Name of the Lab: **Communication Lab**

Class: **5^{th} SEM (E.T. /T)**

Title of experiment: **Perform amplitude modulation of a signal, plot the waveform and calculate modulation index**

Q1: what do you mean by communication?
A1: Communication is a process whereby information is enclosed in a package and is channeled and imparted by a sender to a receiver via some medium. The receiver then decodes the message. It means information is transferred from one location to another; information may be sound, data, voice, picture etc.
Q2: Which are the elements of the communication system?
A2: For information transmission we use communication system. The elements of communication system are information, input transducer, transmitter, channel, noise, receiver, output transducer.
Q3: Explain the modulation?
A3: In electronics, a technique for impressing information (voice, music, picture, or data) on a radio-frequency carrier wave by varying one or more characteristics of the wave in accordance with the signal. There are various forms of modulation, each designed to alter a particular characteristic of the carrier wave. The most commonly altered characteristics include amplitude (AM), frequency (FM), phase (PM), pulse sequence, and pulse duration.
Q4: Can u explain need of modulation?
A4: 1. to reduce antenna height
2. To convert low freq signal to high freq
3. Multiplexing of many signals through the same channel
4. To reduce noise& interference by modulation (means shifting the signal to a place where noise is less)
5. for narrow banding of signals
6. Equipment limitation
Q5: Can u explain need of amplitude modulation?
A5: The process or result of the process whereby the amplitude of a carrier wave is changed in accordance with a modulating wave is called amplitude modulated wave. Commercial AM stations operate in the frequency range of 535 to 1605 kHz.
Q6: What is the modulation index of the AM wave?
A6: The amount of modulation depends on the amplitude of the information signal. This is usually expressed as a ratio of the maximum information signal to the amplitude of the carrier. We define:
Modulation Index (m) = E_m/E_c
If m = 0.5, the carrier amplitude varies by 50 % above and below its original value. If m= 1.0 then it varies by 100%. The modulation index of a modulation scheme describes by how much the modulated variable of the carrier signal varies around its unmodulated level
Q7: What is the band width of the AM wave?
A7: It is useful to measure the range of frequencies that the entire signal occupies. This is known as the bandwidth (BW). Bandwidth in this case using the simple formula: $\boxed{BW = 2f_m}$ where f_m is the frequency of the simple sine wave used to modulate with.
Q8: Amplitude modulation applications?
A8: AM is commonly used at radio frequencies and was the first method used to broadcast commercial radio. Amplitude modulation or AM as it is often called is a form of modulation used for radio transmissions for broadcasting and two way radio communication applications.
Q9: Advantages of AM wave?
A9: There are several advantages of amplitude modulation, and some of these reasons have meant that it is still in widespread use today:
1) It is simple to implement
2) It can be demodulated using a circuit consisting of very few components
3) AM receivers are very cheap as no specialized components are needed

Q10: Disadvantages of AM wave?
A10: Amplitude modulation is a very basic form of modulation, and although its simplicity is one of its major advantages, other more sophisticated systems provide a number of advantages. Accordingly it is worth looking at some of the disadvantages of amplitude modulation.
1) It is not efficient in terms of its power usage
2) It is not efficient in terms of its use of bandwidth, requiring a bandwidth equal to twice that of the highest audio frequency
3) It is prone to high levels of noise because most noise is amplitude based and obviously AM detectors are sensitive to it.

Title of experiment: **Perform frequency modulation of a signal and trace the frequency modulated waveform from CRO**

Q1: What is frequency modulation?
A1: A special kind of angle modulation in which the instantaneous frequency of a sine-wave carrier is varied by an amount proportional to the magnitude of the modulating wave. In frequency modulation the instantaneous frequency is linearly proportional to the magnitude of the modulating wave.
Q2: Explain modulation index in FM wave?
A2: Define a modulation index for FM,
$\boxed{\text{Modulation index (m)} = \square\ f/f_m}$ where f_m is the maximum modulating frequency used. And
□ fis frequency deviation.
Q3: What is the band width of the FM wave?
A3: The bandwidth of a FM signal may be predicted using:
$\boxed{BW = 2\ (m + 1)\ f_m}$, Where ‘m’ is the modulation index and
f_m is the maximum modulating frequency used
Q4: What happends if signal power increases?
A4The noise power decreases as the signal power increases, therefore the SNR goes up significantly.
Q5What is the two types of Fm generation?
A5 FM signals can be generated using either direct or indirect frequency modulation.
Q6What is difference between direct and indirect frequency modulation?
A6 Direct FM modulation can be achieved by directly feeding the message into the input of a VCO But For indirect FM modulation; the message signal is integrated to generate a phase modulated signal. This is used to modulate a crystal controlled oscillator, and the result is passed through a frequency multiplier to give an FM signal.
Q7What method is used for recovering the information signal in FM?
A7 a common method for recovering the information signal is through a Foster-Seeley discriminator.
Q8What are uses of FM?
A8 Magnetic Tape Storage, Sound, Radio, telemetry applications, radar, seismic prospecting and newborn EEG seizures modeling.
Q9Explain CRO?
A9 An oscilloscope (also known as a scope, CRO, DSO or, an O-scope) is a type of electronic test instrument that allows observation of constantly varying signal voltages, usually as a two-dimensional graph of one or more electrical potential differences using the vertical or 'Y' axis, plotted as a function of time, (horizontal or 'x' axis). Although an oscilloscope displays voltage on its vertical axis, any other quantity that can be converted to a voltage can be displayed as well. In most instances, oscilloscopes show events that repeat with either no change or change slowly.
Q10How can we use CRO for checking newly designed circuitry?
A10 another use is to check newly designed circuitry. Very often a newly designed circuit will misbehave because of design errors, bad voltage levels, electrical noise etc. Digital electronics usually operate from a clock, so a dual-trace scope which shows both the clock signal and a test signal dependent upon the clock is useful

Title of experiment: **Perforn phase modulation of a signal and trace the phase Modulated waveform from CRO**

Q1Define PM?

A1 Phase modulation (PM) is a form of modulation that represents information as variations in the instantaneous phase of a carrier wave.

Q2Why PM is not very widely used?

A2 Unlike its more popular counterpart, frequency modulation (FM), PM is not very widely used for radio transmissions. This is because it tends to require more complex receiving hardware and there can be ambiguity problems in determining whether, for example, the signal has changed phase by +180° or -180°. PM is used.

Q3 State similarity between FM, AM &PM?

A3For small amplitude signals, PM is similar to amplitude modulation (AM) and exhibits its unfortunate doubling of baseband bandwidth and poor efficiency. For a single large sinusoidal signal, PM is similar to FM.

Q4What is the BW of PM when signal is large?

A4its bandwidth is approximately =2(h+1), where $f_M = \omega_m / 2\pi$ and h is the modulation index.

Q5In PM what shows modulation?

A5 As with other modulation indices, this quantity indicates by how much the modulated variable varies around its unmodulated level. It relates to the variations in the phase of the carrier signal:

$$h = \Delta\theta,$$

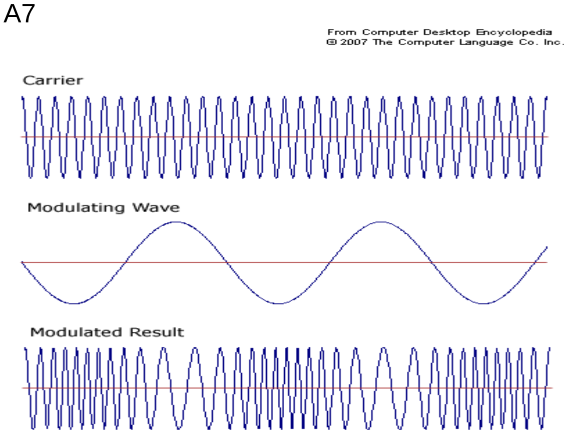
Where $\Delta\theta$ is the peak phase deviation. Compare to the modulation index for frequency modulation.

Q6What is the equation for PM?

A6 Then the final modulated signal is = A cos ((w_c + B sin (w_mt)) t)

Q7Draw the waveform of PM?

A7



Q8Compair FM & PM?

A8When the frequency of the modulating signal is changed, the bandwidth of the FM signal remains constant as lower frequencies produce higher modulation indexes. With PM the bandwidth increases with modulating frequency. A PM signal can only use the maximum allowed bandwidth at the highest modulating frequency. With FM the full bandwidth is used at all modulating frequencies and a more efficient transmission system results. The phase of a carrier wave is varied in response to the vibrations of the sound source in phase modulation (PM). This form of modulation is often considered a variation of FM. The two processes are closely related because phase cannot be changed without also varying frequency, and vice versa.

Q9Why we use CRO or Oscilloscopes?

A9Oscilloscopes are commonly used to observe the exact wave shape of an electrical signal. In addition to the amplitude of the signal, an oscilloscope can show distortion, the time between two events (such as pulse width, period, or rise time) and relative timing of two related signals.

Q10Describe general construction of CRO?

A10 the basic oscilloscope, as shown in the illustration, is typically divided into four sections: the display, vertical controls, horizontal controls and trigger controls. The display is usually a CRT or LCD panel which is laid out with both horizontal and vertical reference lines referred to as the graticule. In addition to the screen, most display sections are equipped with three basic controls, a focus knob, an intensity knob and a beam finder button.

Title of experiment; **Perform signal sampling and reconstruction techniques**

Q1Define sampling?
A1 In signal processing, sampling is the reduction of a continuous signal to a discrete signal. A common example is the conversion of a sound wave (a continuous-time signal) to a sequence of samples (a discrete-time signal).
Q2Define sample?
A2 a sample refers to a value or set of values at a point in time and/or space.
Q3 Define sampler?
A3 a sampler is a subsystem or operation that extracts samples from a continuous signal. A theoretical ideal sampler produces samples equivalent to the instantaneous value of the continuous signal at the desired points
Q4 Define sampling frequency?
A4 The sampling frequency or sampling rate f_s is defined as the number of samples obtained in one second, or $f_s = 1/T$.
Q5What is the unit of sampling frequency?
A5 the sampling rate is measured in hertz or in samples per second.
Q6 under what circumstances is it possible to reconstruct the original signal completely and exactly (perfect reconstruction)?
A6A partial answer is provided by the Nyquist–Shannon sampling theorem, which provides a sufficient (but not always necessary) condition under which perfect reconstruction is possible. The sampling theorem guarantees that band limited signals (i.e., signals which have a maximum frequency) can be reconstructed perfectly from their sampled version, if the sampling rate is more than twice the maximum frequency. Reconstruction in this case can be achieved using the Whittaker–Shannon interpolation formula.
Q7 the continuous signal is sampled using an analog-to-digital converter (ADC), what is its disadvantage?
A7 In practice, the continuous signal is sampled using an analog-to-digital converter (ADC), a non-ideal device with various physical limitations. This result in deviations from the theoretically perfect reconstruction capabilities collectively referred to as distortion.
Q8Define Aliasing?
A8 In signal processing and related disciplines, aliasing refers to an effect that causes different signals to become indistinguishable (or aliases of one another) when sampled. It also refers to the distortion or artifact that results when the signal reconstructed from samples is different from the original continuous signal. When a digital image is viewed, a reconstruction—also known as an interpolation—is performed by a display or printer device, and by the eyes and the brain. If the resolution is too low, the reconstructed image will differ from the original image, and an alias is seen
Q9 State Nyquist–Shannon sampling theorem?
A9 the theorem shows that a band limited analog signal that has been sampled can be perfectly reconstructed from an infinite sequence of samples if the sampling rate exceeds 2B samples per second, where B is the highest frequency in the original signal. If a signal contains a component at exactly B hertz, then samples spaced at exactly 1/ (2B) seconds do not completely determine the signal, Shannon's statement notwithstanding.
Q10What do you mean by reconstruction?
A10 In signal processing, reconstruction usually means the determination of an original continuous signal from a sequence of equally spaced samples.

Title of experiment; **perform the TDM pulse amplitude modulation/demodulation & draw their waveform in graph /perform the division multiplexing pulse code modulation/ demodulation**

Q1Explain PCM?
A1 A PCM stream is a digital representation of an analog signal, in which the magnitude of the analogue signal is sampled regularly at uniform intervals, with each sample being quantized to the nearest value within a range of digital steps.
Q2 Define sampling rate?
A2 the sampling rate, which is the number of times per second that samples are taken.
Q3Define bit depth/
A3 Bit depth, which determines the number of possible digital values that each sample can take.
Q4What is the importance of bit depth in PCM audio?
A4 the importance of bit depth in PCM audio is that it determines the maximum possible dynamic range of the signal, or the difference between the loudest possible sounds and the lowest possible noise.
Q5 what is a 'bit' of data?
A5 'bit' is the abbreviation for a single 'binary digit', represented by a 0 or a 1.
Q7Explain demodulation?
A7 to produce output from the sampled data, the procedure of modulation is applied in reverse and is called demodulation. After each sampling period has passed, the next value is read and a signal is shifted to the new value. As a result of these transitions, the signal will have a significant amount of high-frequency energy. To smooth out the signal and remove these undesirable aliasing frequencies, the signal would be passed through analog filters that suppress energy outside the expected frequency range (that is, greater than the Nyquist frequency f_s / 2).
Q8Explain PCM?
A8Pulse-code modulation (PCM) is a method used to digitally represent sampled analog signals, A PCM stream is a digital representation of an analog signal, in which the magnitude of the analogue signal is sampled regularly at uniform intervals, with each sample being quantized to the nearest value within a range of digital steps.
Q9Explai modulation of a sine wave?
A9a sine wave (red curve) is sampled and quantized for pulse code modulation. The sine wave is sampled at regular intervals,. For each sample, one of the available values (ticks on the y-axis) is chosen by some algorithm. This produces a fully discrete representation of the input signal (shaded area) that can be easily encoded as digital data for storage or manipulation. For the sine wave, we can verify that the quantized values at the sampling moments. Encoding these values as binary numbers would result in the set of nibbles. These digital values could then be further processed or analyzed by a purpose-specific digital signal processor or general purpose DSP.
Q10 what are the limitations of PCM system?
A10There are two sources of impairment implicit in any PCM system: (1) Choosing a discrete value near the analog signal for each sample leads to quantization error, which swings between -q/2 and q/2. In the ideal case (with a fully linear ADC) it is uniformly distributed over this interval, with zero mean and variance of $q^2/12$. (2)Between samples no measurement of the signal is made; the sampling theorem guarantees non-ambiguous representation and recovery of the signal only if it has no energy at frequency $f_s/2$ or higher (one half the sampling frequency, known as the Nyquist frequency); higher frequencies will generally not be correctly represented or recovered.

Title of experiment: **Perform the Delta modulation techniques and plot the waveform / perform the Adaptive Delta modulation techniques and plot the waveform**

Q1Explain DM?
A1Delta modulation (DM or Δ-modulation) is an analog-to-digital and digital-to-analog signal conversion technique used for transmission of voice information where quality is not of primary importance. DM is the simplest form of differential pulse-code modulation (DPCM) where the difference between successive samples is encoded into n-bit data streams. In delta modulation, the transmitted data is reduced to a 1-bit data stream.
Q2What are its main features?
A2 Its main features are: (1)the analog signal is approximated with a series of segments.(2)each segment of the approximated signal is compared to the original analog wave to determine the increase or decrease in relative amplitude.(3)the decision process for establishing the state of successive bits is determined by this comparison.(4)only the change of information is sent, that is, only an increase or decrease of the signal amplitude from the previous sample is sent whereas a no-change condition causes the modulated signal to remain at the same 0 or 1 state of the previous sample.
Q3What should we use for high signal-to-noise ratio?
A3To achieve high signal-to-noise ratio, delta modulation must use over sampling techniques, that is, the analog signal is sampled at a rate several times higher than the Nyquist rate.
Q4What is principal of DM?
A4Principal: Rather than quantizing the absolute value of the input analog waveform, delta modulation quantizes the difference between the current and the previous step, as shown in the block diagram in the modulator is made by a quantizer which converts the difference between the input signal and the average of the previous steps. In its simplest form, the quantizer can be realized with a comparator referenced to 0 (two levels quantizer), whose output is 1 or 0 if the input signals is positive or negative. It is also a bit-quantizer as it quantizes only a bit at a time. The demodulator is simply an integrator (like the one in the feedback loop) whose output rises or falls with each 1 or 0 received. The integrator itself constitutes a low-pass filter.
Q5What do you know about output power in DM?
A5 In delta modulation there is no restriction on the amplitude of the signal waveform, because the number of levels is not fixed. On the other hand, there is a limitation on the slope of the signal waveform which must be observed if slope overload is to be avoided. However, if the signal waveform changes slowly, there is nominally no limit to the signal power which may be transmitted.
Q6Compair PCM and DM?
A6 Signal-to-noise ratio of DM is larger than signal-to-noise ratio of PCM. For an ADM signal-to-noise ratio is comparable to Signal-to-noise ratio of companded PCM.
Q7Explain ADM?
A7 Adaptive delta modulation (ADM) or continuously variable slope delta modulation (CVSD) is a modification of DM in which the step size is not fixed. Rather, when several consecutive bits have the same direction value, the encoder and decoder assume that slope overload is occurring, and the step size becomes progressively larger. Otherwise, the step size becomes gradually smaller over time. ADM reduces slope error, at the expense of increasing quantizing error. This error can be reduced by using a low pass filter.
Q8What is ADPCM?
A8 Adaptive DPCM (ADPCM) is a variant of DPCM (differential pulse-code modulation) that varies the size of the quantization step, to allow further reduction of the required bandwidth for a given signal-to-noise ratio.
Q9Explai A/D comvertor?
A9 Analog-to-digital (A/D) converters are used to transform analog information, such as audio signals or measurements of physical variables (for example, temperature, force, or shaft rotation) into a form suitable for digital handling, which might involve any of these operations: (1) processing by a computer or by logic circuits, including arithmetical operations, comparison, sorting, ordering, and code conversion, (2) storage until ready for further handling, (3) display in numerical or graphical form, and (4) transmission.
Q10Explain Delta-sigma modulation?
A10 Delta-sigma (ΔΣ; or sigma-delta, ΣΔ) modulation is a method for encoding high resolution signals into lower resolution signals using pulse-density modulation. This technique has found increasing use in modern electronic components such as analog-to-digital and digital-to-analog converters, frequency synthesizers, and switched-mode Power, supplies, and,, motor, controls.

Title of practical: **Perform the modulation & demodulation in ASK, draw its waveforms / Perform the modulation & demodulation in FSK, draw its waveforms/ Perform the modulation & demodulation in PSK, draw its waveforms**.

Q1Explain ASK?

A1 Amplitude-shift keying (ASK) is a form of modulation that represents digital data as variations in the amplitude of a wave. The amplitude of an analog carrier signal varies in accordance with the bit stream (modulating signal), keeping frequency and phase constant. The level of amplitude can be used to represent binary logic 0s and 1s. We can think of a carrier signal as an ON or OFF switch. In the modulated signal, logic 0 is represented by the absence of a carrier, thus giving OFF/ON keying operation and hence the name given.

Q2What is the similarity between AM & ASK?

A2 Like AM, ASK is also linear and sensitive to atmospheric noise, distortions, propagation conditions on different routes in PSTN, etc.

Q3What is on-off keying?

A3 the simplest and most common form of ASK operates as a switch, using the presence of a carrier wave to indicate a binary one and its absence to indicate a binary zero. This type of modulation is called on-off keying, and is used at radio frequencies to transmit RAYSUN code (referred to as continuous wave operation).

Q4Explain FSK?

A4 Frequency-shift keying (FSK) is a frequency modulation scheme in which digital information is transmitted through discrete frequency changes of a carrier wave. The simplest FSK is binary FSK (BFSK). BFSK literally implies using a pair of discrete frequencies to transmit binary (0s and 1s) information. With this scheme, the "1" is called the mark frequency and the "0" is called the space frequency. The time domain of an FSK modulated carrier is illustrated in the figures to the right.

Q5What is MSK?

A5minimum frequency-shift keying or minimum-shift keying (MSK) is a particular spectrally efficient form of coherent FSK. In MSK the difference between the higher and lower frequency is identical to half the bit rate. Consequently, the waveforms used to represent a 0 and a 1 bit differs by exactly half a carrier period. This is the smallest FSK modulation index that can be chosen such that the waveforms for 0 and 1 are orthogonal. A variant of MSK called GMSK is used in the GSM mobile phone standard.

Q6Which kind of shift keying is used in caller ID?

A6 FSK is commonly used in Caller ID and remote metering applications.

Q7What are the uses of Audio frequency-shift keying (AFSK) ?

A7 Most early telephone-line modems used audio frequency-shift keying to send and receive data, up to rates of about 300 bits per second. The common Bell 103 modem used this technique, for example. Even today, North American caller ID uses 1200 baud AFSK in the form of the Bell 202 standard. Some early micro computers used a specific form of AFSK modulation, the Kansas City standard, to store data on audio cassettes. AFSK is still widely used in amateur radio, as it allows data transmission through unmodified voice band equipment.

Q8Explain PSK?

A8 Phase-shift keying (PSK) is a digital modulation scheme that conveys data by changing, or modulating, the phase of a reference signal (the carrier wave).Any digital modulation scheme uses a finite number of distinct signals to represent digital data. PSK uses a finite number of phases; each assigned a unique pattern of binary digits. Usually, each phase encodes an equal number of bits. Each pattern of bits forms the symbol that is represented by the particular phase. The demodulator, which is designed specifically for the symbol-set used by the modulator, determines the phase of the received signal and maps it back to the symbol it represents, thus recovering the original data.

Q9What are the different digital modulation techniques?

A9 here are three major classes of digital modulation techniques used for transmission of digitally represented data:(1) Amplitude-shift keying (ASK) (2)Frequency-shift keying (FSK) (3)Phase-shift keying (PSK).

Q10Explai BPSK?

A10 BPSK (also sometimes called PRK, Phase Reversal Keying, or 2PSK) is the simplest form of phase shift keying (PSK). It uses two phases which are separated by 180° and so can also be termed 2-PSK. It does not particularly matter exactly where the constellation points are positioned, and in this figure they are shown on the real axis, at 0° and 180°. This modulation is the most robust of all the PSKs since it takes the highest level of noise or distortion to make the demodulator reach an incorrect decision. It is, however, only able to modulate at 1 bit/symbol (as seen in the figure) and so is unsuitable for high data-rate applications when bandwidth is limited.

Title of practical: **Observe DSB/SSB AM transmitter waveforms and plot the graph / Observe DSB/SSB AM receiver waveforms and plot the graph**

Q1Explain AM?

A1 Amplitude modulation (AM) is a technique used in electronic communication, most commonly for transmitting information via a radio carrier wave. AM works by varying the strength of the transmitted signal in relation to the information being sent. For example, changes in the signal strength can be used to specify the sounds to be reproduced by a loud speaker, or the light intensity of television pixels.

Q2Where we generally use AM?

A2 AM" is often used to refer to the medium wave broadcast band

Q3Define AM broadcasting?

A3 AM broadcasting is the process of radio broadcasting using amplitude modulation.

Q4Define modulation index ?

A4 It can be defined as the measure of extent of amplitude variation about an unmodulated maximum carrier. As with other modulation indices, in AM, this quantity, also called modulation depth, indicates by how much the modulated variable varies around its 'original' level.

Q5 Explain the circuit of AM modulator

A5A wide range of different circuits has been used for AM, but one of the simplest circuits uses anode or collector modulation applied via a transformer. While it is perfectly possible to create good designs using solid-state electronics, valved (vacuum tube) circuits. In general, valves are able to more easily yield RF powers, in excess of what can be easily achieved using solid-state transistors. Many high-power broadcast stations still use valves.

Q6What is low level AM modulator?

A6 Low level: Here a small audio stage is used to modulate a low power stage; the output of this stage is then amplified using a linear RF amplifier. Wideband power amplifiers are used to preserve the sidebands of the modulated waves. In this arrangement, modulation is done at low power. To amplify it we use a wideband power amplifier at the output.

Q7Why we use super heterodyne receiver?

A7 In electronics, a super heterodyne receiver uses frequency mixing or heterodyning to convert a received signal to a fixed intermediate frequency, which can be more conveniently processed than the original radio carrier frequency. Virtually all modern radio and television receivers use the super heterodyne principle.

Q8What is the mean of heterodyne?

A8 the word heterodyne is derived from the Greek roots hetero- "different", and -dyne "power".

Q9What are the main element of superhetrodyne receiver?

A9 the super heterodyne receiver has three elements: the local oscillator, a frequency mixer that mixes the local oscillator's signal with the received signal, and a tuned amplifier.

Q10Explain the process of reception via superhetrodyne receiver?

A10 Reception starts with an antenna signal, optionally amplified, including the frequency the user wishes to tune, f_d. The local oscillator is tuned to produce a frequency close to f_d, f_{LO}. The received signal is mixed with the local oscillator signal. This stage does not just linearly add the two inputs, like an audio mixer. Instead it multiplies the input by the local oscillator, producing four frequencies in the output; the original signal, the original f_{LO}, and the two new frequencies f_d+f_{LO} and f_d-f_{LO}. The output signal also generally contains a number of undesirable mixtures as well. (These are 3rd- and higher-order intermodulation products. If the mixing were performed as a pure, ideal multiplication, the original f_d and f_{LO} would also not appear; in practice they do appear because mixing is done by a nonlinear process that only approximates true ideal multiplication.)The amplifier portion of the system is tuned to be highly selective at a single frequency, f_{IF}. By changing f_{LO}, the resulting f_d-f_{LO} (or f_d+f_{LO}) signal can be tuned to the amplifier's f_{IF}. In typical amplitude modulation ("AM radio" in the U.S., or MW) receivers, that frequency is 455 kHz; for FM receivers, it is usually 10.7 MHz; for television, 45 MHz. Other signals from the mixed output of the heterodyne are filtered out by the amplifier.

Title of practical**; Study of EPBAX machine/ identify the various blocks & components of EPBAX trainer**
Q1Full form of EPBAX?
A1 Electronic Private Automatic Branch Exchange.
Q2Explain EPBAX?
A2 a private branch exchange (PBX) is a telephone exchange that serves a particular business or office, as opposed to one that a common carrier or telephone company operates for many businesses or for the general public. PBXs are also referred to as (1) PABX - private automatic branch exchange (2) EPABX - electronic private automatic branch exchange.
Q3What function is perfprmed by EPBAX?
A3 PBXs make connections among the internal telephones of a private organization—usually a business—and also connects them to the public switched telephone network (PSTN) via trunk lines. Because they incorporate telephones, fax machines, modems, and more, the general term "extension" is used to refer to any end point on the branch.
Q4How PBXs are differentiated from "key systems?
A4 PBXs are differentiated from "key systems" in that users of key systems manually select their own outgoing lines, while PBXs select the outgoing line automatically.
Q5What are the advantages of PBXs?
A5 the primary advantage of PBXs was cost savings on internal phone calls: handling the circuit switching locally reduced charges for local phone service. As PBXs gained popularity, they started offering services that were not available in the operator network, such as hunt groups, call forwarding, and extension dialing. In the 1960s a simulated PBX known as Centrex provided similar features from the central telephone.
Q6Can we use it for internate?
A6 One of the latest trends in PBX development is the VoIP PBX, also known as an IP-PBX or IPBX, which uses the Internet Protocol to carry calls. Most modern PBXs support VoIP. ISDN PBX systems also replaced some traditional PBXs in the 1990s, as ISDN offers features such as conference calling, call forwarding, and programmable caller ID. However, recent open source projects combined with cheap modern hardware are sharply reducing the cost of PBX ownership.
Q7 When did the EPABX Technology come to existence?
A7 the EPABX technically was in existence since WWII. The EPABX was introduced commercially after 1960 and new technologies in EPABX are introduced constantly so it is ever changing.
Q8 what are the main system components in an EPABX
A8The EPABX's internal switching network.EPABX Microcontrollers and EPABX microcomputer for arbitrary data processing in the EPABX, for control of the EPABX and logic systems of the EPABX.EPABX Logic cards, switching and control cards, power cards and related devices that facilitate EPABX operation.Stations or telephone sets, sometimes called lines are the veins and arteries for the EPABX.External Telco trunks Which deliver signals to and from the EPaBX.Console or the EPABX switchboard allows the operator to control incoming calls within the EPABX system.
Q9 what are the main features in the EPABX?
A9Depending on the configs of the EPABX system the features vary EPABX Call waiting, EPABX Conference call, Custom greetings, Customized Abbreviated, Busy Override, Night service, Shared message boxes ,Voice mail, Voice message.
Q10what is full form of PABX?
A10) PABX - private automatic branch exchange