage level and then the second 1 goes a negative level alternating and so on

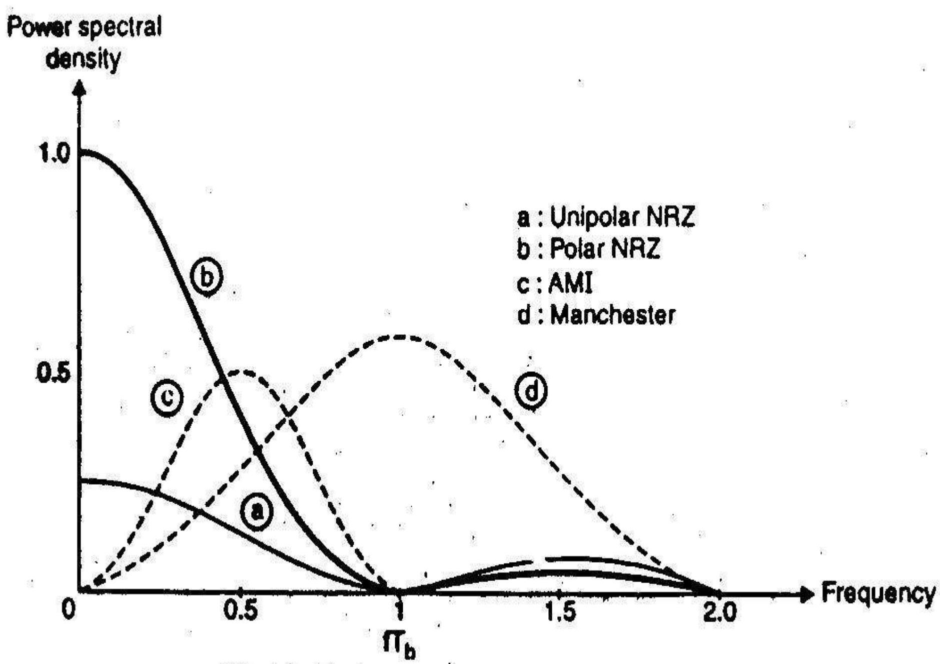


Fig. 7 : Power spectra of different line codes

Properties of Line Codes:

Each line code has advantages and disadvantages. For example, the unipolar NRZ line code has the advantage of using circuits that require only one power supply, but it has the disadvantage of requiring channels that are DC coupled (i.e. with frequency response down to f = 0), because the waveform has a non–zero DC value. The polar NRZ line code does not require a DC coupled channel, provided that the data toggles between binary 1's and 0's often and that equal numbers of 1's and 0's are sent. However, the circuitry that produces the polar NRZ signal requires a negative voltage power supply as well as the positive voltage power supply. The Manchester NRZ line code has the advantage of always having a 0 DC value, regardless of the data sequence, but it has twice the bandwidth of the unipolar NRZ or polar NRZ code because the pulses are half the width.

The following are some of the desirable properties of a line code:–

Self–Synchronization: There is enough timing information built into the code so that bit synchronizers can extract the timing or clock signal. A long series of binary 1's or 0's should not cause a problem in time recovery.

Low Probability of Bit Error: Receivers can be designed that will recover the binary data with a low probability of bit error when the input data is corrupted by noise or ISI

A Spectrum that is Suitable for the Channel: For example, if the channel is AC coupled, the PSD of the line code signal should be negligible at frequencies near 0. In addition, the signal bandwidth needs to be sufficiently small compared to the channel bandwidth, so that ISI will not be a problem

Transmission Bandwidth: This should be as small as possible – Error Detection Capability: It should be possible to implement this feature easily by the addition of channel encoders and decoders, or the feature should be incorporated into the line code.

Comparison of Line Codes:

Sr. No	Parameter	Polar RZ	Polar NRZ	AMI	Manchester	Polar Quaternary NRZ
1.	Transmission of DC component	Yes	Yes	No	No	Possible
2.	Signaling rate	$1/T_b$	$1/T_b$	$1/T_b$	$1/T_b$	$1/2T_b$
3.	Noise immunity	Low	Low	High	High	High
4.	Synchronizing capability	Poor	Poor	Very good	Very good	Poor
5.	Bandwidth required	$1/T_b$	$1/2T_b$	$1/2T_b$	$1/T_b$	$1/2T_b$
6.	Crosstalk	High	High	Low	Low	Low

Procedure:-

1) Switch on the power supply.
2) Connect one of the bit patterns as I/P to data format.
3) Observe the bit pattern.
4) Repeat the procedure for other bit patterns.

Observation Table:

Sr.No	Test points	Frequency(Hz)Bit Period	Voltage(V)
1			
2			
3			
4			
5			
6			
7			

GRAPHS:

Plot the observed line codes of a given sequence.

Conclusion:

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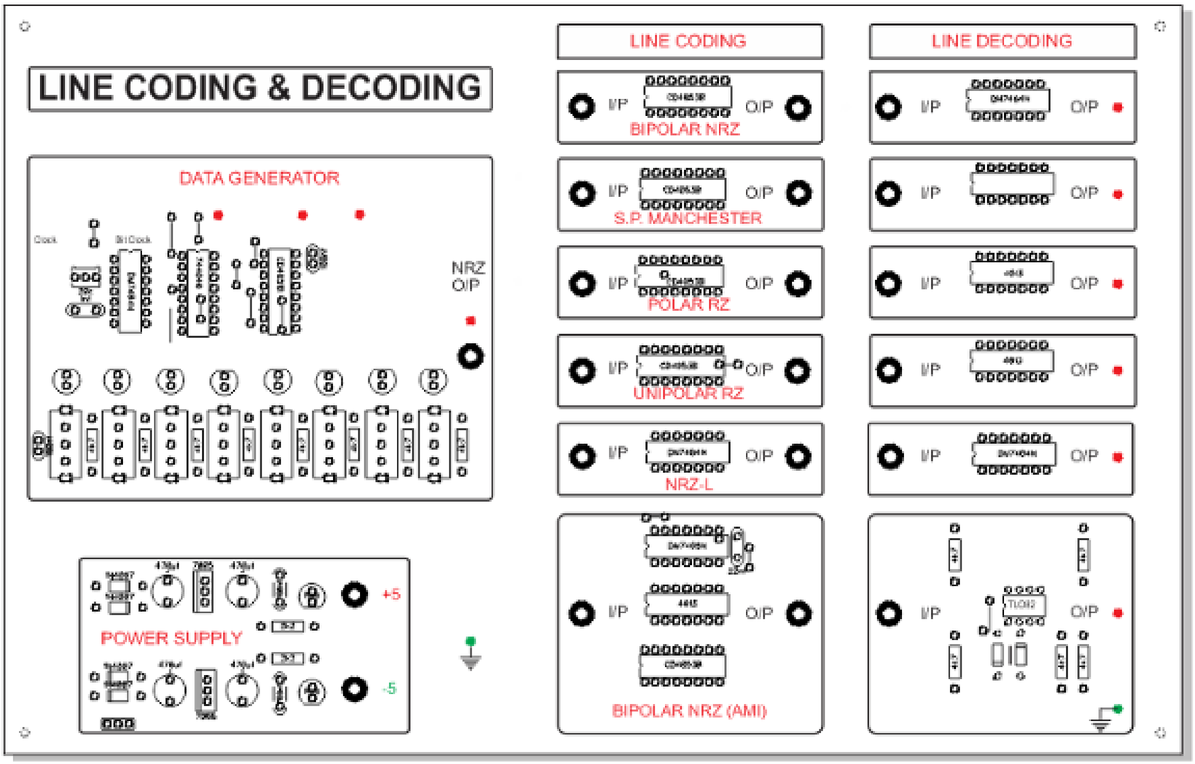
Questions:

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

LINE CODING & DECODING
MANUAL



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.

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

W is the highest frequency. This rate of sampling is called as Nyquist rate. A theorem called, Sampling Theorem, was stated on the theory of this Nyquist rate.

Sampling Theorem

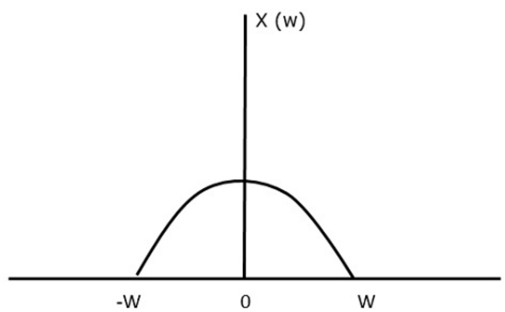
The sampling theorem, which is also called as Nyquist theorem, delivers the theory of sufficient sample rate in terms of bandwidth for the class of functions that are bandlimited.

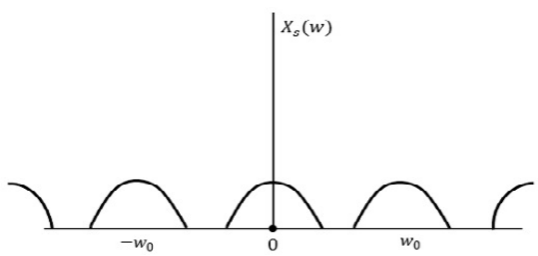
The sampling theorem states that, "a signal can be exactly reproduced if it is sampled at the rate fs which is greater than twice the maximum frequency W."

To understand this sampling theorem, let us consider a band-limited signal, i.e., a signal whose value is non-zero between some –W and W Hertz.

Such a signal is represented as x(f)=0 for |f|>W x(f)=0 for |f|>W

For the continuous-time signal x(t) the band-limited signal in frequency domain, can be represented as shown in the following figure.





We need a sampling frequency, a frequency at which there should be no loss of information, even after sampling. For this, we have the Nyquist rate that the sampling frequency should be two times the maximum frequency. It is the critical rate of sampling.

If the signal xtt is sampled above the Nyquist rate, the original signal can be recovered, and if it is sampled below the Nyquist rate, the signal cannot be recovered.

The following figure explains a signal, if sampled at a higher rate than 2w in the frequency domain.

The above figure shows the Fourier transform of a signal xs(t)xs(t). Here, the information is reproduced without any loss. There is no mixing up and hence recovery is possible.

The Fourier Transform of the signal xs(t)xs(t) is

$$X_s(w) = \frac{1}{T_s} \sum_{n=-\infty}^{\infty} X(w - nw_0)$$

Where T_s = **Sampling Period** and $w_0 = \frac{2\pi}{T_s}$

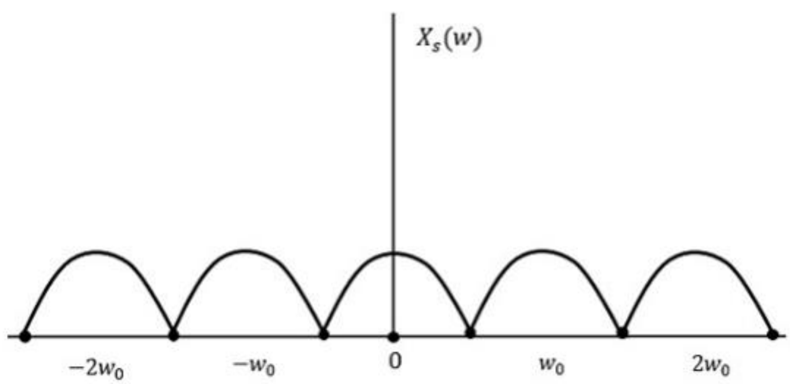
Let us see what happens if the sampling rate is equal to twice the highest frequency (**2W**)

That means,

$$f_s = 2W$$

Where,

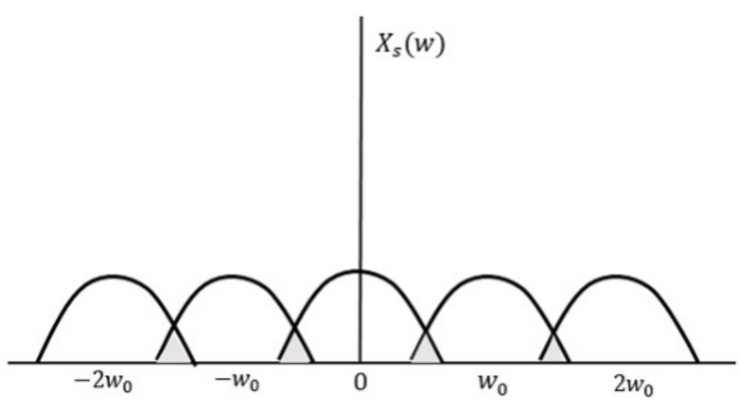
- f_s is the sampling frequency
- **W** is the highest frequency



The result will be as shown in the above figure. The information is replaced without any loss. Hence, this is also a good sampling rate. Now, let us look at the condition,

$$F_s < 2W$$

The resultant pattern will look like the following figure.



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:

Conclusion:

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Questions:

1. What does sampling mean? Name the various sampling techniques.
2. What do you know about impulse sampling? Mention its disadvantage?
3. Explain Anti aliasing filter
4. Explain Meaning of CLC, close all, Clear all instructions in Matlab?
5. Explain subplot stem and plot in Matlab?

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 $\Delta 0$, 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 $\Delta 1$, the quantizer output is constant equal to “q1”. Thus in each range from $\Delta 0$ to $\Delta 7$, 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?

- If we do not use the quantizer block in the PCM transmitter, then we will have to convert each and every sampled value into unique digital word. This will need a large number of bits per word (N). This will increase the bit rate and hence the bandwidth requirement of the channel.
- To avoid this, if we use a quantizer will only 256 quantization levels then all the sampled values will be finally approximated into only 256 distinct voltage levels. So we need 8 bits per word to represent each quantized sample value. Thus the number of bits per word can be reduced. This will eventually reduce the bit rate and bandwidth requirement.

Quantization Error OR Quantization Noise:

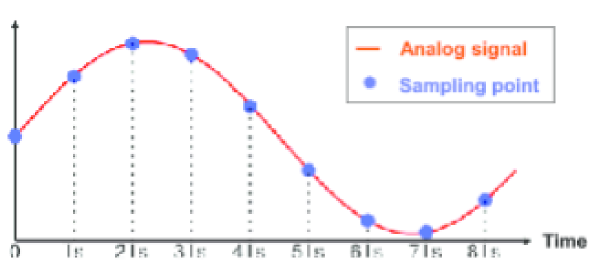
- The difference between the instantaneous values of the quantized signal and input is called as quantization error or quantization noise.

  = Xq (t) – X (t).
- Shaded portions of the waveform shown in figure show the quantization error.
- The maximum value of quantization error is + S/2 where S is step size. Therefore to reduce the quantization error we have to reduce the step size by increasing the number of quantization levels.
- The mean square value of the quantization is given by,
- Mean square value of quantization error = S2 / 12.
- The relation between the number of quantization levels Q and the number of bits per word (N) in the transmitted signal can be found as follows.

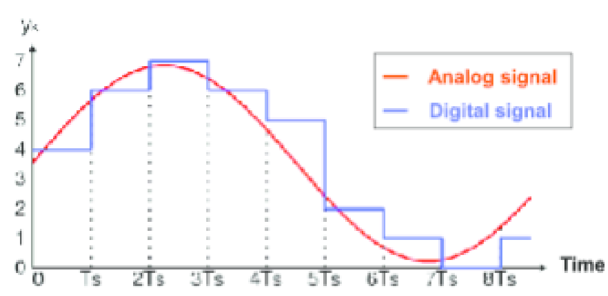
  Because each quantized level is to be converted into unique N bit digital word, assuming a binary coded output signal,
- The number of quantization levels Q = Number of combination of bits / word.

  That is Q = 2N.

  Thus if N=4 i.e 4 bits per word then the number of quantization levels will be 16.



(a)



(b)

Signal to Quantization noise ratio (SNRq):

- The ratio is the figure of merit for the PCM systems. The signal to quantization noise ratio with a sinusoidal input signal to the PCM system is expressed as,

- Si / Nq =[1.8 + 6N] db for sinusoidal signal.
- This equation shows that the signal to quantization noise ration is solely dependent on the number of bits per word i.e N. This ratio should be as high as possible, which can be achieved by increasing N. But this increase the bit rate and hence bandwidth of the PCM system. Therefore the number of bits per word is a compromise between high SNRq and bandwidth requirements.

Signaling Rate and Transmission Bandwidth of PCM:

- We know that, Q=2N where, Q= Number of quantization levels.
  N= Number of bits per word.
- The input signal X (t) is sampled at the sampling rate fs, i.e. there is fs number of samples per each second. Each of these samples is then converted into an N bit digital word.
- Therefore Number of bits / sec. = Number of samples/sec x Number of bits/sample.
  = fs x N.
  But signaling rate is nothing but the number of bits per second.
  Therefore signaling rate of PCM = Nfs.
  The transmission bandwidth of PCM is equal to half the signaling rate.
  Therefore Transmission bandwidth of PCM = ½ Nfs.

Application of PCM:

- In telephony with (with advent of fiber optic cables).
- In space communication where a spacecraft transmits signal to earth. Here the transmitted power is very low (10 to 15W) and the distances are huge, (a few million km.). Still due to high noise immunity, only PCM systems can be used in such applications.

Advantages of PCM:

- Very high noise immunity.
- Due to digital nature of signal, repeaters can be placed between the transmitter and the receivers. The repeaters actually regenerate the received PCM signal. This is not possible in analog systems. Repeaters further reduce the effect of noise.
- It is possible to store the PCM signal due to its digital nature.
- It is possible to use various coding techniques so that only desired person can decode the received signal.

Disadvantage of PCM:

- The encoding, decoding and quantizing circuitry of PCM is very complex.
- PCM requires a large bandwidth as compared to the systems.

Delta Modulation:

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.**

The type of modulation, where the sampling rate is much higher and in which the stepsize after quantization is of a smaller value **Δ**, such a modulation is termed as **delta modulation**.

Features of Delta Modulation

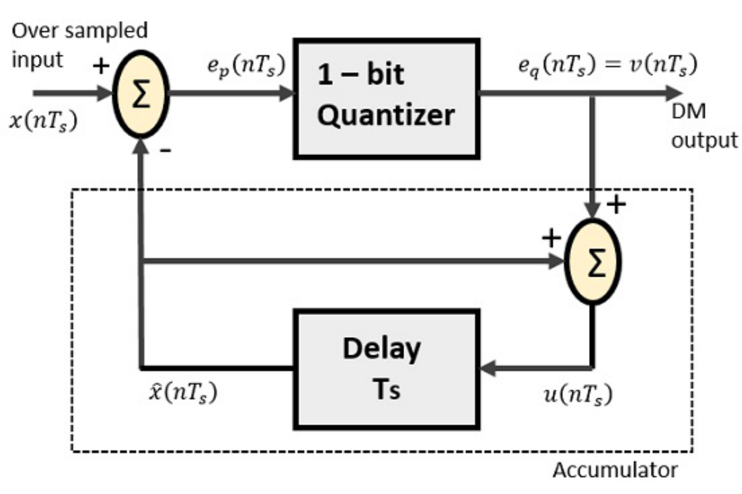
Following are some of the features of delta modulation.

- 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**
- 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.

Delta Modulator

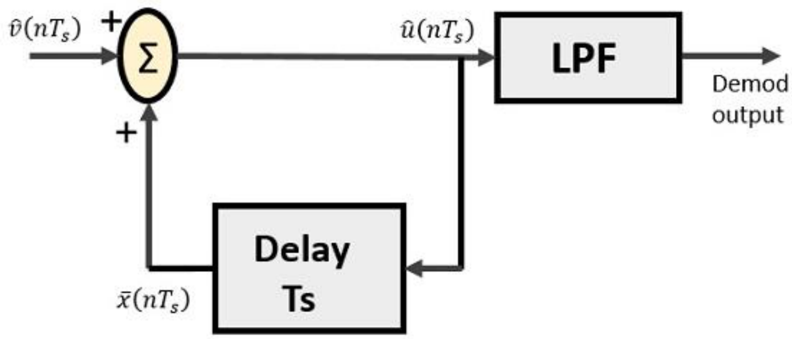
The Delta Modulator comprises of a 1-bit quantizer and a delay circuit along with two summer circuits. Following is the block diagram of a delta modulator.



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:**

Conclusion:

Questions:

1. Draw and explain block diagram of PCM?
2. Explain Quantization Process?
3. Explain Quantization Noise?
4. Write definition of sampling quantization and encoding?
5. Write advantages of Digital signals over analog signals?
6. List Application of PCM?
7. State advantages and disadvantages of Delta Modulation?
8. Explain drawbacks of Delta Moduation?
9. Differentiate in PCM, DM and ADM?
10. Give application of Delta Modulation ?

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.

➡It does not have any DC components as it creates balance between positive voltage levels and negative levels during encoding process in line coding techniques such as R8ZS and HDB3.

➡It offers error detection capability.

Drawbacks or disadvantages of Scrambling

Following are the drawbacks or disadvantages of Scrambling:

➡Multiplicative descrambler produces error multiplication. A single bit error at the input of such descrambler produces error in descrambled data. Hence it is used with other FEC techniques.

➡Additive scrambler should be reset by "frame sync" initially during descrambling otherwise massive error propagation occurs.

➡Both scrambler types fail to produce random sequences in worst case conditions.

Program Code: Scrambler_input=[80 255 16 9 48 255 80 0 25 0 145]

```
s=20255; %Initialization of scrambler circuit

rand_data=zeros(size(Scrambler_input));

for j=1:size(Scrambler_input,2);

for i=1:8

msb=bitxor(bitget(s,1),bitget(s,2));

s=bitshift(s,-1);

s=bitset(s,15,msb);

t=bitxor(bitget(Scrambler_input(j),9-i),msb);

rand_data(j)=bitset(rand_data(j),9-i,t);

end

end

scrambler_out=rand_data
```

Descrambler MATLAB Code

s=20255; %Initialization of de-scrambler circuit descrambler_in=zeros(size(scrambler_out)); for j=1:size(scrambler_out,2); for i=1:8 msb=bitxor(bitget(s,1),bitget(s,2)); s=bitshift(s,-1);

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:**

Conclusion: