Government Girls' Polytechnic, Bilaspur

Name of the Lab: **Communication Lab**

Practical: **Principles of Communication Lab**

Class :**IV sem (ET&T), V sem (IT)**
FOR ET&T
Teachers Assessment: 10 End Semester Examination:50
FOR IT
Teachers Assessment: 30 End Semester Examination 70

EXPRIMENT NO.1

Objective: Perform amplitude modulation of a signal, plot the waveform and calculate modulation index

Apparatus Required: CRO, CRO probes, Function Generator (2 nos.), Power supply, Breadboard, connecting wires. Transistor: - BC548/549 (1 nos.),Resistor: - 330W (2 nos.), Inductor: - 1mH (1 nos.)

Theory: The message signal x(t) is modulated by a carrier waveform Ac cos(2πfct). The amplitude modulated waveform xc(t) is.

$$\begin{aligned} x_c(t) &= A_c x(t)\cos(2\pi f_c t) + A_c\cos(2\pi f_c t) \\ &= A_c[1 + x(t)]\cos(2\pi f_c t) \quad (2) \\ &= A(t)\cos(2\pi f_c t) \end{aligned}$$

where A(t)j is the envelope of the AM signal. The message signal amplitude must be small and must have aDC component equal to zero in order to use simple demodulation schemes. The spectrum of the modulatedsignal is

$$X_c(f) = A_c/2[X(f + f_c) + X(f_{¡} - f_c)] + A_c/2[(f_{¡} \pm f_c) + (f - f_c)]$$

Thus, the carrier waveform is transmitted along with the sidebands to make it available at the receiver forcoherent demodulation without needing complex carrier recovery circuits. However, this advantage of AMover DSB demodulation comes at the expense of using some portion of transmitted power in the carrierwaveform which does not convey any information

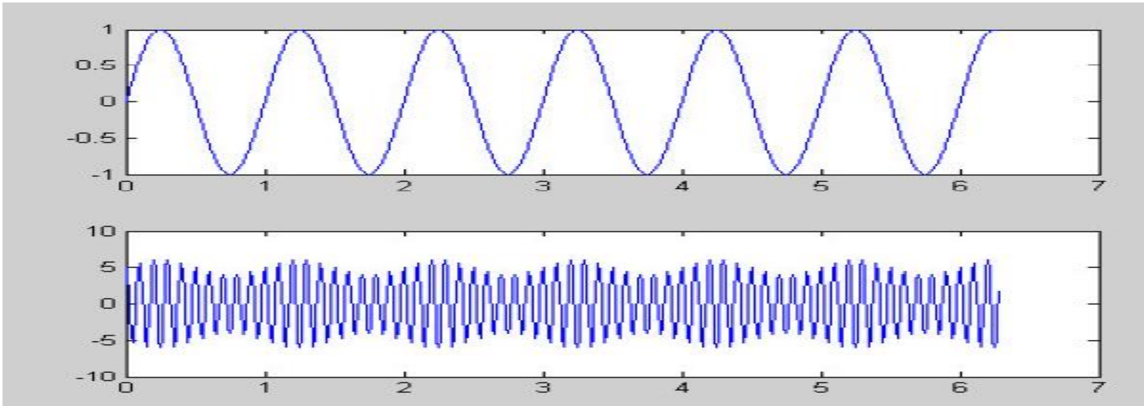
Modulation index of AM signals

The two main differences of AM with respect to DSB signals are the presence of a carrier component and that the envelope A(t) of the modulated carrier has the shape of x(t) under the conditions that fc >> fx and A(t) > 0 for all values of t. Thus, having jx(t)j < 1 with no DC component guarantees that A(t) is always positive. The modulation index m of an AM signal is defined as

$$m = (E_{max} - E_{min}) / (E_{max} + E_{min})$$

When m > 1 the envelope has no longer the shape of x(t) resulting in envelope distortion. This condition is referred to as overmodulation. In the particular case when the message signal is a sinusoid of frequency fx, its amplitude is equal to the modulation index m of the corresponding AM signal

$$x_c(t) = A_c[1 + m\sin(2\pi f_x t)]\cos(2\pi f_c t)$$



Result: Study of Amplitude modulation is completed.

EXPRIMENT NO.2

Objective: Perform frequency modulation of a signal and trace the frequency modulated waveform from CRO.

Apparatus Required: FM Modulator kit, Connecting wires,CRO.

Theory:

In frequency modulation, the information-bearing signal or the message signal we wish to transmit is used to modulate the frequency of another signal that is the carrier signal rather than amplitude variations in the carrier signal as in case of amplitude modulation.Frequency Modulation is part of a more general class of modulation schemes known as angle modulation. Angle modulation includes both phase modulation and frequency modulation. Theories and concepts are similar for phase modulation and frequency modulation, but we will only refer to frequency modulation in this lab.

Single frequency FM modulation

In frequency modulation, the amplitude of the modulated carrier signal is kept constant while its frequency is varied by the modulating message signal. The basic idea of frequency modulation is shown in figure (1). The carrier frequency is controlled at each instant by the voltage of the modulating signal. The frequency of the modulated signal is increased if the input signal is positive, whereas the frequency is reduced if the input signal is negative.

The general definition of frequency modulated signal $S_{FM}(t)$ is given by the formula:

$$S_{FM}(t) = A_C \cos(2\pi f_C t + \theta(t)) = A_C \cos(2\pi f_C t + 2\pi K_f \int_{-\infty}^{t} m(\tau)d\tau)$$

where,

$m(\tau)$	is the modulating signal.
A_C	is the amplitude of the carrier.
f_C	is the carrier frequency.
K_f	is the frequency deviation constant measured in Hz/V.

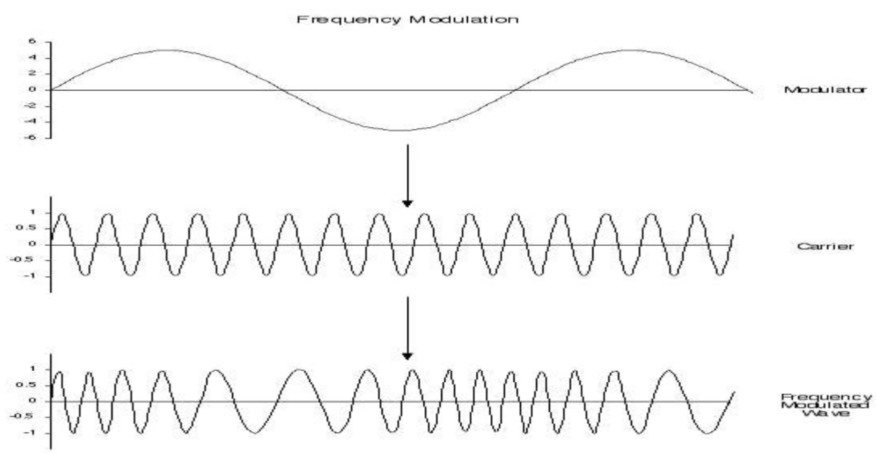


Fig. 1: frequency modulation process

For the case of a single frequency sinusoidal modulating signal, $m(t) = A_m \cos(2\pi f_m t)$, the frequency modulated signal $S_{FM}(t)$ will be expressed as:

$$S_{FM}(t) = A_C \cos(2\pi f_C t + \frac{K_f A_m}{f_m} \sin(2\pi f_m t)) \qquad \ldots\ldots\ldots\ldots\ldots\ldots \quad (2)$$

The factor $k_f A_m$ is called the frequency deviation. It is defined as the maximum frequency shift away from f_c.

The frequency modulation index β_f, is expressed as:

$$\beta_f = \frac{k_f A_m}{f_m} = \frac{\Delta f}{f_m} \qquad (3)$$

So, the $S_{FM}(t)$ signal can be represented as:

$$S_{FM}(t) = A_C \cos(2\pi f_C t + \beta_f \sin(2\pi f_m t)) \qquad (4)$$

Frequency modulated signals are classified into two categories based on the value of β_f.

Narrow Band Frequency Modulation (NBFM)

For small values of the frequency modulation index (β_f <<1), we have Narrow Band Frequency Modulation (NBFM). In this case, the frequency modulated signal $S_{FM}(t)$ becomes:

$$S_{FM}(t) = A_C \cos(2\pi f_C t) - A_C \beta_f \sin(2\pi f_c t) \sin(2\pi f_m t) \qquad (5)$$

The derivation of equation (5) can be found in reference [2].

Wide Band Frequency Modulation (WBFM)

As the modulation index increases, the signal occupies more bandwidth. In this case the modulation scheme is called Wide Band Frequency Modulation (WBFM).

Therefore, for a single frequency sinusoidal modulating signal the frequency modulated signal could be written in the form:

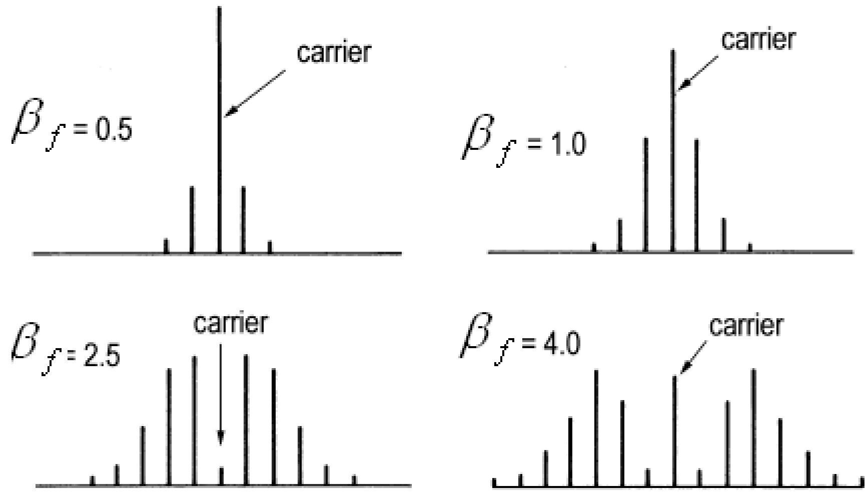
$$S_{FM}(t) = A_C \sum_{n=-\infty}^{\infty} J_n(\beta) \cos(2\pi (f_C + n f_m) t) \qquad (6)$$

The derivation of equation (6) can be found in reference [2].

FM spectrum

As with amplitude modulation, the modulation process causes sidebands to be produced at frequencies above and below the carrier. However, for a frequency modulation based system, there are a lot more, all spaced at multiples of f_m from the carrier frequency f_C. As a result, the bandwidth needed to accommodate a frequency modulated signal is considerably larger than that for amplitude modulated signal having the same modulating frequency.

Figure (2) shows the spectrum of a frequency modulated signal for various values of the modulation index β_f. The modulating signal in these examples is a single frequency sinusoidal signal.



When a sinusoidal signal such as $m(t) = A_m \cos(2\pi f_m t)$ is used, the spectrum contains a carrier component and many number of sidebands located on either side of the carrier frequency, spread at integer multiples of the modulating frequency f_m ($f_c \pm nf_m$), for all positive n (n=0 is the carrier frequency component).

The only exception is at a very low frequency modulation index, most of the information is contained within the range of the first upper and lower sidebands, which makes the total bandwidth sufficient for transmission about the same as for amplitude modulation based system, that is $2f_m$. With larger frequency modulation indexes, the number of sidebands increases and we obtain larger bandwidth.

Fortunately, something else is happening which keeps the total bandwidth reasonable. To get a large frequency modulation index, we need a large frequency deviation but a small modulation frequency, according to the modulation index definition. The modulation frequency; however, determines the spacing between sidebands. So, at high modulation index, we may have many sidebands, but they will be closely spaced, so the total occupied bandwidth will not be much larger. This is demonstrated in figure (3).

Theoretically, the bandwidth of a frequency modulated carrier is infinite.In practice, however, we find that the frequency modulated signal is effectively limited to a finite number of significant sideband frequencies within an approximate bandwidth, B_T, given by Carson's rule.

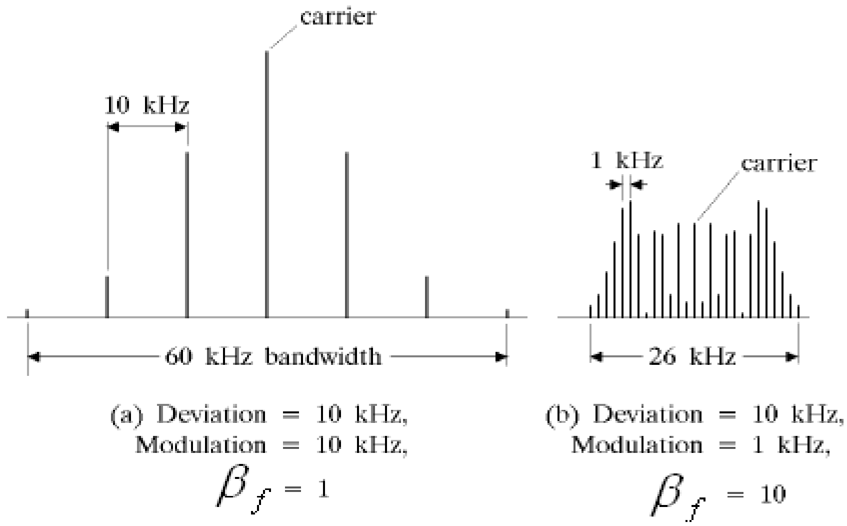


Fig. 3: Same deviation, but different modulation index

Carson bandwidth rule is a rule defining the approximate bandwidth requirements of communications system components for a carrier signal that is frequency modulated.

In case of a single frequency modulation, the empirical Carson's rule is given by

$$B_T \cong 2f_m(\beta_f + 1) \qquad (7)$$

For more practical case, an arbitrary modulating signal m(t) is considered and its highest frequency component is denoted by W. Then, replacing β by D and replacing f_m with W in equation (7) we get

$$B_T \cong 2W(D+1) \qquad (8)$$

D is called the deviation ratio and it is defines as the ratio of the frequency deviation Δf , which corresponds to the maximum possible amplitude of the modulating signal m(t) ,to the highest modulation frequency W.

The maximum frequency deviation depends on the maximum amplitude of the modulating signal and the sensitivity of the modulator. The sensitivity of the modulator is called the frequency-deviation constant, K_f . Thus, D is given by the following formula:

$$D = (K_f * \max|m(t)|)/W \qquad (9)$$

where, W is the highest modulation frequency, m(t) is the message signal and K_f is the frequency-deviation constant.

The deviation ration D plays the same role for arbitrary modulation that the modulation index β_f plays for the case of a single sinusoidal modulation. From a practical viewpoint, Carson's rule somewhat underestimated the bandwidth.

Result: Study of Frequency modulation is completed.

EXPERIMENT NO 3

Objective: Perform phase modulation of a signal and trace the phase modulated waveforms from CRO

Theory:

Phase modulation is very similar to the frequency modulation. The only difference is that the phase of the carrier is varied instead of varying the frequency. The amplitude of the carrier remains constant. As the modulating signal goes positive, the amount of phase lag increases with the amplitude of the modulating signal. The effect is that the carrier signal is stretched out or its frequency is lowered. When the modulating signal goes negative, the phase shift becomes leading. This causes the carrier wave to be effectively speeded up or compressed. The effect is as if the carrier frequency has been increased. Thus phase modulation produces frequency modulation.

Phase Modulation (PM):-

The phase modulation is another type of angular modulation. PM and FM are closely related. It is possible to obtain FM from PM, using the method called "Armstrong Method"

The PM wave is obtained by varying the phase angle Φ of a carrier in proportion with the amplitude of the modulating voltage.

If the carrier voltage is expressed as,

$$e_c = A \sin(\omega_c t + \Phi)$$

Then the PM wave can be expressed as,

$$e_{PM} = A \sin[\omega_c t + \Phi_m \sin \omega_m t]$$

Here $\Phi_m =$ maximum phase change corresponding to the maximum amplitude of the modulating signal. For the sake of uniformity let us modify the equation as,

$$e_{PM} = A \sin[\omega_c t + m_p \sin \omega_m t]$$

Where $m_p = \Phi_m =$ Modulation index of PM.

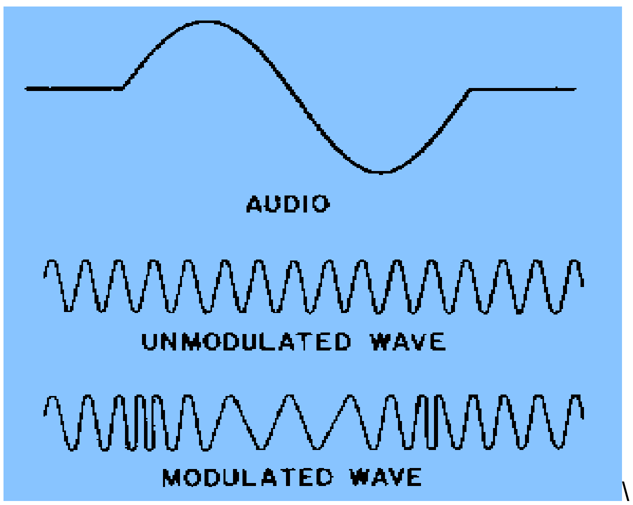


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Fig: waveform of phase modulation

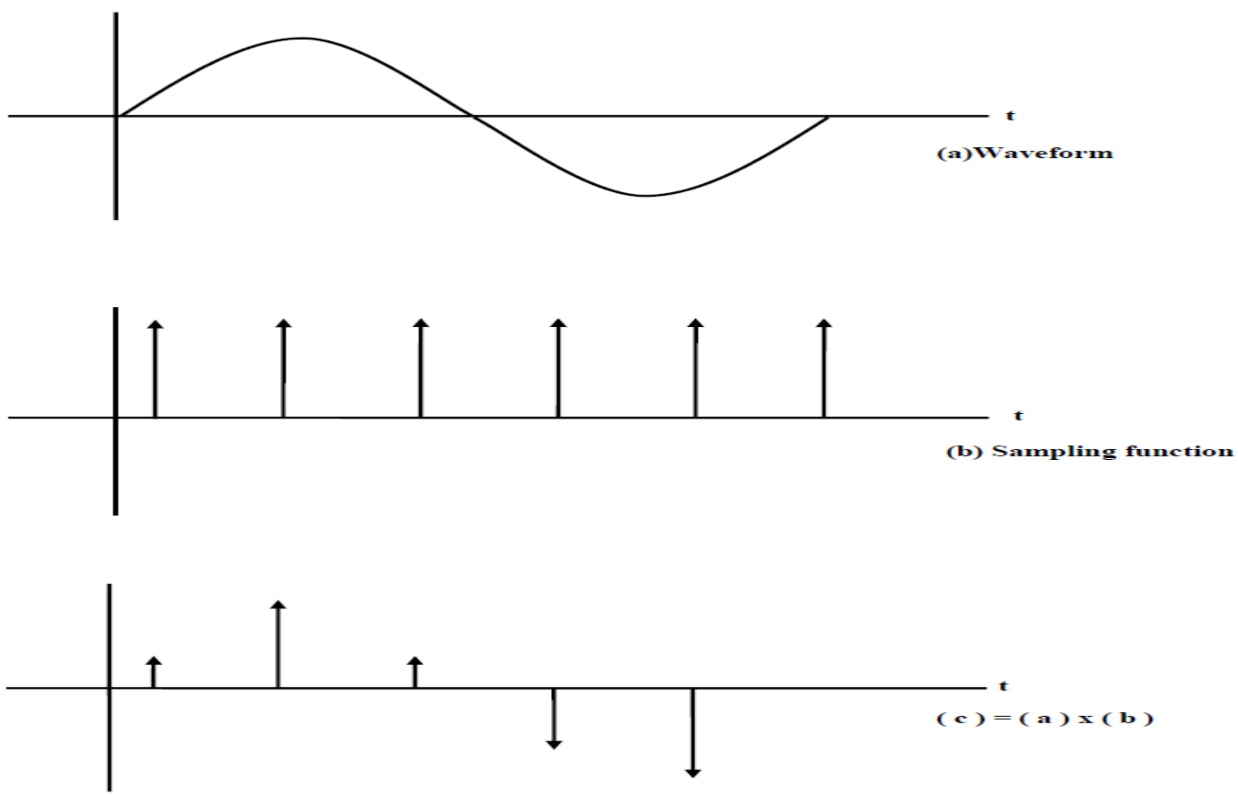
The FM and PM waves look identical when their modulation index are identical. However if we change the modulating frequency f_m then m_f will change but there is no change in value of m_p.

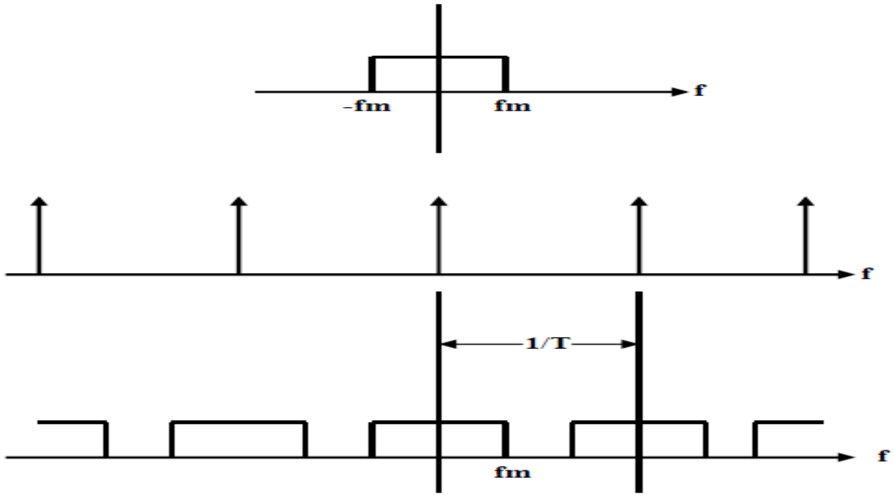
Result: study of modulation & demodulation in Phase modulation is completed.

EXPRIMENT NO.4

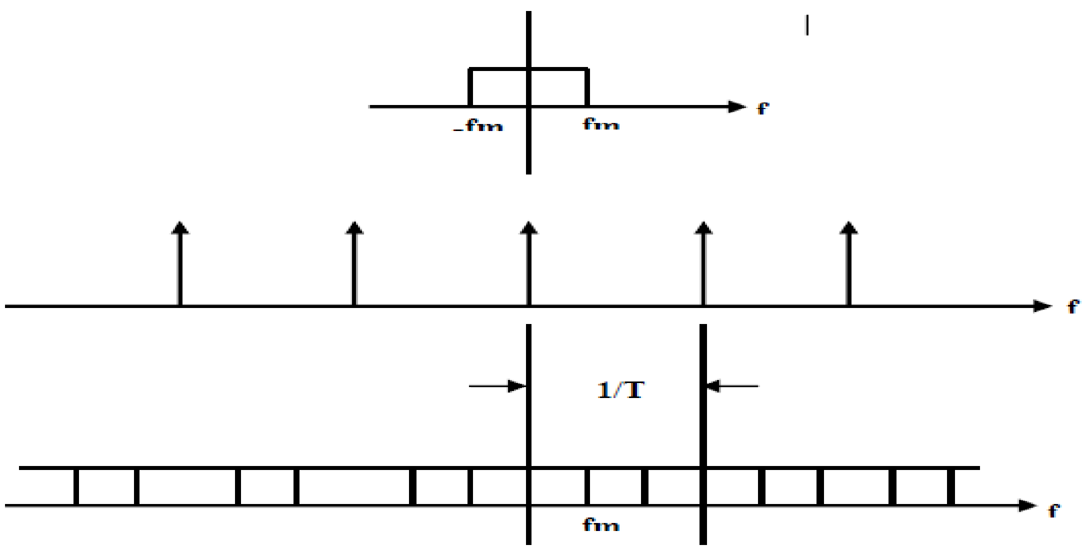
Objective: Perform signal sampling and reconstruction techniques.

Theory: In certain communication processes, such as pulse code modulation system described. it is necessary to sample a waveform at regular intervals in order to communicate discrete information rather than continuous information. The process of sampling is equivalent to multiplying the waveform to be sampled by a series of regularly spaced delta functions as shown in fig.Such a series of delta pulses is termed the sampling function which has the interesting Property that an infinite series of delta pulses in the time domain has a spectrum which is also an Infinite delta series in the frequency domain. Communications engineers often have to work Simultaneously in both the frequency and the time domain, and probably the best known rule which connects manipulations in these two domains is that a 'multiplication of waveforms in the time domain transforms in the convolution of their corresponding amplitude spectra in the frequency domain'. Convolution may sound as if it's a difficult process, and indeed may be so mathematically, but it is in fact a simple geometrical process which is described in more detail in the appendix to this chapter. Thus if sampling the multiplication of the analog waveform by a delta series in the time domain, the spectrum of the sample signal is the convolution of the analog waveform spectrum with another delta series. this is shown in fig.





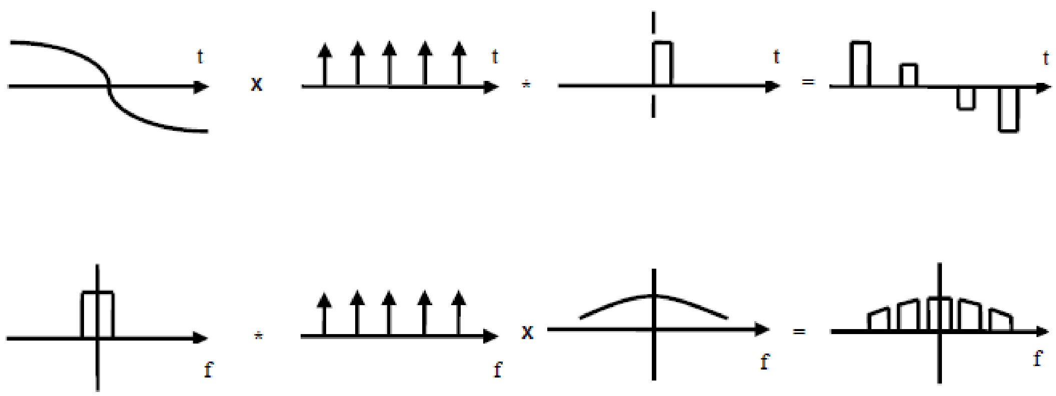
If T is the interval between pulses in the time domain, i.e. in fig.2.2, then the corresponding interval between the frequencies which contain signal energy is 1/T. Consider an analog waveform which has a spectrum which extends from zero Hz to an upper limit of f m Hz. It can now be seen that provided 1/T is greater than 2f m then a complete replica of the spectrum of the sampled signal lies below the frequency 1/2T and the introduction of a low pass filter would restore the original signal unchanged. If, however, the frequency 1/T is less than 2f m then overlap of the spectra of the sampled signal will occur resulting in distortion. This mechanism of distortion is sometimes referred to as 'aliasing' and is shown in fig.2.4. The preceding argument holds true even if the analog signal spectrum begins at a frequency above zero Hz, and indeed in the limit it could be a sinusoid with energy only at frequency f m . In general, therefore, if a waveform has frequencies in its spectrum extending from a lower frequency limit to an upper frequency limit f m Hz it is possible to convey all the information in that waveform by 2f m or more equally spaced samples per second of the amplitude of the waveform. This rate is often referred to as the Nyquist sampling rate.



In practice, an analog signal is sampled with pulses which have a nonzero width. The way in which the pulse width affects the spectrum will be used to demonstrate the effect in practice. Thinking geometrically again, a series of a broad samples is a delta series convolved with a single broad pulse. Using x for the multiplication process and * for the convolution process, fig.2.5(a) shows the procedure in constructing the practical sampled waveform. Using the rule referred to before where x and * change places when moving between time and frequency domains, fig. 2.5(b) shows how the practical spectrum is constructed. The extra ingredient used in moving from fig. 2.5(a) to fig.2.5(b) is the relationship between a square pulse and its amplitude density spectrum. This is shown in fig.2.6 which also shows that the narrower the pulse the broader the amplitude density spectrum between its central zero crossing points. In the limit of a zero width(delta) pulse this amplitude density spectrum is flat, which, when multiplied by any other spectrum, does not alter its shape, as is excepted.

(a)

(b)



Result: Study of sampling theorem is completed.

EXPRIMENT NO.5

Objective: Perform the TDM pulse amplitude modulation/demodulation & draw their waveform in the Graph.

Apparatus Required: TDM kit, Connecting wires, CRO.

Theory: Time-division multiplexing (TDM) is a type of digital or (rarely) analog multiplexing in which two or more signals or bit streams are transferred apparently simultaneously as sub-channels in one communication channel, but are physically taking turns on the channel. The time domain is divided into several recurrent timeslots of fixed length, one for each sub-channel. A sample byte or data block of sub-channel 1 is transmitted during timeslot 1, sub-channel 2 during timeslot 2, etc. One TDM frame consists of one timeslot per sub-channel plus a synchronization channel and sometimes error correction channel before the synchronization. After the last sub-channel, error correction, and synchronization, the cycle starts all over again with a new frame, starting with the second sample, byte or data block from sub-channel 1, etc. By using quite short carrier pulses (small width), other sampled pulses can be placed in the gaps between the pulses of PAM (Pulse Amplitude Modulation). This process is called as Time Division Multiplexing (TDM). Therefore, more than one message signal can be carried via a single communication channel. The operational fundamentals of TDM will be examined by using the PAM, however TDM can be applied to other type of pulse modulations as well. If two messages are sampled at the same rate but at slightly different times, then two of trains of samples can be added without mutual interaction. In Figure-1.a the signal $x(t)$ and the corresponding PAM signal are depicted. Also in Figure-1.b, TDM of two signals is shown:

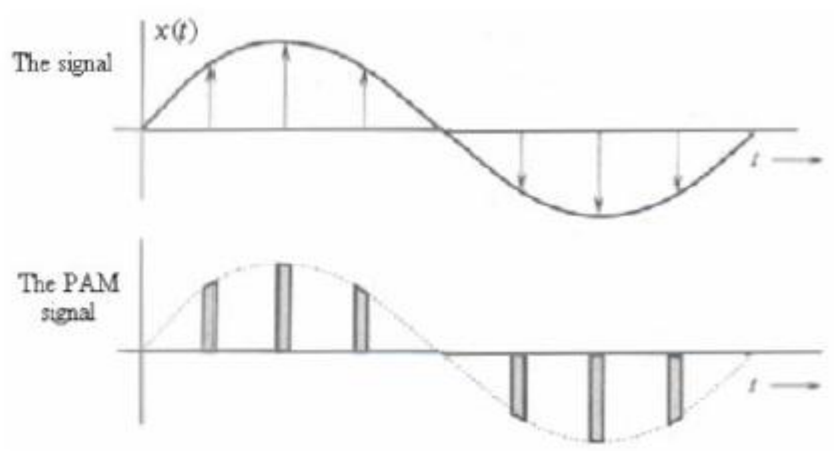


Figure 1. (a) The signal $x(t)$ and its PAM signal

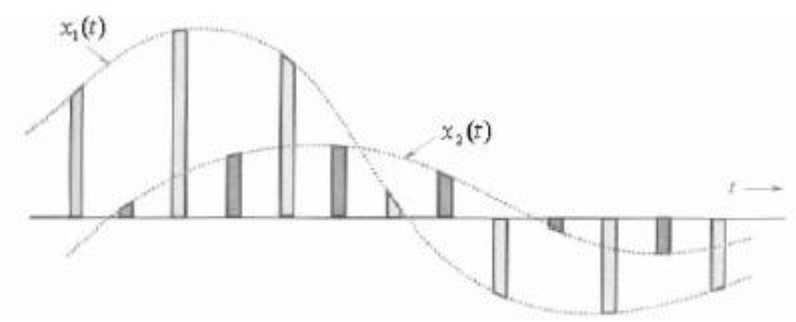


Figure 1 (b) TDM of two signals

The block diagram of PAM/TDM signal production is given in Figure 2 below:

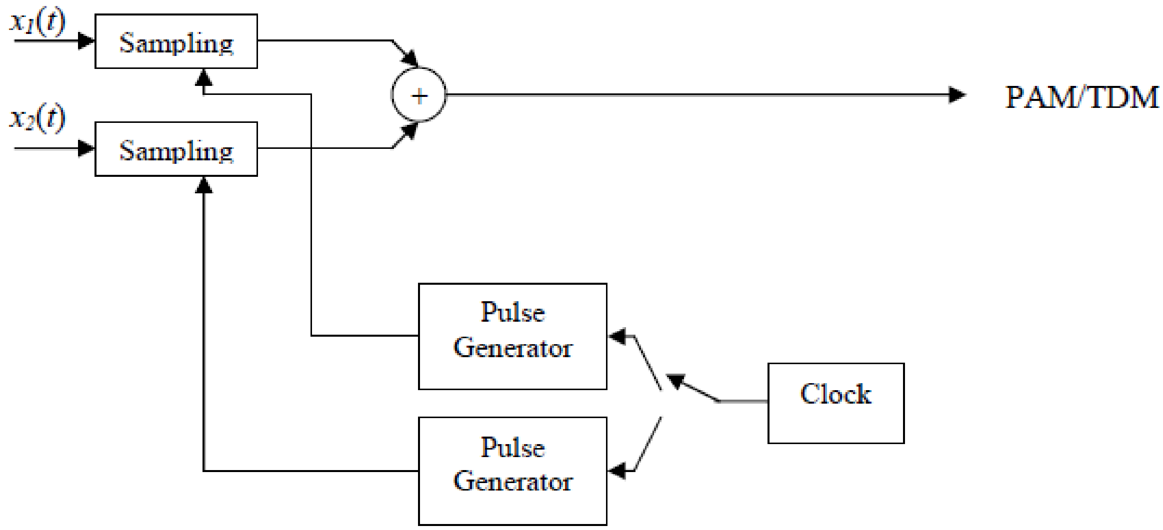


Figure 2. Production of TDM Signal

In Figure 2, if the band-widths of both signals (*x1*(*t*) and *x2*(*t*)) is 3 kHz, according to the sampling theorem each signal should be sampled with the frequency of 6 kHz. But in this case the clock frequency should be 12 kHz. The distance between the pulses is *Tn=Ts/n*. Here, n indicates the number of the input signals, and *Ts* denotes the sampling period required for one signal. The obtained TDM wave is shown in Figure 3.

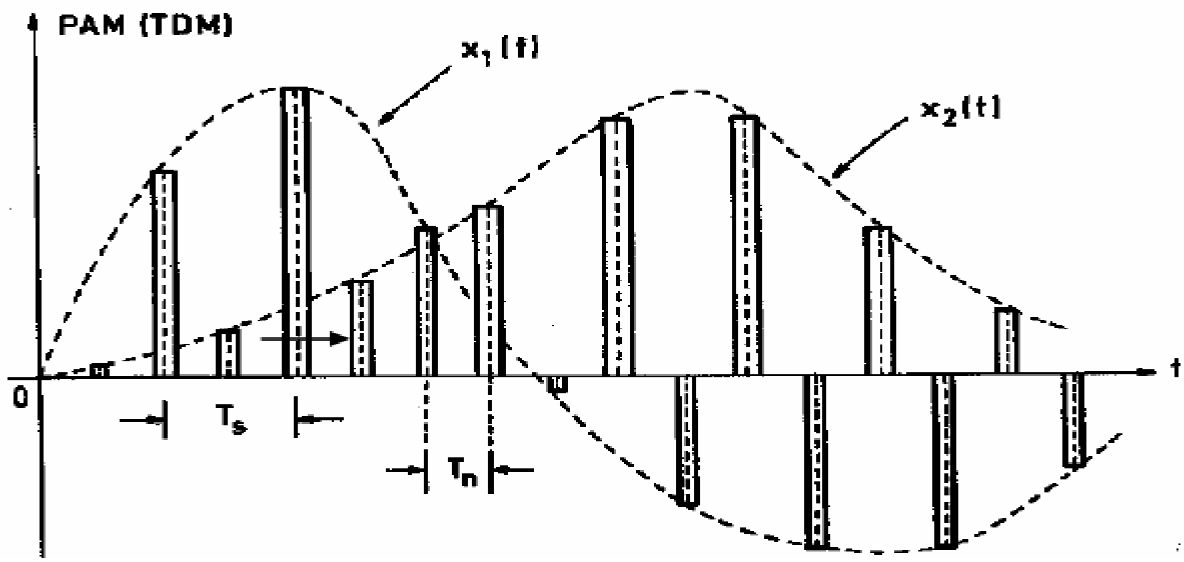
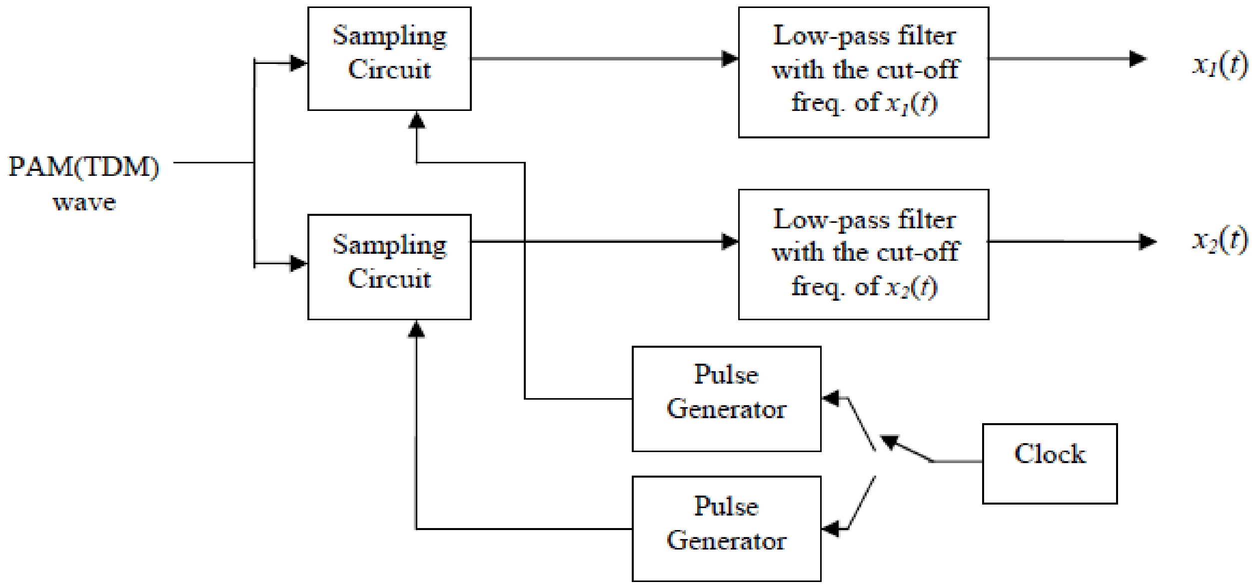


Figure 3. PAM(TDM) Wave

As considered before, if two different signals have the frequency of 3 kHz, the sampling period of each signal will be *Ts*=1/6000 = 166.7 µsec. Since number of input signals n=2, from *Tn=Ts/n*; the distance between samples becomes *Tn=Ts/2* = 83.3 µsec. Therefore the minimum bandwidth to transmit these samples by TDM should be; *B*≥ (1 / 166.7.10 $^{-6}$) =6 kHz. A TDM receiver block diagram is shown in Figure 4 below. The most significant issue inrecovery of input signals from

TDM signals is the requirement for the proper synchronization between TDM transmitter and receiver. Therefore, the clock signal in the transmitter should be passed to the receiver correctly.



Result : Study of Time division multiplexing process is completed.

EXPERIMENT NO.6

Object: Perform the division multiplexing pulse code modulation/demodulation**.**

Apparatus required: PCM modulator/demodulator kit, connecting wires.

Theory: Pulse Code Modulating (PCM) is different from Amplitude Modulating (AM) and Frequency Modulating (FM) because, those two are continuous forms of modulation. Pulse Code Modulation (PCM) is used to convert analog signals into binary form. In the absence of noise and distortion it is possible to completely recover a continuous analog modulated signals. But in real time they suffer from transmission distortion and noise to an appreciable extent. In the PCM system, groups of pulses or codes are transmitted which represent binary number corresponding to Modulating Signals Voltage levels. Recovery of the transmitter information does not depend on the height, width, or energy content of the individual pulses, but only on their presence or absence. Since it is relatively easy to recover pulses under these conditions, even in the presence of large amounts of noise and distortion, PCM systems tend to be very immune to interference and noise. Regeneration of the pulse enroute is also relatively easy, resulting in system that produce excellent result for long distance communication.

The decoding process reshapes the incoming pulses and eliminates most of the transmission noise. A serial to parallel circuit passes the bits in parallel groups to a digital to analog convertor (D/A) for decoding. Thus decoded signal passes through a sample and hold amplifier which maintains the pulses level for the duration of the sampling period, recreating the staircase waveform approximation of the modulating signals. A low-pass filter may be used to reduce the quantization noise.

Features:

- Low cost
- ± 10 V input ranges
- No missing codes
- Ratiometer conversion
- Tri-State outputs
- Fast (Tc = 50μs)
- Contains outputs
- TTL compatible
- Supply voltages (5 VDC and -12 VDC)
- Resolution (8-bits)
- Linearity (±1 LSB)
- Conversion speed (40 clock periods)
- Clock Range (50 to 800 kHz)

PROCEDURE

STEP-1: PCM MODULATION WITH D.C. INPUT

1. ‘Switch ON’ the experimental kit.
2. Observe the Basic clock generator outputs and sampling pulse output.
3. Connect the sampling pulse generator output to CH1 of the CRO and trigger CRO w.r.t. CH1 only.
4. Observe the output of the parallel to serial convertor output (PCM data) on the CH2 of the CRO.
5. Make sure that CRO is triggered with the positive going edge of the sample pulse generator.

6. Now connected the variable DC output to the input of the PCM Modulator.
7. Adjust the Time/div Switch of the CRO such that two samples can be seen at a time on the screen.
8. Now vary the D.C. voltage from its minimum to the maximum.
9. At each step observe the parallel data displayed by the LEDs at the ADC output and compare the PCM output (Parallel to serial converter), which is the same of ADC output but is in serial form.
   Note : Between two samples, 8-bit serial data will transmitted.

STEP-II : PCM DEMODULATION WITH DC INPUT

1. Connect the PCM output to the PCM demodulator.
2. Output of the serial to parallel convertor displayed by the LEDs is the same with is displayed by the ADC output LEDs.
3. Observe the output of the D/A converter.
4. Observe the output of the low pass filter and adjust the potentiometer such that the output D.C. voltage is equal to the D.C. input at the PCM Modulator.
   Note: Output D.C output is 1800 out of phase to the input because D to A convertor introduce 1800 out of phase and low pass filter also introduces some delay. Because in all practical PCM systems negative logic is used to reduce the noise in transmission.

STEP-III : PCM MODULATION WITH AC INPUT

1. Now remove DC input and connect the AC voltage to the input of the PCM modulator.
2. Observe the PCM output with follows the sequence of the AC input.
3. Here one has to make sure that like DC input, we cannot see the stable digital output at the PCM modulator output. Because this is dynamic process and with AC input, we cannot send same PCM data between successive samples. But in the DC input case at any sample same voltage is available not like AC input.

Result:

EXPRIMENT NO.7

Objective: Perform the delta modulation techniques and plot the waveforms.

Apparatus Required: Delta modulator/Demodulator kit, connecting wires.

Theory: Delta Modulation (DM) is a simplified PCM. In some type of signals, the neighboring samples are closely correlated with each other. Therefore, once a sample value is known this enables the determination of the following sample values most probably. Thus, instead of sending the real value of each sample at each time, differences (variances) between adjacent samples are sent in DM. In DM, two-level quantizer and one-bit coding is used. Transmitted code pulses do not carry the data related to the message signal itself; instead they carry data regarding the differentials of the message function. The output of a delta modulator is a bit stream of samples at a relatively high rate, the value of each bit being determined according to whether the input message sample amplitude has increased or decreased relative to the previous sample. The operation of a delta modulator is: **i)** periodically sample the input message, **ii)** make a comparison between the current sample and the preceding one and, **iii)** give a single bit as output which indicates the sign of the difference between two samples. Delta Modulator and Demodulator block diagrams are given in Figure 1:

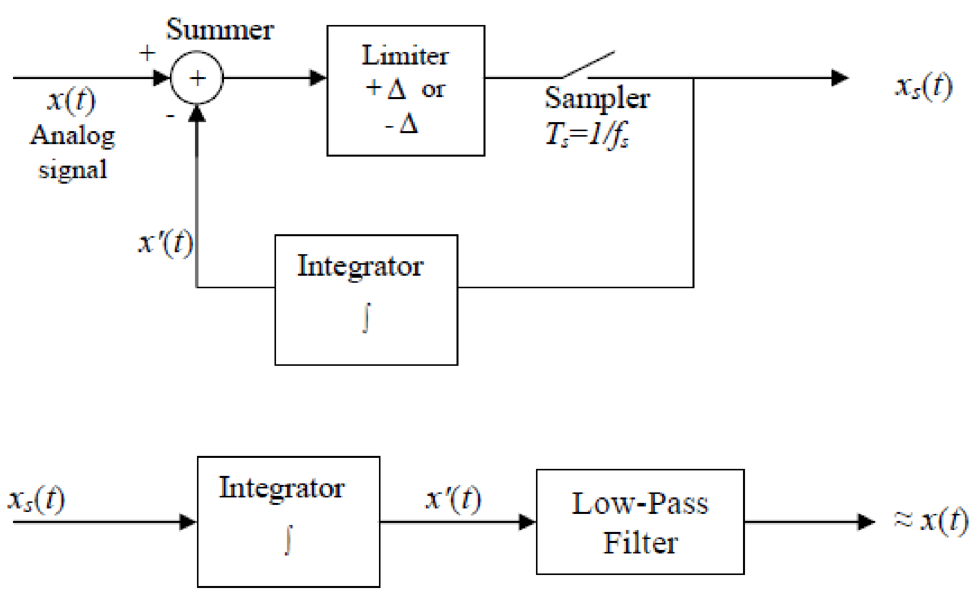


Figure 1. Delta Modulator & Demodulator

The system is in the form of a feedback loop. It is a continuous-time to discrete-time converter. In fact, it is a form of analog to digital converter. After the sampler is clocked, the resulting signal is the delta modulated signal. The output from the sampler is a bipolar signal, in block diagram being either $\pm\Delta$ volts. If the output of 'Summer' (or comparator) is positive than the sample value of DM signal is $+\Delta$, otherwise it is $-\Delta$. The waveform of the DM signal is shown in bottom of Figure 2. It is fed back, in a feedback loop, via an integrator, to a summer.

The integrator output is a saw tooth like waveform as shown in Figure 2.

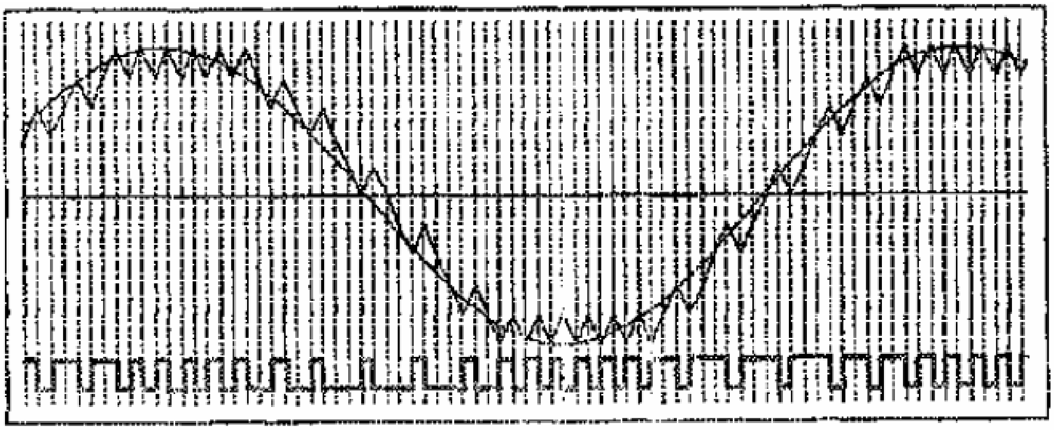


Figure 2. Integrator output superimposed on the message with the Delta Modulator signal in the bottom of the figure.

The saw-tooth waveform is subtracted from the message and the difference – called as error signal – is the signal appearing at the summer output. An amplifier can also be used in the feedback loop (though not drawn in Figure 1) to control the loop gain and the size of the 'teeth' of the saw-tooth waveform. Signal from the integrator, which is a saw-tooth approximation to the message, is adjusted with the amplifier to match it as closely as possible.The binary waveform illustrated at the bottom of Figure 2 is the signal transmitted. This is the delta modulated signal as stated above. The integral of the binary waveform is the saw-tooth approximation to the message. Low pass filtering of the saw-tooth (from the demodulator) gives a better approximation process. The unwanted products of the modulation process, observed at the receiver, are of two kinds. These are due to 'slope overload' and 'granularity'; those will not be examined in the content of this experiment In order to prevent some inappropriate modulation, the pace Δ should be selected according to the following equation. $\Delta f_s > 2\pi f_x$; here f_s is the sampling frequency, and f_x is the greatest frequency component of the input signal

Experimental Procedure

• Generate a sine-wave signal to be delta modulated by using LabView software functions so that the display duration *T* and the duration between each sample *dT* can be changed externally by the user.
• Design a delta modulator vi-file by considering the Figure-1. During the design process, choosing array functions of LabView will be useful and efficient. For instance, integrator function will be performed by using arrays for obtaining the saw-tooth signal.
• Display the original input signal and saw-tooth wave in the same waveform graph screen. Display the DM signal as a separate waveform.
• After low-pass filtering the saw-tooth wave, constitute the recovered (DM demodulated) sine-wave signal and display it in a third waveform graph screen.
• Observe all the resulting waveforms by changing the Δ pace of delta modulation. Compare the recovered signals for very small and very big Δ steps

Result: Study of delta modulation techniques and plot the waveforms is completed.

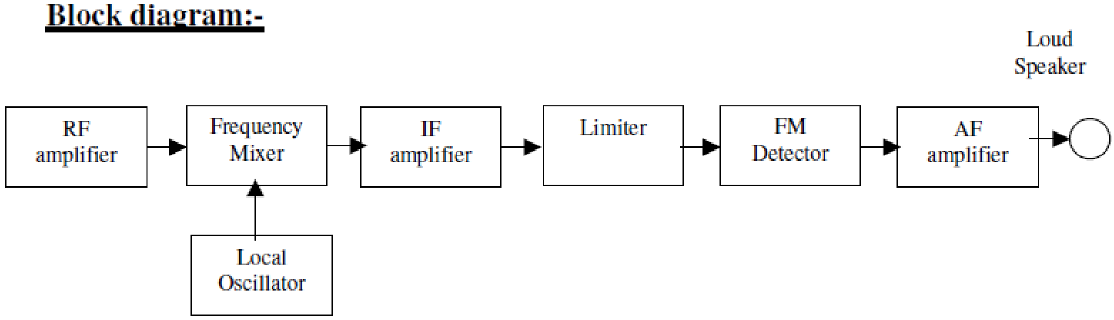
EXPRIMENT NO.8

Objective: Study of FM radio receiver.

Apparatus Required: FM receiver kit.

Theory: The block diagram of FM radio receiver is shown in the diagram. The constituent stages of the FM receiver are as follows.

Block diagram:-



(1) R.F.Amplifier: It increases the level of the signal level appreciably before the signal is fed to the mixer and it also helps images frequency rejection. In FM broadcast the signal bandwidth is large being 150 KHz therefore the RF amplifier must be designed to handle this large bandwidth.
(2) Frequency mixer: It performs the usual function of mixing or heterodyning the signal frequency voltage and the local oscillator voltage to produce the difference voltage and frequency voltage which is the intermediate frequency voltage. Since FM broadcast takes place either in VHF or UHF band single transistor frequency converter is not used. Separate local oscillator is always used and another transistor serves as a frequency mixer. The IF used in FM receiver is higher than that in AM. Typical value of intermediate frequency is 12MHz. This high IF helps in image rejection.
(3) Local oscillator: A separate local oscillator is always used. At ultrahigh frequencies it is preferred to keep the local oscillator frequency smaller than the signal frequency by an amount equal to the IF.
(4) IF amplifier: A multistage IF amplifier is used to provide large gain. Further this IF amplifier should be designed to have high overall bandwidth of the order of 150 KHz. Double tuned circuits may be used but it is preferred particularly at the higher frequencies in the UHF range, to use stagger tuned single tuned circuit which are found to produce more gain bandwidth product than the conventional double tuned circuits.
(5) Limiter: The IF amplifier is followed by limiter which limits the IF voltage to predetermined level and thus removes all amplitude variations which may be incidentally caused due to changes in the transmission path or by manmade static or natural static.
(6) FM detector: This extracts the original audio modulation frequency voltage from the frequency modulated carrier voltage. A discriminator is used as a frequency detector.
(7) Audio amplifier: The output of the FM detector is fed to an audio frequency small signal amplifier and one or more audio frequency large signal amplifiers. The output audio voltage is then fed to the loud speaker. In FM broadcast, the maximum modulating frequency permitted is 15 KHz and hence the audio frequency must be designed to accommodate such large bandwidth. Similarly the loud speaker must be capable of reproducing all high frequency tones up to 15 KHz. Often two or more loud speaker are used each reproducing a limited range of frequencies.

Result: Study of FM receiver is completed.

EXPRIMENT NO.9

Objective: Perform the modulation & demodulation in ASK, draw its waveforms.

Apparatus Required: ASK modulator/demodulator kit.

Theory: Amplitude shift keying - ASK - in the context of digital communications is a modulation process, which imparts to a sinusoid two or more discrete amplitude levels. These are related to the number of levels adopted by the digital message. For a binary message sequence there are two levels, one of which is typically zero. Thus the modulated waveform consists of bursts of a sinusoid.

Figure 1 illustrates a binary ASK signal (lower), together with the binary sequence which initiated it (upper). Neither signal has been band limited.

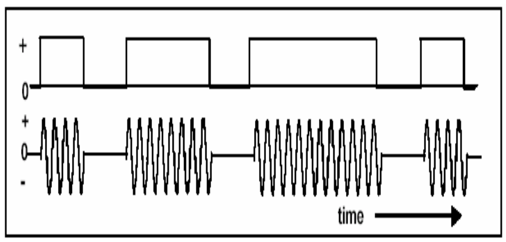


Figure 1: an ASK signal (below) and the message

There are sharp discontinuities shown at the transition points. These result in the signal having an unnecessarily wide bandwidth. Bandlimiting is generally introduced before transmission, in which case these discontinuities would be 'rounded off'. The band limiting may be applied to the digital message, or the modulated signal itself.

The data rate is often made a sub-multiple of the carrier frequency. This has been done in the waveform of Figure 1. One of the disadvantages of ASK, compared with FSK and PSK, for example, is that it has not got a constant envelope. This makes its processing (eg, power amplification) more difficult, since linearity becomes an important factor. However, it does make for ease of demodulation with an envelope detector.

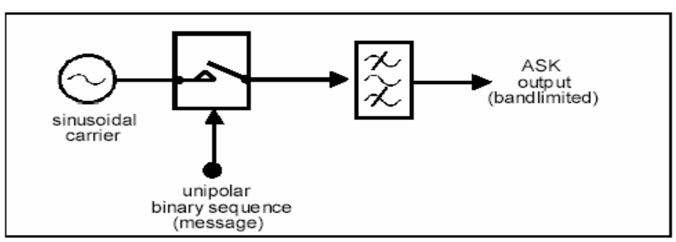


Figure 2: ASK generation method

Figure 3 shows the signals present in a model of Figure 2, where the message has been bandlimited. The shape, after bandlimiting, depends naturally enough upon the amplitude and phase characteristics of the bandlimiting filter.

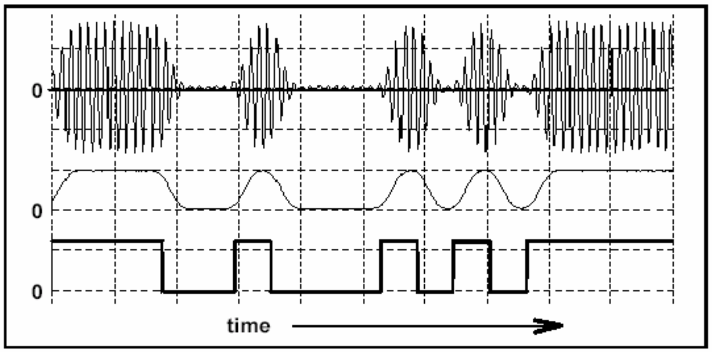


Figure 3: original TTL message (lower), bandlimited message (center), and ASK

It is apparent from Figures 1 and 4 that the ASK signal has a well defined envelope. Thus it is amenable to demodulation by an envelope detector. With bandlimiting of the transmitted ASK neither of these demodulation methods (envelope detection or synchronous demodulation) would recover the original binary sequence; instead, their outputs would be a bandlimited version. Thus further processing - by some sort of decision-making circuitry for example - would be necessary. Thus demodulation is a two-stage process:

1. recovery of the bandlimited bit stream
2. regeneration of the binary bit stream

.

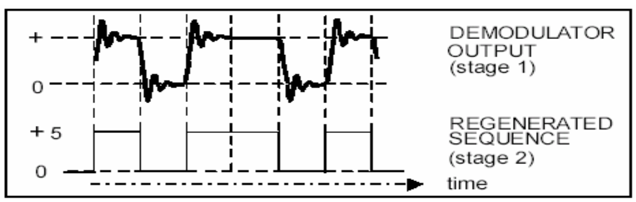


Figure 4: the two stages of the demodulation process

Modeling an ASK Generator

It is possible to model the rather basic generator shown in Figure 2.
The switch can be modeled by one half of a DUAL ANALOG SWITCH module. Being an *analog* switch, the carrier frequency would need to be in the audio range. The TTL output from the SEQUENCE GENERATOR is connected directly to the CONTROL input of the DUAL ANALOG SWITCH. For a synchronous carrier and message use the 8.333 kHz TTL sample clock (filtered by a TUNEABLE LPF) and the 2.083 kHz sinusoidal message from the MASTER SIGNALS module.
If you need the TUNEABLE LPF for bandlimiting of the ASK, use the sinusoidal output from an AUDIO OSCILLATOR as the carrier. For a synchronized message as above, tune the oscillator close to 8.333 kHz, and lock it there with the sample clock connected to its SYNCH input.
This arrangement is shown modeled in Figure 5.

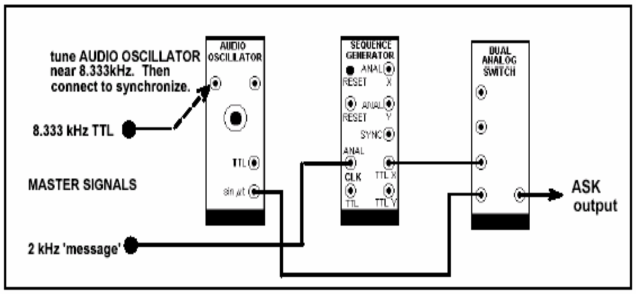


Figure 5: modeling ASK with the arrangement of Figure 2

Demodulation of an ASK signal

Having a very definite envelope, an envelope detector can be used as the first step in recovering the original sequence. Further processing can be employed to regenerate the true binary waveform.

Figure 6 is a model for envelope recovery from a baseband ASK signal.

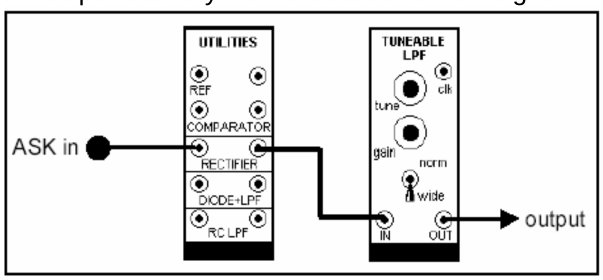


Figure 6: envelope demodulation of baseband ASK

The output from the above demodulators will not be a copy of the binary sequence TTL waveform. Bandlimiting will have shaped it, as (for example) illustrated in Figure 3. If the ASK has been bandlimited before or during transmission (or even by the receiver itself) then the recovered message, in the demodulator, will need restoration ('cleaning up') to its original bi-polar format. Some sort of decision device is then required to regenerate the original binary sequence.

Result: Study of modulation & demodulation in ASK is completed.

EXPERIMENT NO 10

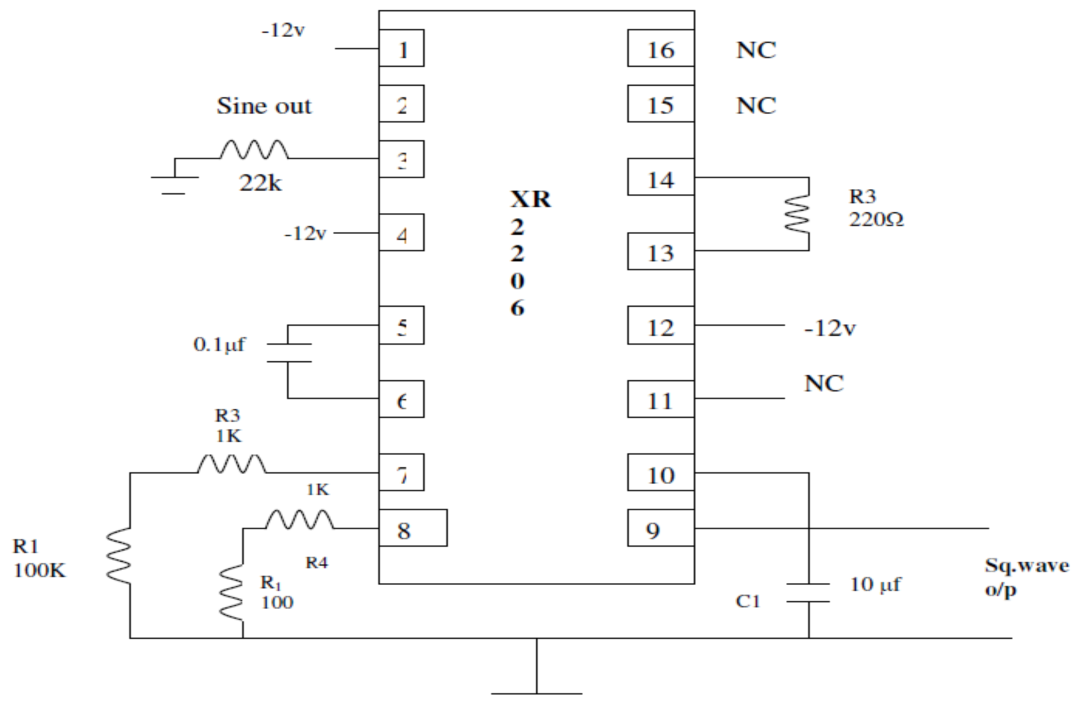
Objective: Perform the modulation & demodulation in FSK, draw its waveforms.

Requirements:- Breadboard , CRO dual channel, function generator, power supply, CRO probes, connecting wires.

Components:-IC:-XR2206, Resistors:-22K,1K(2),100K(2),220WCapacitors-1mf,10mf

Theory:. Frequency shift keying is the digital system of frequency modulation .Digital signal generated in system like telegraphy is not transmitted as it is instead they are transmitted using keying techniques .In FSK the nominal unmodulated carrier frequency .Corresponding to mark condition and space condition is transmitted for logic level 1 and logic level 0 respectively.

Circuit Diagram:-



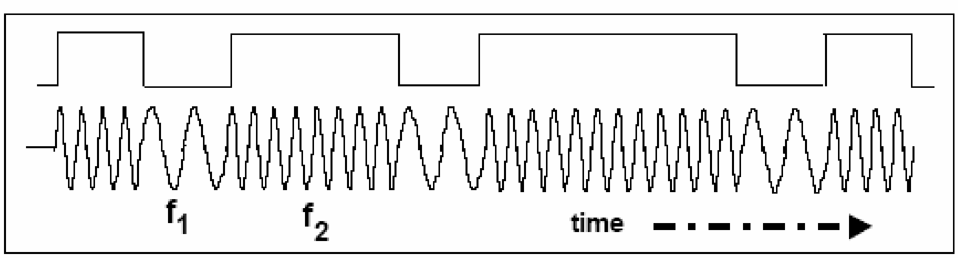


Figure 1: an FSK waveform, derived from a binary message

Procedure:-

1. Assemble components and make connection on breadboard as shown in fig.
2. Get it checked before turn on the power supply .
3. Apply square wave input.
4. Observe the waveform on CRO & note down mark & space frequencies.
5. Write a MATLAB program to generate FSK signal.
6. Take the print out of the simulation result.

Observation:-

The standard mark frequency f(mark)= 1/R1C3
Space frequency f(space)= 1/R2C3
Observe I/P square wave
Frequency=
Amplitude =
Observe O/P sine wave
Mark frequency=
Space frequency=

Result: - Mark frequency is ……. And Space frequency is ……
Conclusion: -Mark frequency is greater than space frequency. FSK system is used for digital data transmission.

EXPRIMENT NO.11

Objective: Perform the modulation in PSK, draw its waveforms.

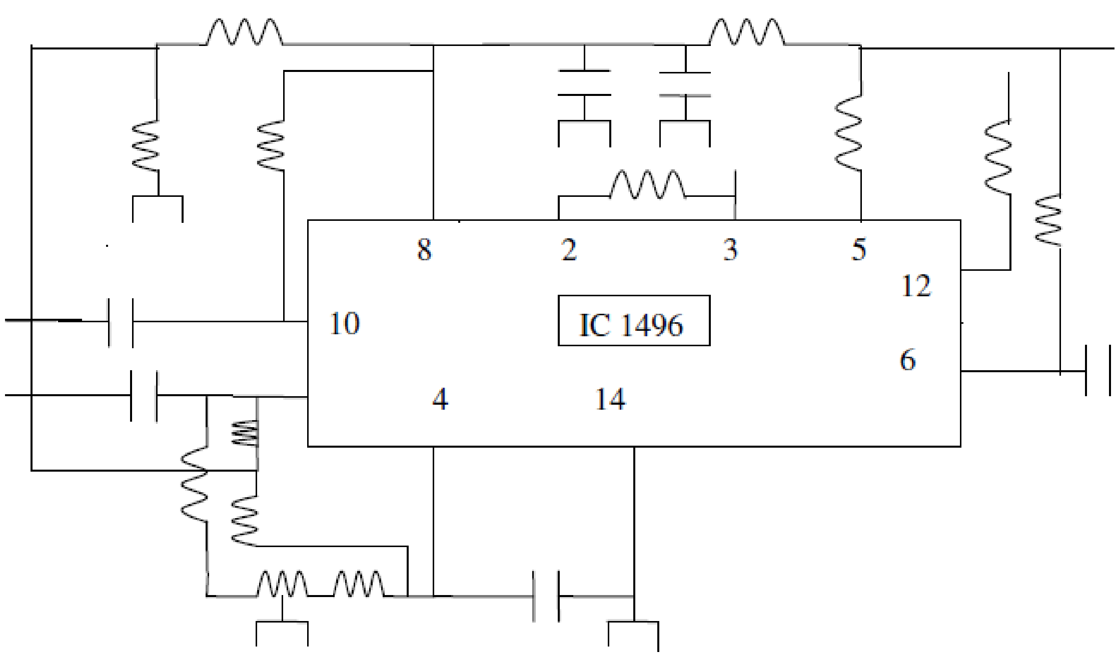
Apparatus Required: PSK kit, Function generator dual channel, CRO, Digital multimeter, IC1496.

Theory: The original source of information, text, speech, the most commonly used coding scheme is binary sequence such as 0011101011. For transmission purpose this has to be converted to a continuous electrical waveform, conversion process is referred to as the modulation .The o/p of the PCM system is also a binary data. If they are to be transmitted over copper wires, they can be Directly Transmitted as two voltage levels +v and –v. But if they are to be transmitted through space using antenna, Phase Modulation is used. As the modulating signal consists of only two levels the modulation technique is known as Phase Shift Keying.

Phase Shift Keying is a modulation in which the phase of the carrier signal changes with respect to the digital signal. The binary signal to be transmitted changes the phase of the sine wave carrier depending upon whether 0 or 1 is transmitted. It is also called as Binary Phase Shift Keying (BPSK).

The BPSK can be implemented by using a Balanced Modulator. IC1496 is used as a Balanced Modulator for implementing BPSK. The circuit diagram is shown in the manual. The carrier is applied to pin no.8 and Modulating Binary signal is applied to pin no.1 The BPSK O/P is taken from pin no. 12.IC 1496 internally consists of differential amplifier configuration .its carrier suppression is rated at a minimum of –5db with a typical value of –65db at 500khz.

Circuit Diagram:-



Procedure:

1) Study the circuit provided on the front panel of kit.
2) Apply the sine wave of 400Hz to pin no. 8.
3) Apply the square wave i.e. non linear binary data minimum 200Hz pin no.1
4) Connect CRO at the o/p i.e. pin no. 12.
5) Switch ON the power supply.
6) Observe the PSK O/P on the CRO.
7) Draw the observed waveform on the graph paper.

Observation Table:

Sr. no.	Carrier signal		Modulating signal	
	Amplitude (V)	frequency (KHz)	Amplitude(V)	Freq(KHz)

Result: The carrier signal changes phase as Binary signal changes its state from logic 0 to logic 1.

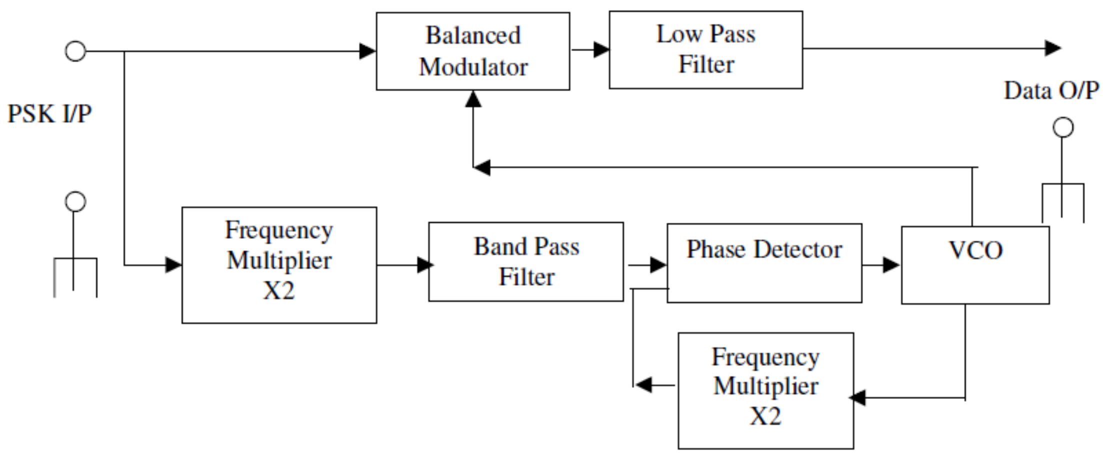
EXPRIMENT NO.12

Objective: Perform the Demodulation in PSK, draw its waveforms

Apparatus Required: PSK generator, CRO and connecting probes.

Theory:
Demodulation of BPSK signal is done with a balanced modulator. The balanced modulator is constructed using IC for better resolution and sensitivity, while it could be constructed using diode and transformer. The BPSK signal and carrier signal is applied to the balanced modulator IC for demodulation of the binary signal that is embedded into the carrier signal. The key to demodulating BPSK is that a carrier with the correct frequency and phase relationship must be applied to the balanced modulator along with the BPSK signal.

Circuit Diagram:



Procedure:

(1) Study the circuit given on the front panel of the kit.
(2) Provide a PSK signal at the input terminal of the kit.
(3) Connect the CRO probe at the output terminal for the kit.
(4) Switch on the circuit.
(5) Vary the rate of digital data in order to check the output from the CRO, which Reproduces the data that is being transmitted.

Result: The PSK Demodulator circuit gives the original binary transmitted data at the o/p. And as the bit rate at the I/P of PSK modulator changes the O/P also changes accordingly.

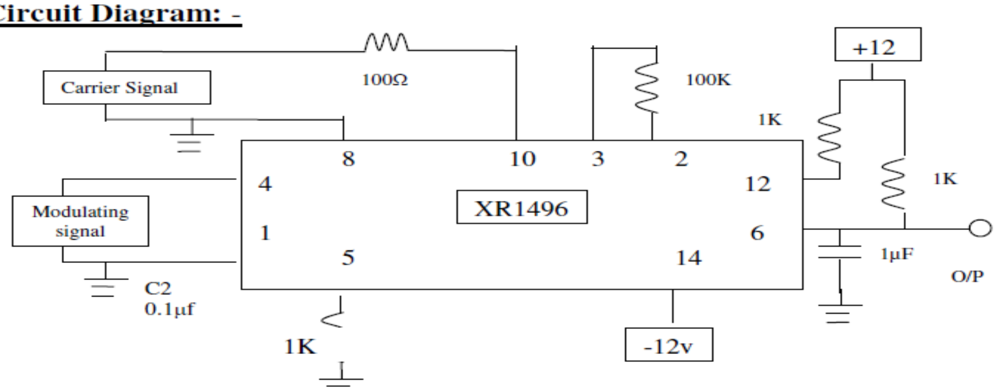
EXPRIMENT NO.13

Objective: Observe DSB/SSB AM transmitter waveforms and plot the graph

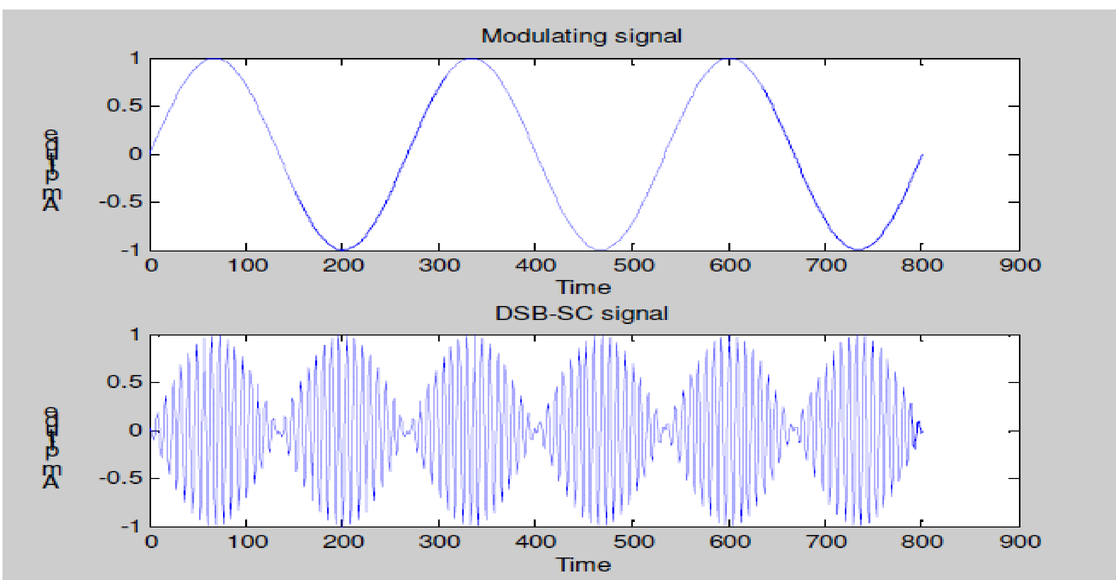
Apparatus Required: Function Generator, CRO, Connecting wires. IC 1496 (1 nos.), Resistors: 1KW (3 nos.), 100W (2 nos.)

Theory: The amplitude-modulated signal is simple to produce but has two practical drawbacks in
application to many real communications systems: the bandwidth of the AM signal is twice that of the modulating signal and most of the power is transmitted in the carrier, not in the information bearing sidebands. To overcome these problems with AM, versions on AM have been developed. These other versions of the AM are used in applications were bandwidth must be conserved or power used more effectively.
If the carrier could somehow be removed or reduced, the transmitted signal would consist of two information-bearing sidebands, and the total transmitted power would be information. When the carrier is reduced, this is called as ***double sideband suppressed carrier*** AM or DSB-SC. Instead of two third of the power in the carrier, nearly all being the available power is used in sidebands.

Circuit Diagram: -



Simulation Result:



Procedure:

1. Assemble components and make connections of the breadboard.
2. Adjust the function generator to obtain the sine wave of frequencies 2 KHz and 1 MHz respectively.
3. Apply the sine wave of modulating signal and carrier signal to the ckt.
4. Observe the resulting DSB-SC signal on CRO.
5. Draw the time domain and frequency domain representation of DSB-SC on graph.

Observation Table:

Sr. No.	Signals	Frequency	Amplitude in V_s
	Modulating signal		
	Carrier signal		

Result: The Balanced modulator IC 1496 can be used for the generation of DSB-SC signal. The frequency domain representation of DSB-SC signal shows that DSB-SC signal do not contain the carrier component and it consists of two symmetrically placed sidebands.

EXPRIMENT NO.14

Objective: Study of EPABX machine

Theory: The electronics Private Branch Exchange (EPBX) is an essential element that supports the critical infrastructure of both government agencies and U.S. industry. An EPBX is a computer-based switch that can be thought of as essentially a small, in-house phone company for the organization that operates it. Protection of the PBX is thus a high priority. Failure to secure an EPBX can result in exposing the organization to toll fraud, theft of proprietary or confidential information, and loss of revenue or legal entanglements.

The EPBX system has the main three components:

1. Analog system and A/D converter
2. Cross connection matrix(Switches)
3. EPBX system

The threats to EPBX telephone systems are many, depending on the goals of attackers. Threats include:-Theft of service – i.e., toll fraud, probably the most common of motives for attackers Disclosure of information - data disclosed without authorization, either by deliberate Action or by accident. Examples include both eavesdropping on conversations or unauthorized access to routing and address data.

Data modification - data altered in some meaningful way by reordering, deleting or modifying it. For example, an intruder may change billing information, or modify system tables to gain additional services. Unauthorized access - actions that permit an unauthorized user to gain access to system resources or privileges.Denial of service - actions that prevent the system from functioning in accordance with its intended purpose. A piece of equipment or entity may be rendered inoperable or forced to operate in a degraded state; operations that depend on timeliness may be delayed.

Traffic analysis - a form of passive attack in which an intruder observes information about calls (although not necessarily the contents of the messages) and makes inferences, e.g. from the source and destination numbers, or frequency and length of the messages. For example, an intruder observes a high volume of calls between a company's legal department and the Patent Office, and concludes that a patent is being filed.

PBXs are sophisticated computer systems, and many of the threats and vulnerabilities associated with operating systems are shared by PBXs. But there are two important ways in which PBX security is different from conventional operating system security: External access/control. Like larger telephone switches, PBXs typically require remote maintenance by the vendor. Instead of relying on local administrators to make operating system updates and patches, organizations normally have updates installed remotely by the switch manufacturer. This of course requires remote maintenance ports and access to the switch by a potentially large pool of outside parties.Feature richness. The wide variety of features available on PBXs, particularly administrative features and conference functions, provide the possibility of unexpected attacks. A feature may be used by an attacker in a manner that was not intended by its designers. Features may also interact in unpredictable ways leading to system compromise even if each component of the system conforms to its security requirements and the system is operated and administrated correctly.

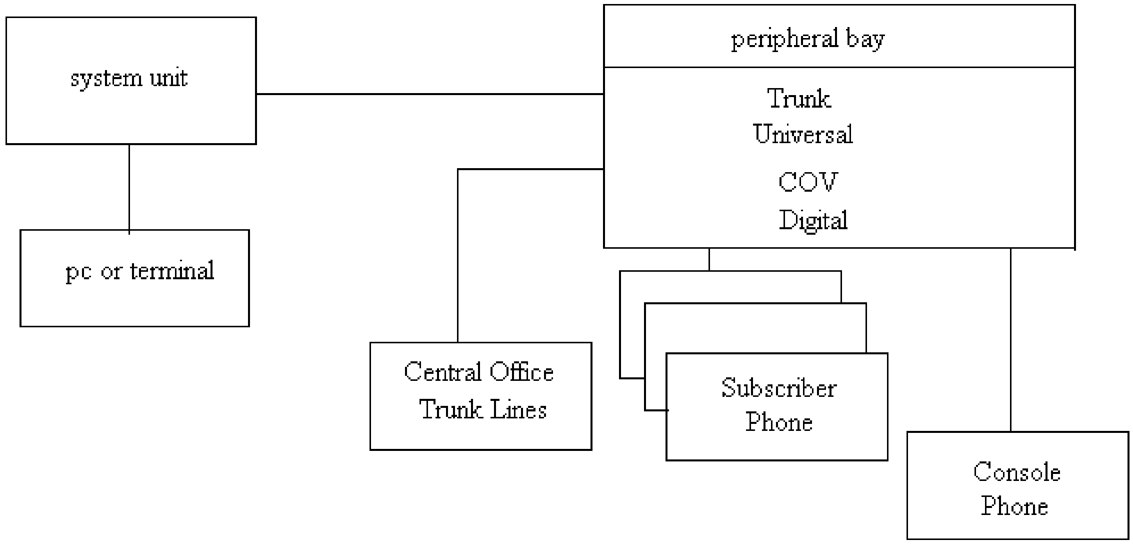


FIG: EPBX BLOCK DIAGRAM

SWITCHING ALGORITHM

Switching is performed using time division multiplexing techniques where each voice (digitized) and data port is assigned a time slot. Under control of the call processing routines, incoming time slots are connected to outgoing time slots. If the number of incoming slots does not exceed the number of outgoing slots, there will be no contention for switching resources. This is commonly known as non-blocking switching.

Dual Connections. To investigate for vulnerabilities, attempts should be made to route another incoming time slot to an outgoing time slot in addition to the intended time slot. This might be accomplished by a database entry or by a modification to the PBX control software. After accomplishing this, test calls should be made to verify the dual connection and to determine whether the calling or called party can detect the false connection. If the PBX under study has status or maintenance query features, attempts should be made to detect the modification.

The documentation accompanying the PBX forms the basis for learning its structure and operation. The manufacturer may have additional documentation that will be useful during the course of the evaluation. It may be beneficial to have technical discussions with the manufacturer to fully understand how PBX functions are implemented. Since this information is usually proprietary, it may be necessary to negotiate a non-disclosure agreement between the evaluating organization and the manufacturer to protect this data.

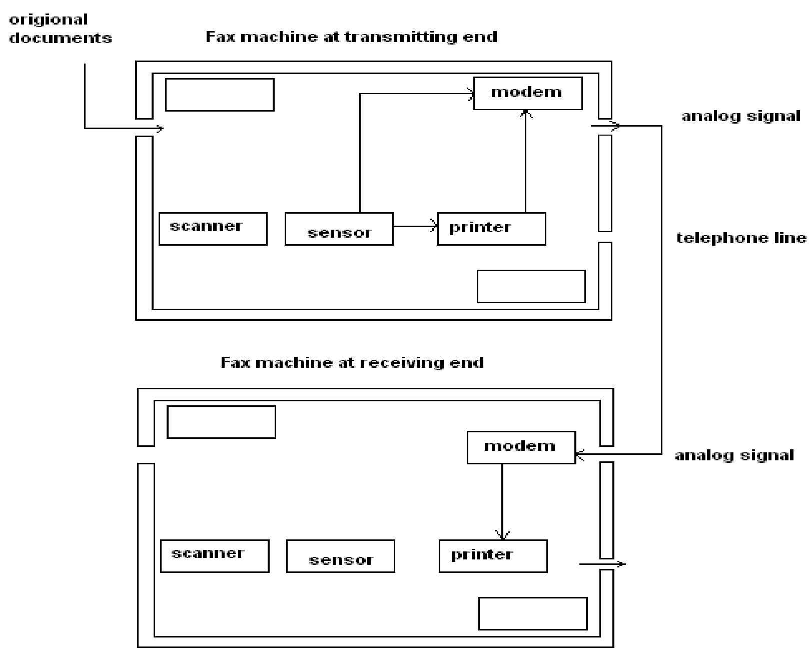
Result: Study of EPBX is completed.

EXPRIMENT NO.15

Objective: Study of FAX machine

Theory: A Fax machine is an electronic device used for transmitting and receiving printed matter via a telephone line. Fax machines are used to send printed material, photographs, or drawings. Facsimile transmission can be sent by several means. The most common way to send a fax is through a telephone line, however, today's technology is allowing us to send and receive faxes through our emails, cell-phones, and of course our hand-held organizers. In addition, radio, satellite, and cable are also means to send a fax transmission.

The essential parts of a fax system are the transmitting devices that translate the graphic material into electrical impulses according to a set pattern, and a synchronized receiving device that retranslates these impulses and prints that. In a typical system the fax scanner consists of a rotating cylinder, a source projecting a narrow beam of light, and a photoelectric cell. The copy to be transmitted is wrapped around the cylinder and is scanned by the light beam, which moves along the cylinder as it revolves. A block diagram of a Fax machine is shown in fig.



The output of the photoelectric cell is amplified and transmitted to the receiving end, where a similar cylinder, covered with specially impregnated paper, revolves in synchronism with the transmitting cylinder. A light of varying intensity moves along the rotation cylinder and darkens the paper by chemically reproducing the pattern of the original. When the fax is done, it pops out as a blueprint of what the other person sent.

One must also consider the art of sending a good fax machine. If you're not careful you can put too many papers in at once (jamming it), or accidentally write in the wrong color, which will make it hard for the other person to read.

Result: Study of Fax machine is completed.

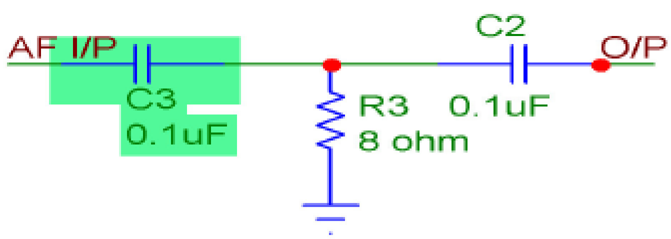
EXPRIMENT NO.16

Objective: Generation of pre-emphasis circuit on breadboard system & to plot pre-emphasis curve.

Apparatus Required: Breadboard system, dual channel CRO, function generator, CRO probes, connecting wires. Resistors of 10 K, 2.7 K. Capacitor of 0.1 μf. Inductor 0.2 mH.

Theory: The noise has greater effects on higher modulating frequencies than on lower ones. Thus, if the higher frequencies were artificially boosted at the transmitter and correspondingly cut at the receiver, an improvement in noise immunity could be expected, thereby increasing the signal-to-noise ratio.

Frequency modulation is more immune to noise than amplitude modulation & is significantly more immune than phase modulation. Natural tendency of audio is that the amplitude of high frequency signal is lower as compared to amplitude of low frequency signal. The S/N ratio reduces as the audio frequency increases. In turn the S/N ratio is not same over th e entire spectrum of audio signals. Thus in order to reduce the effect of noise the higher frequency signals are boosted before transmission. This is called **pre-emphasis**. When these signals are recovered at the receiver in order to restore the original amplitudes of higher frequency signals they are suppressed. This is known as deemphasis.



Procedure:-

1. Assemble components and make connections on breadboard as shown in fig.
2. Apply AF input.
3. Observe the waveform on CRO.
4. Vary the AF i/p frequency between 20-20KHZ.
5. Observe the variation of o/p amplitude with respect to input frequency
6. Plot the graph between gain and frequency.

Observation Table:

Sr no.	Fin	Vin	Vout	Gain=Vout/Vin

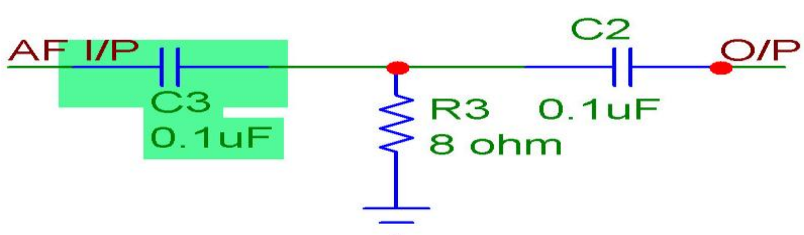
Result: As the I/P frequency increases the O/P amplitude increases in case of pre-emphasis.

EXPRIMENT NO.17

Objective: Generation of De-emphasis circuit on breadboard system & to plot De-emphasis curve.

Apparatus Required: Breadboard system, dual channel CRO, function generator, CRO probes, connecting wires, Resistors of 10 K, 2.7 K. Capacitor of 0.1 µf. Inductor 0.2 mH.

Theory: The noise has greater effects on higher modulating frequencies than on lower ones. Thus, if the higher frequencies were artificially boosted at the transmitter and correspondingly cut at the receiver, an improvement in noise immunity could be expected, thereby increasing the signal-to-noise ratio. Frequency modulation is more immune to noise than amplitude modulation & is significantly more immune than phase modulation. Natural tendency of audio is that the amplitude of high frequency signal is lower as compared to amplitude of low frequency signal. The S/N ratio reduces as the audio frequency increases. In turn the S/N ratio is not same over the entire spectrum of audio signals. Thus in order to reduce the effect of noise the higher frequency signals is boosted before transmission. This is called pre-emphasis. When these signals are recovered at the receiver in order to restore the original amplitudes of higher frequency signals they are suppressed. This is known as **de-emphasis.**



Procedure:-

1. Assemble components and make connections on breadboard as shown in fig.
2. Apply AF input.
3. Observe the waveform on CRO.
4. Vary the AF i/p frequency between 20-20KHZ.
5. Observe the variation of o/p amplitude w.r.t. i/p signal frequency.
6. Plot the graph between gain and frequency.

Sr no.	Fin	Vin	Vout	Gain=Vout/Vin

Result: As the i/p frequency increases the o/p amplitude decreases in case of de-emphasis.