UNIT-I

Introduction:

The measurement of any quantity plays very important role not only in science but in all branches of engineering, medicine and in almost all the human day to day activities.

The technology of measurement is the base of advancement of science. The role of science and engineering is to discover the new phenomena, new relationships, the laws of nature and to apply these discoveries to human as well as other scientific needs. The science and engineering is also responsible for the design of new equipments. The operation, control and the maintenance of such equipments and the processes is also one of the important functions of the science and engineering branches. All these activities are based on the proper measurement and recording of physical, chemical, mechanical, optical and many other types of parameters.

The measurement of a given parameter or quantity is the act or result of a quantitative comparison between a predefined standard and an unknown quantity to be measured. The major problem with any measuring instrument is the error. Hence, it is necessary to select the appropriate measuring instrument and measurement procedure which minimises the error. The measuring ~~instrument should not affect the quantity to be measured.~~

An electronic instrument is the one which is based on electronic or electrical principles for its measurement function. The measurement of any electronic or electrical quantity or variable is termed as an electronic measurement.

Advantages of Electronic Measurement

The advantages of an electronic measurement are

1. Most of the quantities can be converted by transducers into the electrical or electronic signals.
2. An electrical or electronic signal can be amplified, filtered, multiplexed, sampled and measured.
3. The measurement can easily be obtained in or converted into digital form for automatic analysis and recording.

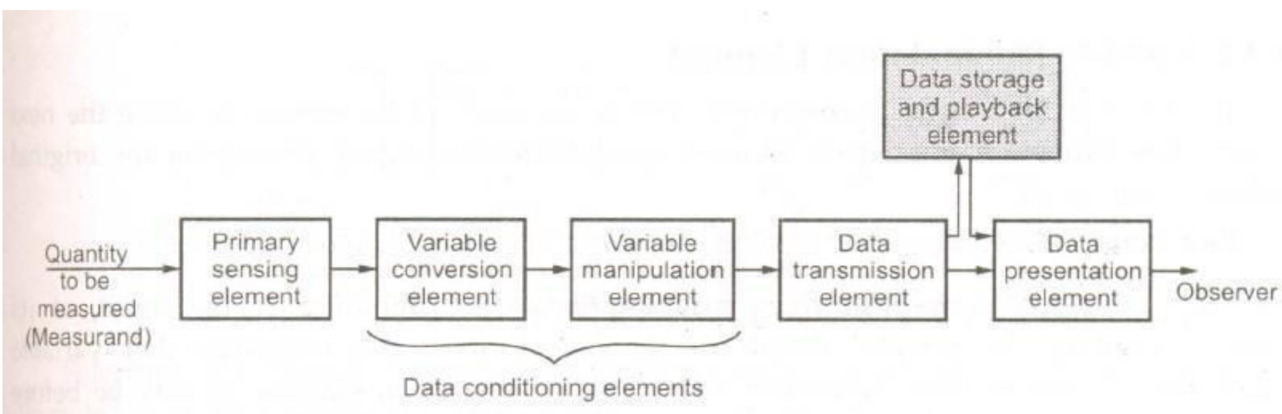
4 The measured signals can be transmitted over long distances with the help of cables or radio links, without any loss of information.

5. Many measurements can be carried either simultaneously or in rapid succession.
6. Electronic circuits can detect and amplify very weak signals and can measure the events of very short duration as well.

7. Electronic measurement makes possible to build analog and digital signals. The digital signals are very much required in computers. The modern development in science and technology are totally based on computers.
8. Higher sensitivity, low power consumption and a higher degree of reliability are the important features of electronic instruments and measurements. But, for any measurement, a well defined set of standards and calibration units is essential. This chapter provides an introduction to different types of errors in measurement, the characteristics of an instrument and different calibration standards.

Functional elements of an instruments:

Any instrument or a measuring system can be described in general with the help of a block ~~diagram. While describing the general form of a measuring system, it is not necessary to go into the~~ details of the physical aspects of a specific instrument. The block diagram indicates the necessaryelements and their functions in a general measuring system. The entire operation of an instrument can be studied interms of these functional elements. The Fig. 1.1 shows the block diagram showing the functional elements of an instrument.



Calibration:

Calibration is the process of making an adjustment or marking a scale so that the readings of an instrument agree with the accepted and the certified standard.

The calibration offers a guarantee to the device or instrument that it is operating with required accuracy, under the stipulated environmental conditions. It creates the confidence of using the properly calibrated instrument, in user's mind. The periodic calibration of an instrument is very much necessary.

The calibration characteristics can be determined by applying known values of quantities to be measured and recording the corresponding output of the instrument. Such output values are then compared with the input, to determine the error. Such a record obtained from calibration is called calibration record. It is generally recorded in the tabular form. If it is represented in the graphical form, it is called calibration curve. Such a calibration record or calibration curve is useful to obtain the performance characteristics of an instrument. The performance of the instrument is not guaranteed by the calibration. It only mdicates whether the performance of the instrument is meeting the accuracy and range specification or not. If the device has been repaired, aged, adjusted or modified, then recalibration is carried out.

Static characteristics:

As mentioned earlier, the static characteristics are defined for the instruments which measure the quantities which do not vary with time. The various static characteristics are accuracy, precision, resolution, error, sensitivity, threshold, reproducibility, zero drift, stability and linearity.

Accuracy:

It is the degree of closeness with which the instrument reading approaches the true value of the quantity to be measured. It denotes the extent to which we approach the actual value of the quantity. It indicates the ability of instrument to indicate the true value of the quantity. The accuracy can be expressed in the following ways.

1) Accuracy as 'Percentage of Full Scale Reading : In case of instruments having uniform scale, the accuracy can be expressed as percentage of full scale reading.

For example, the accuracy of an instrument having full scale reading of 50 units may be expressed as ± 0.1% of full scale reading. From this accuracy indication, practically accuracy is expressed in terms of limits of error. So for the accuracy limits specified above, there will be ± 0.05 units error in any measurement. So for a reading of 50 units, there will be error of ± 0.05 units i.e. ± 0.1 % while for a reading of 25 units, there will be error of ± 0.05 units in the reading i.e. ± 0.2%. Thus as reading decreases, error in measurement is ± 0.05 units but net percentage error is more. Hence, specification of accuracy in this manner is highly misleading.

2) Accuracy as 'Percentage of True Value' : This is the best method of specifying the accuracy. It is to be specified in terms of the true value of quantity being measured. For example, it can be specified as ± 0.1% of true value. This indicates that in such cases, as readings get smaller, error also gets reduced. Hence accuracy of the instrument is better than the instrument for which it is specified as percent of full scale reading.

Precision:

It is the measure of consistency or repeatability of measurements.

Let us see the basic difference between accuracy and precision. Consider an instrument on which, readings upto 1/1000th of unit can be measured. But the instrument has large zero adjustment error. Now every time reading is taken, it can be taken down upto '1000th of unit. So as the readings agree with each other, we say that the instrument is highly precise. But, though the readings are precise upto 10100th of unit, the readings are inaccurate due to large zero adjustment error. Every reading will be inaccurate, due to such error. Thus a precise instrument may not be accurate. Thus the precision means sharply or clearly defined and the readings agree among themselves. But there is no guarantee that readings are accurate. An instrument having zero error, if calibrated properly, can give accurate readings but in that case still, the readings can be obtained down upto 1/10th of unit only. Thus accuracy can be improved by calibration but not the precision of the instrument.

The precision is composed of two characteristics:

- Conformity and
- Number of significant figures.

Conformity:

Consider a resistor having true value as 2385692.0Ω, which is being measured by an ohmmeter. Now, the meter is consistently measuring the true value of the resistor. But the reader, can read consistently, a value as 2.4 MΩ due to nonavailability of proper scale. The value 2.4 MΩ is estimated by the reader from the available scale. There are no deviations from the observed value. The error created due to the limitation of the scale reading is a precision error.

~~The example illustrates that the conformity is a necessary, but not sufficient condition for~~ precision. Similarly, precision is necessary but not the sufficient condition for accuracy.

Significant Figures:

The precision of the measurement is obtained from the number of significant figures, in which the reading is expressed. The significant figures convey the actual information about the magnitude and the measurement precision of the quantity.

The **precision** can be mathematically expressed as :

$$\boxed{P = 1 - \left|\frac{X_n - \overline{X}_n}{\overline{X}_n}\right|}$$

where P = Precision

X_n = Value of n^{th} measurement

$\overline{X}_n$ = Average of the set of measured values

Example:

The table shows the set of 5 *measurements.Calculate the precision of the 3rd measurement*

Measurement Number	Value of Measurement
1	49
2	51
3	52
4	50
5	49

Resolution:

It is the smallest increment of quantity being measured which can be detected with certainty by an instrument.

So if a nonzero input quantity is slowly increased, output reading will not increase until some minimum change in the input takes place. This minimum change which causes the change in the output is called resolution. The resolution of an instrument is also referred to as discrimination of the instrument. The resolution can affect the accuracy of the measurement.

Errors:

Static error = measured value – true value

The most important static characteristics of an instrument is its accuracy, which is generally expressed in terms of the error called static error.

Mathematically it can be expressed as, $e = At - Am$ J

where e = Error

A_m = Measured value of the quantity

A_t = True value of the quantity

In this expression, the error denoted as e is also called absolute error. The absolute error does not indicate precisely the accuracy of the measurements. For example, absolute error of ± 1 V is negligible when the voltage to be measured is of the order of 1000 V but the same error of ±1 V becomes significant when the voltage under measurement is 5 V or so. Hence, generally instead of specifying absolute error, the relative or percentage error is specified.

Sensitivity:

If the calibration curve is not linear as shown in the Fig. 1.3 (b), then the sensitivity varies with the input. The sensitivity is always expressed by the manufacturers as the ratio of the magnitude of quantity being measured to the magnitude of the response. Actually, this definition is the reciprocal of the sensitivity is called inverse sensitivity or deflection factor. But manufacturers call this inverse sensitivity as a sensitivity.

$$\text{Inverse sensitivity} = \text{Deflection factor}$$

$$\therefore \quad \boxed{\text{Deflection factor} = \frac{1}{\text{Sensitivity}} = \frac{\Delta q_i}{\Delta q_o}}$$

The units of the sensitivity are millimeter per micro-ampere, millimeter per ohm, counts per volt, etc. while the units of a deflection factor are micro-ampere per millimeter, ohm per millimeter, volts per count, etc. The sensitivity of the instrument should be as high as possible and to achieve this range of an instrument should not greatly exceed the value to be measured.

Drift : Gradual shift in the meassured value ,over an extended period, when there is no change in input.

Threshold: The minimum value of input for which the device just starts to respond.

Range/Span: The minimum and maximum value of quantity so that the device is capable of measuring.

Repeatability: A measure of how well the output returns to a given value when the same precise input is applied several times. Or The ability of an instrument to reproduce a certain set of reading within a given accuracy.

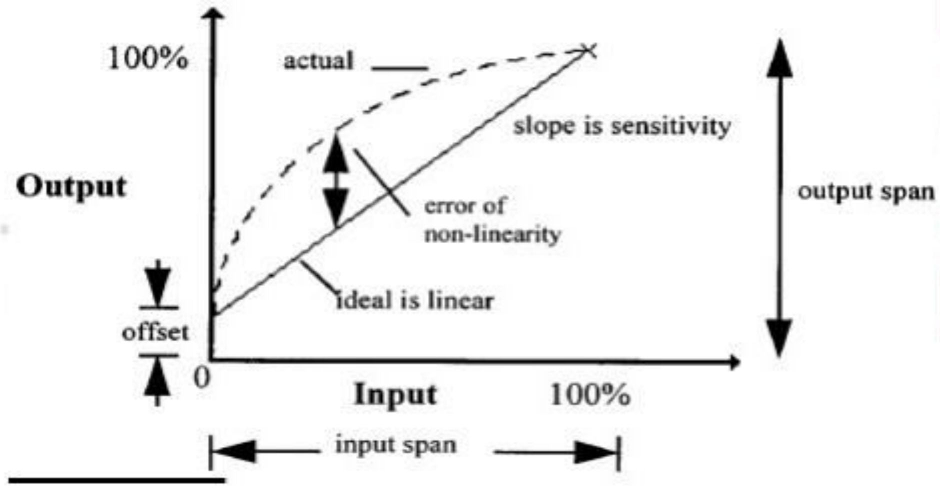
Linearity: Input output relationship of a device must be linear.

But practical systems shows small deviations from the linear shape (allowed within the specified limits)

Hysteresis: Input is increased from negative value, output increases as indicated by curve 1

• Then the input is steadily decreased , output does not follow the same path , but lag by a certain value as indicated by curve 2 •

The difference between the two curves is called Hysterisis.



DYNAMIC CHARACTERISTICS:

The response of instruments or systems to dynamic I/P s are also functions of time.

Instruments rarely respond instantaneously to changes in the measured variables.

Instead, they exhibit slowness or sluggishness due to such things as mass, thermal capacitance, fluid capacitance or electric capacitance.

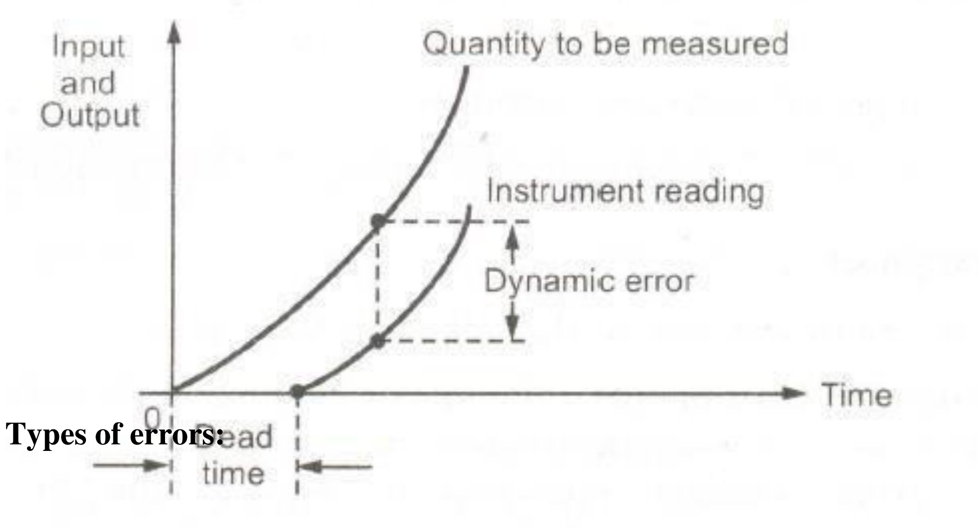
• **Speed of Response**: It is the ability of a system to respond to a sudden changes in the input signal/quantity

• **Fidelity**: It is the degree to which an instrument indicates the changes in the measured variable without dynamic error (Indication of how much faithfully system responds to the changes in input).

Lag: It is the retardation or delay in the response of an instrument to changes in the measured variable. Two types : Process lag(process) and Control lag (Instrument)

Dynamic error:

It is the difference between the true value of the variable to be measured, changing with time and the value indicated by the measurement system, assuming zero static error. The Fig. 1.13 shows the dead time, i.e. time delay and the dynamic error.



Types of errors:

The static error is defined earlier as the difference between the true value of the variable and the value indicated by the instrument. The static error may arise due to number of reasons. The static errors are classified as:

1) Gross errors

2) Systematic errors

3) Random errors

Gross errors:

The gross errors mainly occur due to carelessness or lack of experience of a human being. These cover human mistakes in readings, recordings and calculating results. These errors also occur due to incorrect adjustments of instruments. These errors cannot be treated mathematically. These errors are also called personal errors. Some gross errors are easily detected while others are very difficult to detect.\

Systematic errors:

The systematic errors are mainly resulting due to the shortcomings of the instrument and the characteristics of the material used in the instrument, such as defective or worn parts, ageing effects, environmental effects, etc.

A constant uniform deviation of the operation of an instrument is known as a systematic error. There are three types of systematic errors as

1) Instrumental errors 2) Environmental errors 3) Observational errors

Instrumental errors :

These errors are mainly due to following three reasons

• Short-comings of instrument

These are because of the mechanical structure of the instruments eg. Friction in the bearings of various moving parts, irregular spring tensions, hysteresis, gear backlash, variation in air gap etc.

Misuse of instrument A good instrument if used in abnormal way gives misleading results. Poor initial adjustments, Improper zero setting, Using leads of high resistance. Elimination: Use the instrument intelligently & Correctly

• Loading effects Loading effects due to Improper way of using the instrument

• **Ellimination.**

– Selecting proper instrument and the transducer for the measurement.

– Recognize the effect of such errors and apply the proper correction factors.

– Calibrate the instrument carefully against standard.

Environmental Errors (due to the External Conditions)

• The various factors : Temperature changes, Pressure, vibratons, Thermal emf., stray capacitance, cross capacitance, effect of External fields, Aging of equipments and Frequency sensitivity of an instrument.

Elimination • Using proper correction factors and using the instrument Catalogue • Using Temperature & Pressure control methods etc. • Reducing the effect of dust, humidity on the components in the instruments. • The effects of external fields can be minimized by using the magnetic or electrostatic shields of screens.

Observational Errors:

Error introduced by the observer

Few souces are:

• Parallax error while reading the meter,

• wrong scale selection,

• habits of individual obsever

• Elimination

Use the

• instrument with mirrors,

• instrument with knife edge pointers,

• Instrument having digital display

Random errors:

Some errors still result, though the systematic and instrumental errors are reduced or atleast accounted for. The causes of such errors are unknown and hence, the errors are called **random** errors. These errors cannot be determined in the ordinary process of taking the measurements.

Absolute and relative errors:

When the error is specified interms of an absolute quantity and not as a percentage, then it is called an absolute error.

Thus the voltage of 10 ± 0.5 V indicated ± 0.5 V as an absolute error. When the error is expressed as a percentage or as a fraction of the total quantity to be measured, then it is called relative error. Generally the relative error in case of resistances is specified as percentage of tolerances. Another method of expressing error is by specifying it as parts per million (ppm), relative to the total quantity. So it is a relative error specification . Generally change in resistance with temperature is indicated in ppm/ °C shows the variation in resistance with Temperature temperature. Thus if a resistance of 100 kΩ. has a temperature coefficient of 50 *ppm/C* means 50 parts per millionth per degree celcius. Thus one millionth of 100 kohm. is 0.1 ohm and 50 such parts means 5 D.

Limiting errors:

The manufacturers specify the accuracy of the instruments within a certain percentage of full scale reading. The components like the resistor, inductor, capacitor are guaranteed to be within a certain percentage of rated value. This percentage indicates the deviations from the nominal or specified value of the particular quantity. These deviations from the specified value are called **Limiting Errors.** These are also called **Guarantee Errors.**

Thus the actual value with the limiting error can be expressed mathematically as,

$$\boxed{A_a = A_s \pm \delta A}$$

where A_a = Actual value

A_s = Specified or rated value

δA = Limiting error or tolerance

Relative limiting error:

This is also called fractional error. It is the ratio of the error to the specified magnitude of a quantity.

Thus $$\boxed{e = \frac{\delta A}{A_s}}$$

where e = Relative timing error

From the above equation, we can write,

$$\delta A = e \cdot A_s$$

and $$A_a = A_s \pm \delta A$$

$$= A_s \pm e\, A_s$$

$$\therefore \quad \boxed{A_a = A_s [1 \pm e]}$$

The percentage relative limiting error is expressed as

$$\boxed{\%\, e = e \times 100}$$

The relative limiting error can be also be expressed as,

$$\boxed{e = \frac{\text{Actual value } (A_a) - \text{Specified value } (A_s)}{\text{Specified value } (A_s)}}$$

Voltmeters and multimeters:

Basic meter:

A basic d.c. meter uses a motoring principle for its operation. It stntes that any current carrying coil placed in a magnetic field experiences a force, which is proportional to the magnitude of current passing through the coil. This movement of coil is called D'Arsonval movement and basic meter is called D'Arsonval galvanometer.

D.C instruments:

a) Using shunt resistance, d.c. current can be measured. The instrument is d.c. microammeter, milliammeter or ammeter.

b) Using series resistance called multiplier, d.c. voltage can be measured. The instrument is d.c. millivoltmeter, voltmeter or kilovoltmeter.

c) Using a battery and resistive network, resistance can be measured. The instrument is ohmmeter.

A.C instruments:

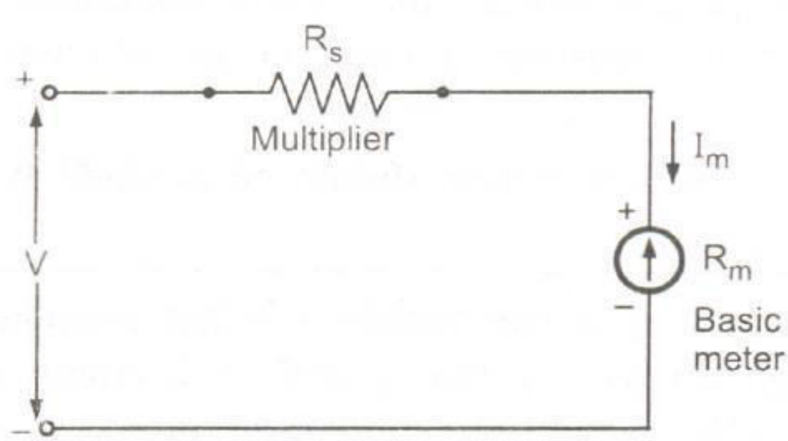
a) Using a rectifier, a.c. voltages can be measured, at power and audio frequencies. The instrument is a.c. voltmeter.

b) Using a thermocouple type meter radio frequency (RF) voltage or current can be measured.

c) Using a thermistor in a resistive bridge network, expanded scale for power line voltage can be obtained.

Basic DC voltmeter:

The basic d.c. voltmeter is nothing but a permanent magnet moving coil (PMMC) 0' Arsonval galvanometer. The resistance is required to be connected in series with the basic meter to use it as a voltmeter. This series resistance is called a **multiplier.** The main function of the multiplier is to limit the current through the basic meter so that the meter current does not exceed the full scale deflection value. The voltmeter measures the voltage across the two points of a circuit or a voltage across a circuit component. The basic d.c. voltmeter is shown in the Fig.



The voltmeter must be connected across the two points or a component, to measure the potential difference, with the proper polarity.

The multiplier resistance can be calculated as:

Let R_m = internal resistance of coil i.e. meter

R_s = series multiplier resistance

I_m = full scale deflection current

V = full range voltage to be measured

From Fig. 2.1, $\therefore\ V = I_m(R_m + R_s)$

$$\therefore \quad V = I_m R_m + I_m R_s$$

$$\therefore \quad I_m R_s = V - I_m R_m$$

$$\therefore \quad \boxed{R_s = \frac{V}{I_m} - R_m}$$

The multiplying factor for multiplier is the ratio of full range voltage to be measured and the drop across the basic meter.

Let v = drop across the basic meter $= I_m R_m$

$\therefore\ m$ = multiplying factor $= \dfrac{V}{v}$

$$= \frac{I_m (R_m + R_s)}{I_m R_m}$$

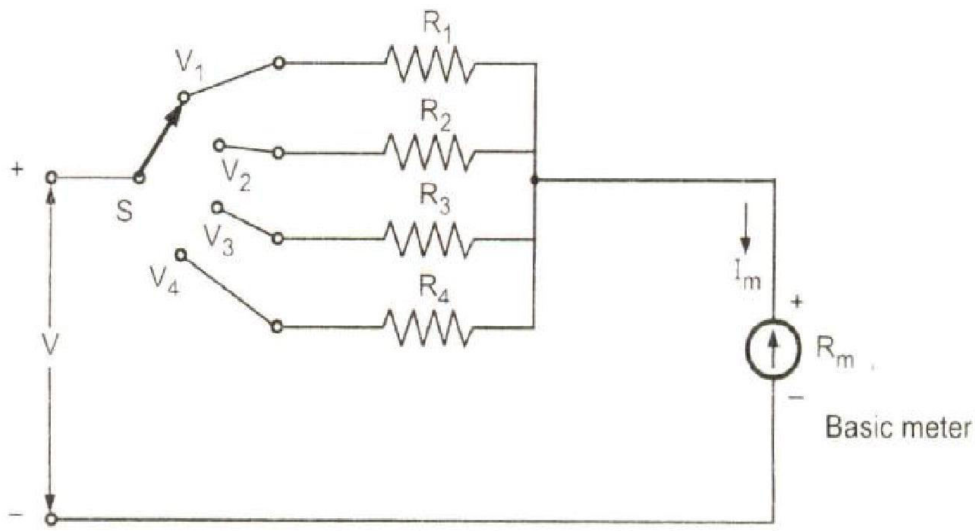
$$\boxed{m = 1 + \frac{R_s}{R_m}}$$

Hence multiplier resistance can also be expressed as,

$$R_s = (m-1) R_m$$

Multirange voltmeters:

The range of the basic d.c. voltmeter can be extended by using number of multipliers clnd a selector switch. Such a meter is called **multirange** voltmeter.



Multirange voltmeter

The R1, R2, R3 and R4 are the four series multipliers. When connected in series with the meter, they can give four different voltage ranges as V1, V2, V3, and V4. The selector switch S is multiposition switch by which the required multiplier can be selected in the circuit.

The mathematical analysis of basic d.c.*voltmeter* is equally applicable for such multirange *voltmeter.* Thus,

$$R_1 = \frac{V_1}{I_m} - R_m \qquad R_2 = \frac{V_2}{I_m} - R_m \qquad \text{and so on.}$$

Sensitivity of voltmeters:

In a multirange voltmeter, the ratio of the total resistance Rt to the voltage range remains same. This ratio is nothing but the reciprocal of the full scale deflection current of the meter i.e. 1/101. This value is called sensitivity of the voltmeter. Thus the sensitivity of the voltmeter is defined,

$$S = \frac{1}{\text{Full scale deflection current}}$$

$$\boxed{S = \frac{1}{I_m}\ \Omega/V \text{ or } k\Omega/V}$$

Loading effect:

While selecting a meter for a particular measurement, the sensitivity rating IS very important. A low sensitive meter may give the accurate reading in low resistance circuit but will produce totally inaccurate reading in high resistance circuit.

The voltmeter is always connected across the two points between which the potential difference is to be measured. If it is connected across a low resistance then as voltmeter resistance is high, most of the current will pass through a low resistance and will produce the voltage drop which will be nothing but the true reading. But if the voltmeter is connected across the high resistance then due to two high resistances in parallel, the current will divide almost equally through the two paths. Thus the meter will record the voltage drop across the high resistance which will be much lower than the true reading. Thus the low sensitivity instrument when used in high resistance circuit 'gives a lower than the true reading. This is called loading effect of the voltmeters. It is mainly caused due to low sensitivity instruments.

A.C voltmeters using rectifier:

~~The PMMC movement used in d.c. voltmeters can be effectively used in a.c. voltmeters. The~~ rectifier is used to convert a.c. voltage to be measured, to d.c. This d.c., if required is amplified and then given to the PMMC movement. The PMMC movement gives the deflection proportional to the quantity to be measured.

The r.m.s. value of an alternating quantity is given by that steady current (d.c.) which when flowing through a given circuit for a given time produces the same amount of heat as produced by the alternating current which when flowing through the same circuit for the same time. The r.m.s value is calculated by measuring the quantity at equal intervals for one complete cycle. Then squaring each quantity, the average of squared v,llues is obtained. The square root of this average value is the r.m.s. value. The r.m.s means root-mean square i.e. squaring, finding the mean i.e. average and finally root.

If the waveform is continuous then instead of squaring and calculating mean, the integratioll is used. Mathematically the r.m.s. value of the continuous a.c. voltage having time period T is given by,

$$V_{rms} = \sqrt{\frac{1}{T}\int_0^T V_{in}^2 \, dt}$$

The $\frac{1}{T}$ term indicates the mean value or average value.

For **purely sinusoidal quantity,**

$$V_{rms} = 0.707 \; V_m$$

where V_m = peak value of the sinusoidal quantity

If the a.c. quantity is continuous then average value can be expressed mathematically using integration as,

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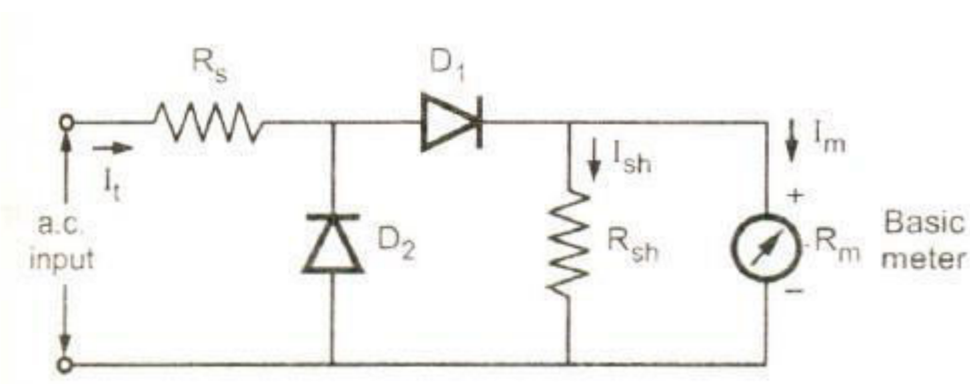
The form factor is the ratio of r.m.s. value to the average value of an alternating quantity.

$$K_f = \frac{\text{r.m.s. value}}{\text{average value}} = \text{form factor}$$

Basic rectifier type voltmeter:

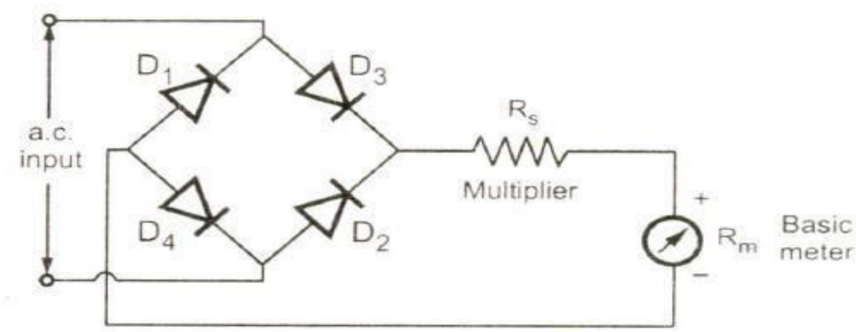


The diodes D1 and D2 are used for the rectifier circuit. The diodes show the nonlinear behaviour for the low currents hence to increase the current through diode D1, the meter is shunted with a resistance Rsh' This ensures high current through diode and its linear behaviour.

When the a.c. input is applied, for the positive half cycle, the diode 01 conducts and causes the meter deflection proportional to the average value of that half cycle. In the negative cycle, the diode D2 conducts and D1 is reverse biased. The current through the meter is in opposite direction and hence meter movement is bypassed. Thus due to diodes, the rectifying action produces pulsating d.c. and lile meter indicates the average value of the input.

A.C voltmeter using fullwave rectifier:

The a.c. voltmeter using full wave rectifier is achieved by using bridge rectifier consisting of four diodes, as shown in the Fig



OHMMETER (SERIES TYPE OHMMETER)

A D'Arsonval movement is connected in series with a resistance R_1 and a battery which is connected to a pair of terminals A and B, across which the unknown resistance is connected. This forms the basic type of series ohmmeter, as shown in Fig. 4.30 (a).

The current flowing through the movement then depends on the magnitude of the unknown resistance. Therefore, the meter deflection is directly proportional to the value of the unknown resistance.

Referring to Fig. 4.30 (a)

R_1 = current limiting resistance
R_2 = zero adjust resistance
V = battery
R_m = meter resistance
R_x = unknown resistance

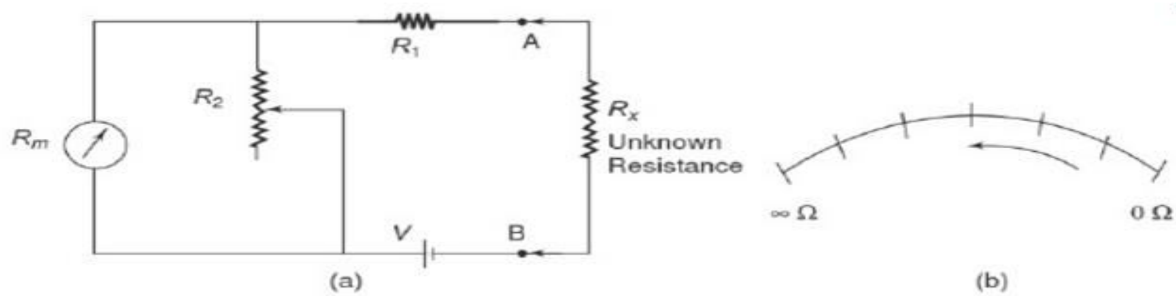


Fig. 4.30 (a) Series type ohmmeter (b) Dial of series ohmmeter

Calibration of the Series Type Ohmmeter

To mark the "0" reading on the scale, the terminals A and B are shorted, i.e. the unknown resistance $R_x = 0$, maximum current flows in the circuit and the shunt resistance R_2 is adjusted until the movement indicates full scale current (I_{fsd}). The position of the pointer on the scale is then marked "0" ohms.

Similarly, to mark the "∞" reading on the scale, terminals A and B are open, i.e. the unknown resistance $R_x = \infty$, no current flow in the circuit and there is no deflection of the pointer. The position of the pointer on the scale, is then marked as "∞" ohms.

By connecting different known values of the unknown resistance to terminals A and B, intermediate markings can be done on the scale. The accuracy of the instrument can be checked by measuring different values of standard resistance, i.e. the tolerance of the calibrated resistance, and noting the readings.

A major drawback in the series ohmmeter is the decrease in voltage of the internal battery with time and age. Due to this, the full scale deflection current drops and the meter does not read "0" when A and B are shorted. The variable shunt resistor R_2 across the movement is adjusted to counteract the drop in battery voltage, thereby bringing the pointer back to "0" ohms on the scale.

It is also possible to adjust the full scale deflection current without the shunt R_2 in the circuit, by varying the value of R_1 to compensate for the voltage drop. Since this affects the calibration of the scale, varying by R_2 is much better solution. The internal resistance of the coil R_m is very low compared to R_1. When R_2 is varied, the current through the movement is increased and the current through R_2 is reduced, thereby bringing the pointer to the full scale deflection position.

The series ohmmeter is a simple and popular design, and is used extensively for general service work.

Therefore, in a series ohmmeter the scale marking on the dial, has "0" on the right side, corresponding to full scale deflection current, and "∞" on the left side corresponding to no current flow, as given in Fig. 4.30 (b).

Values of R_1 and R_2 can be determined from the value of R_x which gives half the full scale deflection.

$$R_h = R_1 + R_2 \,||\, R_m = R_1 + \frac{R_2 R_m}{R_2 + R_m}$$

where R_h = half of full scale deflection resistance.

The total resistance presented to the battery then equals $2R_h$ and the battery current needed to supply half scale deflection is $I_h = V/2\,R_h$.

To produce full scale current, the battery current must be doubled.

Therefore, the total current of the ckt, $I_t = V/R_h$

The shunt current through R_2 is given by $I_2 = I_t - I_{fsd}$

The voltage across shunt, V_{sh}, is equal to the voltage across the meter.

Therefore $\quad V_{sh} = V_m$

$$I_2 R_2 = I_{fsd} R_m$$

Therefore $\quad R_2 = \dfrac{I_{fsd} R_m}{I_2}$

But $\quad I_2 = I_t - I_{fsd}$

$\therefore \quad R_2 = \dfrac{I_{fsd} R_m}{I_t - I_{fsd}}$

But $\quad I_t = \dfrac{V}{R_h}$

Therefore $\quad R_2 = \dfrac{I_{fsd} R_m}{V/R_h - I_{fsd}}$

Therefore $\quad R_2 = \dfrac{I_{fsd} R_m R_h}{V - I_{fsd} R_h}$

As $\quad R_h = R_1 + \dfrac{R_2 R_m}{R_2 + R_m}$

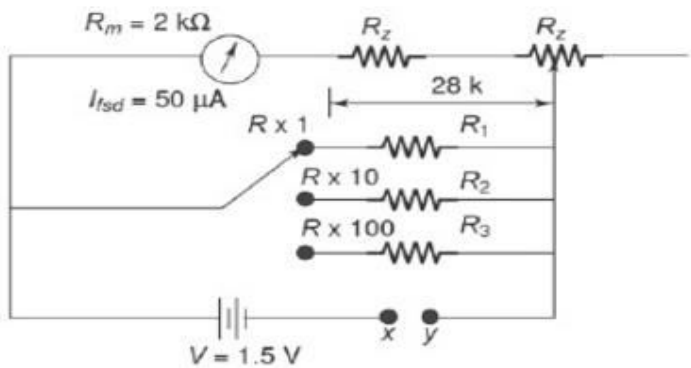
Therefore $\quad R_1 = R_h - \dfrac{R_2 R_m}{R_2 + R_m}$

Hence $$R_1 = R_h - \frac{\dfrac{I_{fsd}\, R_m\, R_h}{V - I_{fsd}\, R_h} \times R_m}{\dfrac{I_{fsd}\, R_m\, R_h}{V - I_{fsd}\, R_h} + R_m}$$

Therefore $$R_1 = R_h - \frac{I_{fsd}\, R_m\, R_h}{V}$$

Hence, R_1 and R_2 can be determined.

Multirange Ohmmeter The ohmmeter circuit shown in Fig. is only for a single range of resistance measurement. To measure resistance over a wide range of values, we need to extend the ohmmeter ranges. This type of ohmmeter is called a multirange ohmmeter, shown in Fig. 4.31.



SHUNT TYPE OHMMETER

The shunt type ohmmeter given in Fig consists of a battery in series with an adjustable resistor R_1, and a D'Arsonval movement

The unknown resistance is connected in parallel with the meter, across the terminals A and B, hence the name shunt type ohmmeter.

In this circuit it is necessary to have an ON/OFF switch to disconnect the battery from the circuit when the instrument is not used.

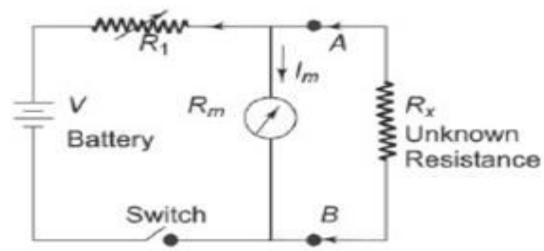


Fig. Shunt type ohmmeter

Calibration of the Shunt Type Ohmmeter

To mark the "0" ohms reading on the scale, terminals A and B are shorted, i.e. the unknown resistance $R_x = 0$, and the current through the meter movement is zero, since it is bypassed by the short-circuit. This pointer position is marked as "0" ohms.

Similarly, to mark "∞" on the scale, the terminals A and B are opened, i.e. $R_x = \infty$, and full current flows through the meter movement; by appropriate selection of the value of R_1, the pointer can be made to read full scale deflection current. This position of the pointer is marked "∞" ohms. Intermediate marking can be done by connecting known values of standard resistors to the terminals A and B.

This ohmmeter therefore has a zero mark at the left side of the scale and an ∞ mark at the right side of the scale, corresponding to full scale deflection current as shown in Fig.

Fig. Dial of shunt type ohmmeter

The shunt type ohmmeter is particularly suited to the measurement of low values of resistance. Hence it is used as a test instrument in the laboratory for special low resistance applications.

CALIBRATION OF DC INSTRUMENT

The process of calibration involves the comparison of a given instrument with a standard instrument, to determine its accuracy. A dc voltmeter may be calibrated with a standard, or by comparison with a potentiometer. The circuit in Fig. is used to calibrate a dc voltmeter; where a test voltmeter reading V is compared to the voltage drop across R. The voltage drop across R is accurately measured with the help of a standard meter. A rheostat, shown in Fig. is used to limit the current.

A voltmeter tested with this method can be calibrated with an accuracy of ±0.01%.

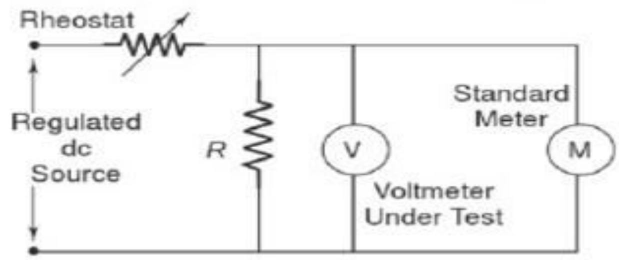


Fig. Calibration of voltmeter

Electronic multimeter:

For the measurement of d.c. as well as a.c. voltage and current, resistance, an electronic multimeter is commonly used. It is also known as Voltage-Ohm Meter (VOM) or multimeter The important salient features of YOM are as listed below.

1) The basic circuit of YOM includes balanced bridge d.c. amplifier.

2) To limit the magnitude of the input signal, RANGE switch is provided. By properly adjusting input attenuator input signal can be limited.

3) It also includes rectifier section which converts a.c. input signal to the d.c. voltage.

4) It facilitates resistance measurement with the help of internal battery and additional circuitry.

5) The various parameters measurement is possible by selecting required function using FUNCTION switch.

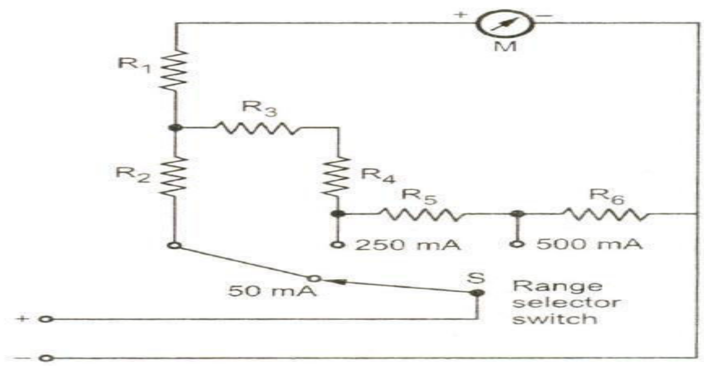
6) The measurement of various parameters is indicated with the help of indicating Meter.

Use of multimeter for D.C measurement:

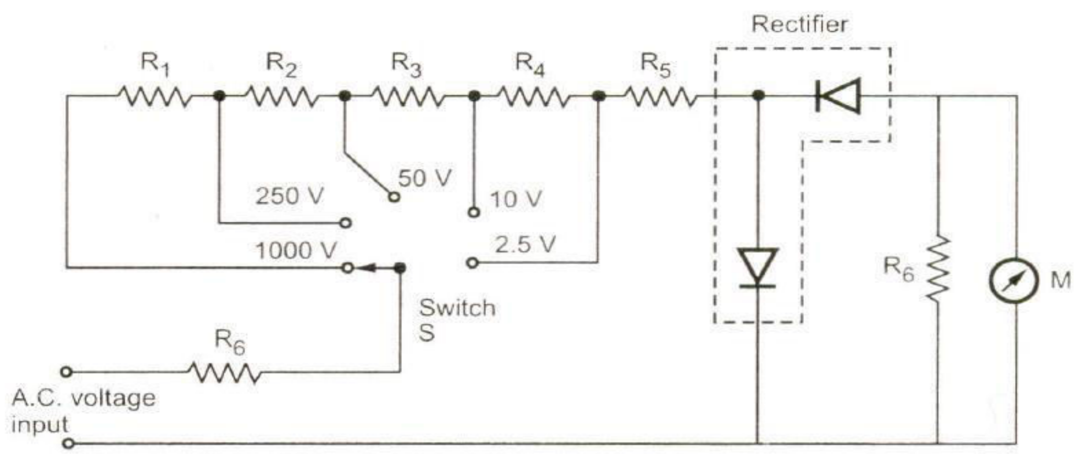
For getting different ranges of voltages, different series resistances are connected in series which can be put in the circuit with the range selector switch. We can get different ranges to measure the d.c. voltages by selecting the proper resistance in series with the basic meter.

Use of multimeter as ammeter:

To get different current ranges, different shunts are connected across the meter with the help of range selector switch. The working is same as that of PMMC ammeter



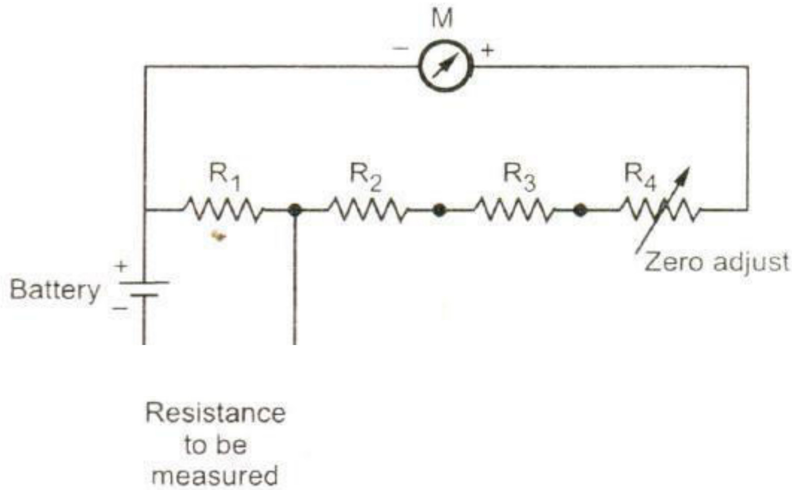
Use of multimeter for measurement of A.C voltage:



The rectifier used in the circuit rectifies a.c. voltage into d.c. voltage for measurement of a.c. voltage before current passes through the meter. The other diode is used for the protection purpose.

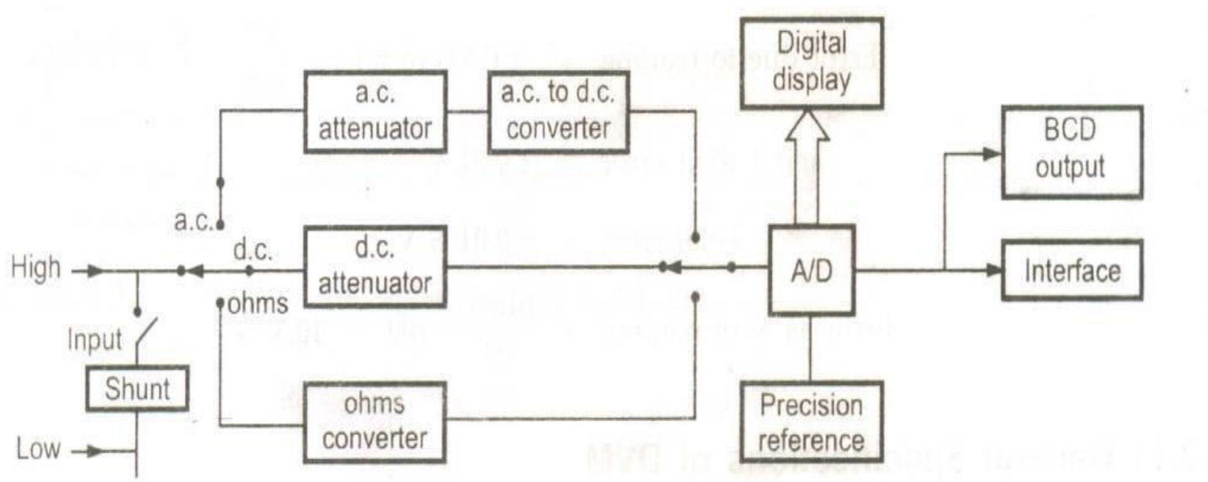
Use of multimeter for resistance measurement:

The Fig shows ohmmeter section of multimeter for a scale multiplication of 1. Before any measurement is made, the instrument is short circuited and "zero adjust" control is varied until the meter reads zero resistance i.e. it shows full scale current. Now the circuit takes the form of a variation of the shunt type ohmmeter. Scale multiplications of 100 and 10,000 can also be used for measuring high resistances. Voltages are applied the circuit with the help of battery.



Digital multimeters:

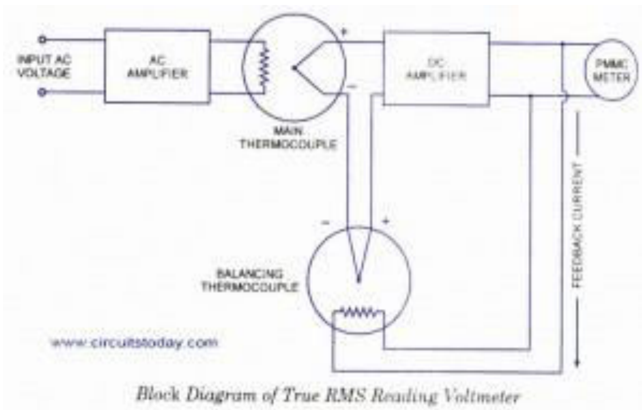
The digital multimeter is an instrument which is capable of measuring a.c. voltages, d.c. voltages, a.c. and d.c. currents and resistances *over* several ranges. The basic circuit of a digital multimeter is always a d.c. voltmeter as shown in the Fig



The current is converted to voltage by passing it through low shunt resistance. The a.c. quantities are *converted* to d.c. by employing various rectifier and filtering circuits. While for the resistance measurements the meter consists of a precision low current source that is applied across the unknown resistance while gives d.c. voltage. All the quantities are digitized using analog to digital converter and displayed in the digital form on the display.

The basic building blocks of digital multimeter are several *AID* converters, counting circuitry and an attenuation circuit. Generally dual slope integration type ADC is prefprred in the multimeters. The single attenuator circuit is used for both a.c. and d.c. measurements in many commercial multimeters.

True RMS Reading Voltmeter



True RMS Responding Voltmeters

RMS value of the sinusoidal waveform is measured by the**average reading voltmeter** of which scale is calibrated in terms of rms value. This method is quite simple and less expensive. But sometimes rms value of the non-sinusoidal waveform is required to be measured. For such a measurement a true rms reading voltmeter is required. True rms reading voltmeter gives a meter indication by sensing heating power of waveform which is proportional to the square of the rms value of the voltage.

Thermo-couple is used to measure the heating power of the input waveform of which heater is supplied by the amplified version of the input waveform. Output voltage of the thermocouple is proportional to the square of the rms value of the input waveform. One more thermo-couple, called the balancing thermo-couple, is used in the same thermal environment in order to overcome the difficulty arising out of non-linear behaviour of the thermo-couple. Non-linearity of the input circuit thermo-couple is cancelled by the similar non-linear effects of the balancing thermo-couple. These thermo-couples form part of a bridge in the input circuit of a dc amplifier, as shown in block diagram.

AC waveform to be measured is applied to the heating element of the main thermocouple through an ac amplifier. Under absence of any input waveform, output of both thermo-couples are equal so error signal, which is input to dc amplifier, is zero and therefore indicating meter connected to the output of dc amplifier reads zero. But on the application of input waveform, output of main thermo-couple upsets the balance and an error signal is produced, which gets amplified by the dc amplifier and fedback to the heating element of the balancing thermo-couple. This feedback current reduces the value of error signal and ultimately makes it zero to obtain the balanced bridge condition. In this balanced condition, feedback current supplied by the dc amplifier to the heating element of the balance thermo-couple is equal to the ac current flowing in the heating element of main thermo-couple. Hence this direct current is directly proportional to the rms value of the input ac voltage and is indicated by the meter connected in the output of the dc amplifier. The PMMC meter may be calibrated to read the rms voltage directly.

By this method, rms value of any voltage waveform can be measured provided that the peak excursions of the waveform do not exceed the dynamic range of the ac amplifier.

Recommended questions:

1. What is measurement? What are the two basic requirements of any measurement?

2. List the advantages of an electronic measurement.

3. Define and cxplain tilc term 'Calibration '.

4. *How the performance characteristics of an instrumcnt are classified?*

5. Definc and explain the following static characteristics of an instrulllcnt :

i) Accuracy ii) Precision iii) Static error iv) Resolution

*v) SensitiVity v/) Threshold vii) Zero drift viii) Rcproducibility [**jan 05,08 jul 07]** ix) Lillearity and x) Stability*

6. Explain how the accuracy can be specified for an instmment.

7. Distinguish clearly between aCCliracy and precision.

9. Explaill tile terms relative error and relative percentage error.

10. What is *scale span of an instrument?*

11. Define a dynamic response of an instrument.

12. Define the following terms,

i) *Speed of response ii) Lag iii) Fidelity ivY Dynamic error.*

13. Define and explain the types of errors possible in an instrument.

14. Define limiting errors. Derive the expression for relative limiting error.

15. A moving coil voltmeter has a uniform scale with 100 *divisions, the full scale reading* is *200 V and 1/10 of scale division can be estimated with a fair degree of certainity. Determine the resolution of the instrument in volt.* [Ans. : 0.2 **V]]**

16. A digital voltmeter has a read out range from 0-9999 counts. Determine the resolution of the instrument in volt when the full scale reading is 9.999 \I. [Ans. : 1 **mV**

17. A true value of voltage across resister is 50 *V. The instrument reads* 49 *V. Calculate*

i) absolute error ii) percentage error iii) percentage accuracy.

18. What is *sensitil'ity of voltmcters* & *Explain.*

19. *What* is *a loading effect* & *Explain with the suitable example.*

20. *Explain the operation of basic d.c. voltmeter.*

21. *Explain the working of d.c. 11ultirange voltmeter.*

22. *State the requirements of a multiplier.*

UNIT-2

AF Wave analyzer

The wave analyzer consists of a very narrow pass-band filter section which can Be tuned to a particular frequency within the audible frequency range (20Hz to 20 KHz)). The block diagram of a wave analyzer is as shown in fig 1.

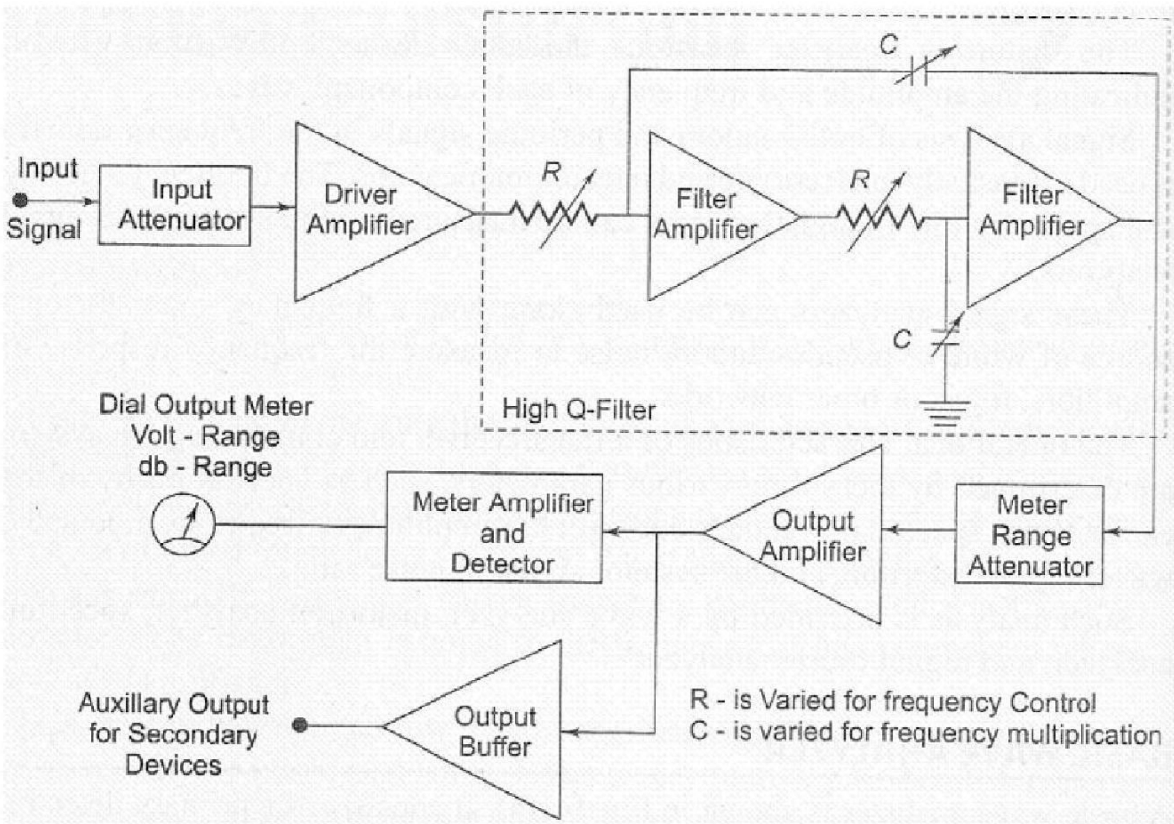


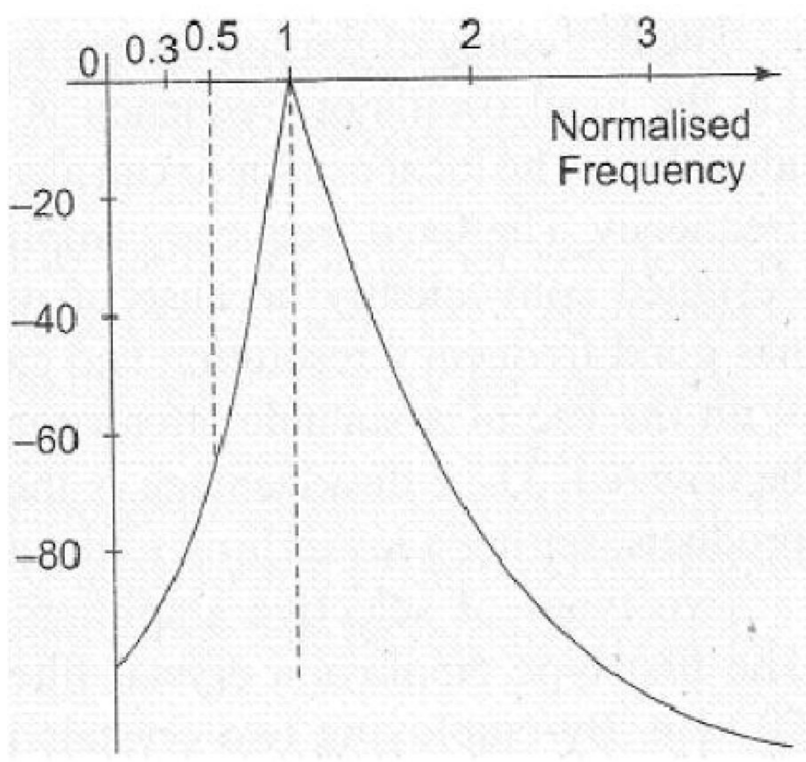
Fig 1: Frequency wave analyzer

The complex wave to be analyzed is passed through an adjustable attenuator which serves as a range multiplier and permits a large range of signal amplitudes to be analyzed without loading the amplifier.

The output of the attenuator is then fed to a selective amplifier, which amplifies the selected requency. The driver amplifier applies the attenuated input signal to a high-Q active filter. This high-Q filter is a low pass filter which allows the frequency which is selected to pass and reject all others. The magnitude of this selected frequency is indicated by the meter and the filter section identifies the frequency of the component. The filter circuit consists of a cascaded RC resonant circuit and amplifiers. For selecting the frequency range, the capacitors generally used are of the closed tolerance polystyrene type and the resistances used are precision potentiometers. The capacitors are used for range changing and the potentiometer is used to change the frequency within the selected pass-band, Hence this wave analyzer is also called a Frequency selective voltmeter. The entire AF range is covered in decade steps by switching capacitors in the RC section.

The selected signal output from the final amplifier stage is applied to the meter circuit and to an unturned buffer amplifier. The main function of the buffer amplifier is to drive output devices, such as recorders or electronics counters.

The meter has several voltage ranges as well as decibel scales marked on it. It is driven by an average reading rectifier type detector. The wave analyzer must have extremely low input distortion, undetectable by the analyzer itself. The band width of the instrument is very narrow typically about 1% of the selective band given by the following response characteristics shows in fig.1.2



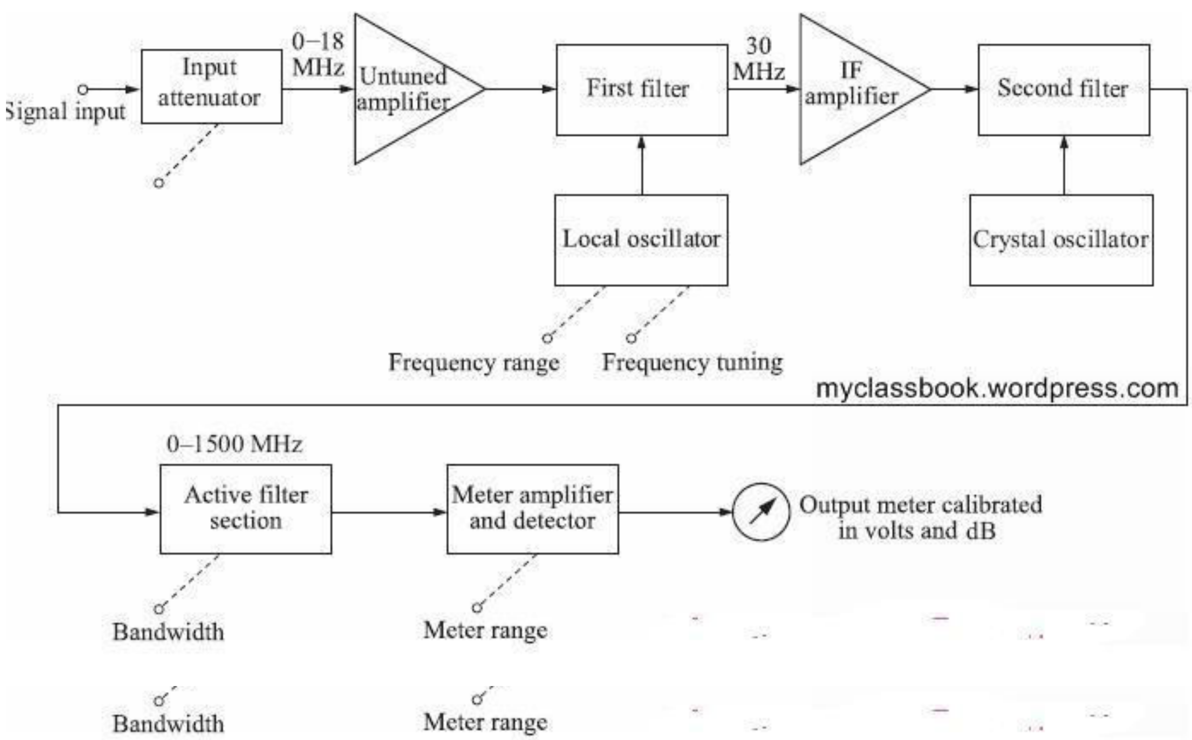
Application of wave analyzer:

1. Electrical measurements
2. Sound measurements
3. Vibration measurements.

In industries there are heavy machineries which produce a lot of sound and vibrations, it is very important to determine the amount of sound and vibrations because if it exceeds the permissible level it would create a number of problems. The source of noise and vibrations is first identified by wave analyzer and then it is reduced by further circuitry.

Heterodyne wave analyzer

A wave analyzer, in fact, is an instrument designed to measure relative amplitudes of single frequency components in a complex waveform. Basically, the instrument acts as a frequency selective voltmeter which is used to the frequency of one signal while rejecting all other signal components. The desired frequency is selected by a frequency calibrated dial to the point of maximum amplitude. The amplitude is indicated either by a suitable voltmeter or CRO.This instrument is used in the MHz range. The input signal to be analysed is heterodyned to a higher IF by an internal local oscillator. Tuning the local oscillator shifts various signal frequency components into the pass band of the IF amplifier. The output of the IF amplifier is rectified and is applied to the metering circuit. The instrument using the heterodyning principle is called a *heterodyning tuned voltmeter.*



The block schematic of the wave analyser using the heterodyning principle is shown in fig. above. The operating frequency range of this instrument is from 10 kHz to 18 MHz in 18 overlapping bands selected by the frequency range control of the local oscillator. The bandwidth is controlled by an active filter and can be selected at 200, 1000, and 3000 Hz.

Wave analyzers have very important applications in the following fields:

1) Electrical measurements
2) Sound measurements and
3) Vibration measurements.

The wave analyzers are applied industrially in the field of reduction of sound and vibrations generated by rotating electrical machines and apparatus. The source of noise and vibrations is first identified by wave analyzers before it can be reduced or eliminated. A fine spectrum analysis with the wave analyzer shows various discrete frequencies and resonances that can be related to the motion of machines. Once, these sources of sound and vibrations are detected with the help of wave analyzers, ways and means can be found to eliminate them.

Harmonic distortion :

The **total harmonic distortion (THD)** is a measurement of the harmonic distortion present in a signal and is defined as the ratio of the sum of the powers of all harmonic components to the power of the fundamental frequency. **Distortion factor**, a closely related term, is sometimes used as a synonym.

In audio systems, lower distortion means the components in a loudspeaker, amplifier or microphone or other equipment produce a more accurate reproduction of an audio recording.

To understand a system with an input and an output, such as an audio amplifier, we start with an ideal system where thetransfer function is linear and time-invariant. When a signal passes through a non-ideal, non-linear device, additional content is added at the harmonics of the original frequencies. THD is a measurement of the extent of that distortion.

When the main performance criterion is the "purity" of the original sine wave (in other words, the contribution of the original frequency with respect to its harmonics), the measurement is most commonly defined as the ratio of the RMS amplitude of a set of higher harmonic frequencies to the RMS amplitude of the first harmonic, or fundamental, frequency

$$\mathrm{THD_F} = \frac{\sqrt{V_2^2 + V_3^2 + V_4^2 + \cdots}}{V_1}$$

where V_n is the RMS voltage of the nth harmonic and $n = 1$ is the fundamental frequency.

Spectrum Analyzer

The modern spectrum analyzers use a narrow band super heterodyne receiver. Supereterodyne is nothing but mixing of frequencies in the super above audio range. The functional block diagram of super heterodyne spectrum analyzer or RF spectrum nalyzer as shown in the Figure

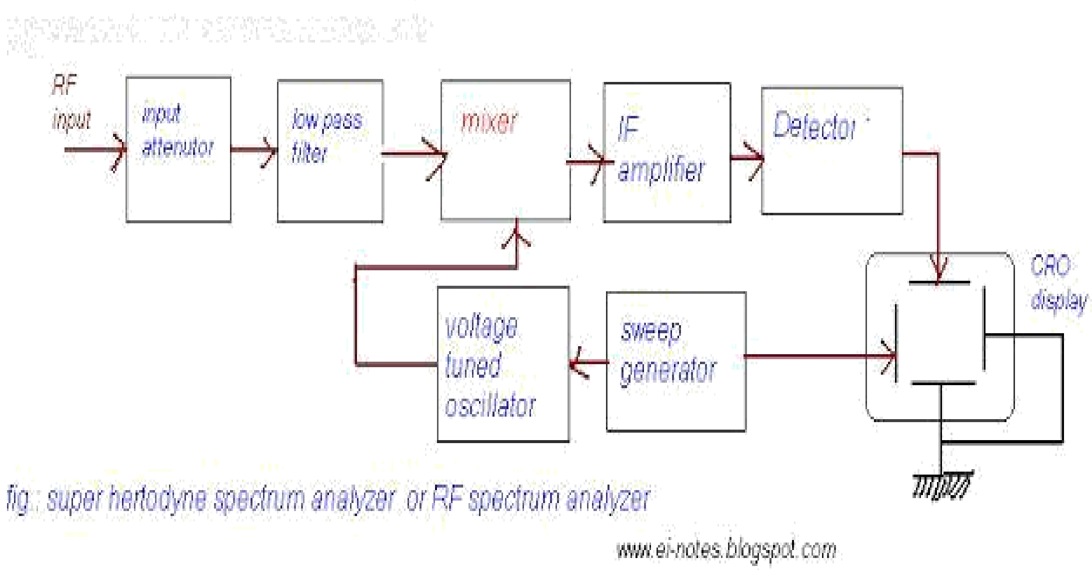


fig.: super hertodyne spectrum analyzer or RF spectrum analyzer

The RF input to be analyzed is applied to the input attenuator. After attenuating, the signal is fed to low pass filter. The low pass filter suppresses high frequency components and allows low frequency components to pass through it. The output of the low pass filter is given to the mixer, where this signal is fixed with the signal coming from voltage controlled or voltage tuned oscillator.

This oscillator is tuned over 2 to 3 GHz range. The output of the mixer includes two signals whose amplitudes are proportional to the input signal but their frequencies are the sum and difference of the input signal and the frequency of the local oscillator.

Since the frequency range of the oscillator is tuned over 2 to 3 GHz, the IF amplifier is tuned to a narrow band of frequencies of about 2 GHz. Therefore only those signals which are separated from the oscillator frequency by 2 GHz are converted to Intermediate Frequency (IF) band. This IF signal is amplified by IF amplifier and then rectified by the detector. After completing amplification and rectification the signal is applied to vertical plates of CRO to produce a vertical deflection on the CRT screen. Thus, when the saw tooth signal sweeps, the oscillator also sweeps linearly from minimum to maximum frequency range i.e., from 2 to 3 GHz.

Here the saw tooth signal is applied not only to the oscillator (to tune the oscillator) but also to the horizontal plates of the CRO to get the frequency axis or horizontal deflection on the CRT screen. On the CRT screen the vertical axis is calibrated in amplitude and the horizontal axis is calibrated in frequency.

FFT spectrum analyzer

A spectrum analyzer, which uses computer algorithm and an analog to digital conversion phenomenon and produces spectrum of a signal applied at its input is known as digital Fourier or digital FFT or digital spectrum analyzer

Principle

When the analog signal to be analyzed is applied, the A/D converter digitizes the analog signal (i.e., converts the analog signal into digital signal). The digitized signal, which is nothing but the set of digital numbers indicating the amplitude of the analog signal as a function of time is stored in the memory of the digital computer. From the stored digitized data, the spectrum of the signal is computed by means of computer algorithm.

Description:

The block arrangement of a digital Fourier analyzer is illustrated in the figure above .The analog signal to be ana1ysed is applied to the low pass filter, which passes only low frequency signals and rejects high pass spurious signals. This filter section is used mainly, to prevent aliasing. The output of low pass filter is given to the attenuator. The attenuator is a voltage dividing network whose function is to set the input signal to the level of the A/D converter. The use of attenuator prevents the converter from overloading. The function of A/D converter is to convert the samples of analog data into digital i.e. ., to digitize the analog signal. When the output of A/D converter is applied to the digital computer, the computer analyzes the digitized data and adjusts the attenuator setting accordingly in order to obtain the maximum output from the inverter without any overloading. As soon as the entire analog signal is sampled and digitized by the A/D converter) computer performs calculations on the data according to the programmed algorithm and the calculated spectral components are stored in the memory of the computer

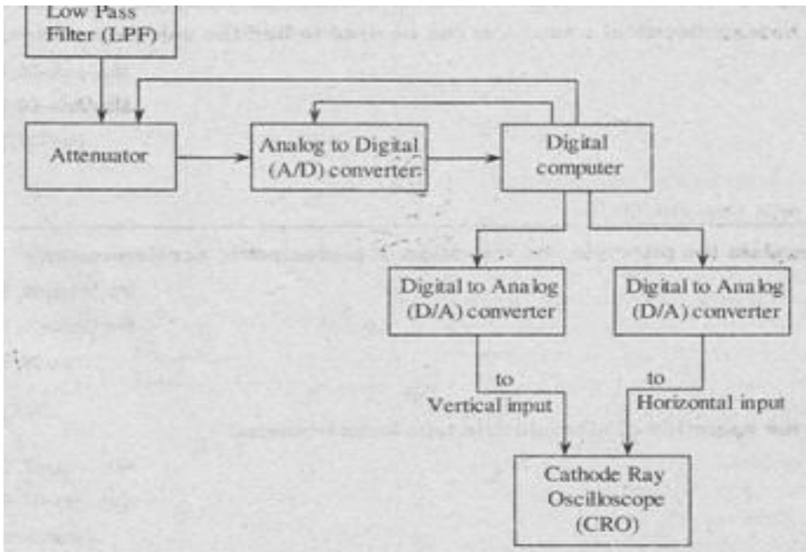


Fig Digital Fourier Analyzer

If the spectral display is to be viewed on the oscilloscope, the digital values of spectral components stored in the computer memory are converted into analog by using D/A converters and then applied to the CRO. Thus the spectral display of the input waveform is obtained on the CRT screen.

Advantages

1. The use of computer avoids most of the hardware circuitry such as electronic switches, Filters and PLLs. The use of less hardware reduces the cost of the analyzer.
2. More mathematical calculations can be carried-out on the spectral display.
3. The rate of sampling analog signal can be modified in order to obtain better spectral display.

STANDARD SIGNAL GENERATOR:-

A standard signal generator produces known and controllable voltages. It is used as power sourcefor the measurement of gain, signal to noise ratio (SN), bandwidth standing wave ratio and other properties.

It is extensively used in the measuring of radio receivers and transmitter instrument is provided with a means of modulating the carrier frequency, which is indicated by the dial setting on the front panel.

The modulation is indicated by a meter. The output signal can be Amplitude Modulated (AM) or Frequency Modulated (FM). Modulation may be done by a sine wave, Square, rectangular, or a pulse wave.

The elements of a conventional signal generator:

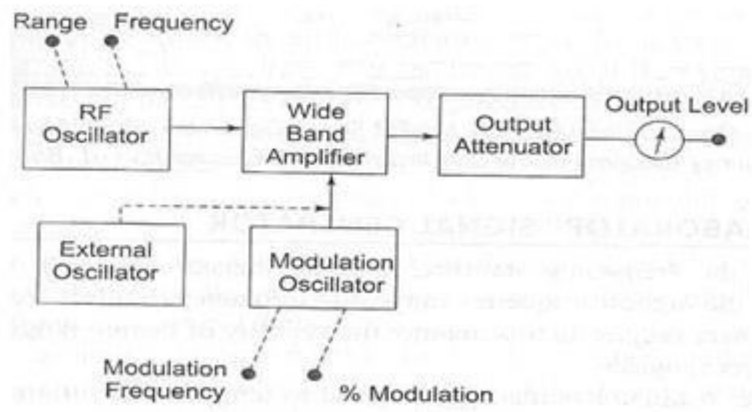
1) RF Osillator
(2) Wide band amplifier.
(3) External Osillator.
4) Modulation Osillator
(5) Out put attenuator.

The carrier frequency is generated by a very stable RF oscillator using an LC tank circuit, having a constant output over any frequency range. The frequency of oscillations is indicated by thefrequency range control and the venire dial setting. AM is provided by an internal sine wavegenerator or from an external source.

The signal generator is called an oscillator. A Wien bridge oscillator is used in this generator.The Wien bridge oscillator is the best of the audio frequency range. The frequency of oscillations can be changed by varying the capacitance in the oscillator.

The frequency can also be changed in steps by switching the resistors of different values. The output of the Wien bridge oscillator goes to the function switch.

The function switch directs the oscillator output either to the sine wave amplifier or to the square wave shaper. At the output, we get either a square or sine wave.The output is varied by means of an attenuator.



The instrument generates a frequency ranging from 10 Hz to 1 MHz continuously vV (rms).The output is taker through a push-pull amplifier. For low output, the impedance is 6000. The square wave amplitudes can be varied from 0 - 20 v (peak). It is possible to adjust the symmetry of the square wave from 30 -70%. The instrument requires only 7W of power at 220V 50Hz.

The front panel of a signal generator consists of the following.

l. Frequency selector: It selects the frequency in different ranges and varies it continuously in a ratio of 1: 11. The scale is non-linear.
2. Frequency multiplier: It selects the frequency range over 5 decades from 10 Hz to 7 MHz
3. Amplitude multiplier: It attenuates the sine wave in 3 decades, x l x 0.1 and x 0.01.
4. Variable amplitude: It attenuates the sine wave amplitude continuously
5. Symmetry control: It varies the symmetry of the square wave from 30% to 70%.
6. Amplitude: It attenuates the square wave output continuously.
7. Function switch: It selects either sine wave or square output.
8. Output available: This provides sine wave or square wave output.
9. Sync: This terminal is used to provide synchronization of the internal signal with an external signal.
10. On-Off Switch

Sweep Generator

It provides a sinusoidal output voltage whose frequency varies smoothly and continuously over an entire frequency band, usually at an audio rate. The process of frequency modulation may be accomplished electronically or mechanically. It is done electronically by using the modulating voltage to vary the reactance of the oscillator tank circuit component, and mechanically by means of a motor driven capacitor, as provided for in a modern laboratory type signal generator. Figure shows a basic block diagram of a sweep generator. The frequency sweeper provides a variable modulating voltage which causes the capacitance of the master oscillator to vary. A representative sweep rate could be of the order of 20 sweeps/second. A manual control allows independent adjustment of the oscillator resonant frequency. The frequency sweeper provides a varying sweep voltage synchronization to drive the horizontal deflection plates of the CRO. Thus the amplitude of the response of a test device will be locked and displayed on the screen.

To identify a frequency interval, a marker generator provides half sinusoidal waveforms at any frequency within the sweep range. The marker voltage can be added to the sweep voltage of the CRO during alternate cycles of the sweep voltage, and appears superimposed on the responsecurve.

The automatic level control circuit is a closed loop feedback system which monitors the RF level at some point in the measurement system. This circuit holds the power delivered to the load or test circuit constant and independent o frequency and impedance changes. A constant power level prevents any source mismatch and also provides a constant readout calibration with frequency.

SQUARE AND PULSE GENERATOR:-

These generators are used as measuring devices in combination with a CRO. They provide both quantitative and qualitative information of the system under test. They are made use of in transient response testing of amplifiers. The fundamental difference between a pulse generator and a square wave generator is in the duty cycle.

Duty cycle = A square wave generator has a 500/o duty cycle.

Requirements of a Pulse

1. The pulse should have minimum distortion, so that any distortion, in the display is solely due to the circuit under test.

2. The basic characteristics of the pulse are rise time, overshoot, ringing, sag, and undershoot.

3. The pulse should have sufficient maximum amplitude, if appreciable output power is required by the test circuit, e.g. for magnetic core memory. At the same time, the attenuation range shouldbe adequate to produce small amplitude pulses to prevent over driving of some test circuit.

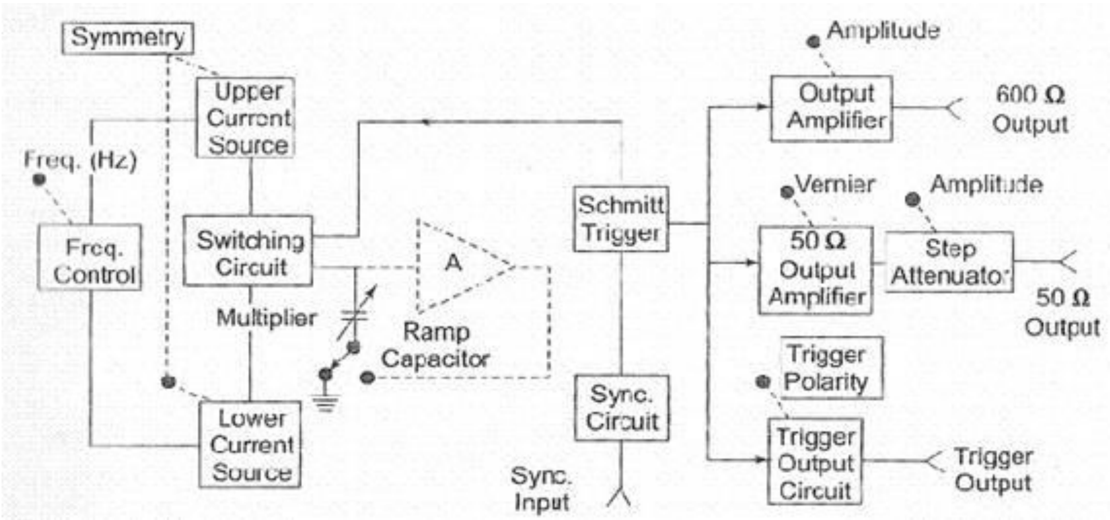
4. The range of frequency control of the pulse repetition rate (PRR) should meet the needs of theexperiment. For example, a repetition frequency of 100 MHz is required for testing fast circuits.Other generators have a pulse-burst feature which allows a train of pulses rather than a continuous output.

5. Some pulse generators can be triggered by an externally applied trigger signal; conversely, pulse generators can be used to produce trigger signals, when this output is passed through a differentiator circuit.

6. The output impedance of the pulse generator is another important consideration. In a fast pulsesystem, the generator should be matched to the cable and the cable to the test circuit. A mismatch would cause energy to be reflected back to the generator by the test circuit, and this may be rereflected by the generator, causing distortion of the pulses.

7. DC coupling of the output circuit is needed, when dc bias level is to be maintained.
The basic circuit for pulse generation is the asymmetrical multi-vibrator.

A laboratory type square wave and pulse generator is shown in Fig 6.1



The frequency range of the instrument is covered in seven decade steps from 1Hz to 10 MHz, with a linearly calibrated dial for continuous adjustment on all ranges.

The duty cycle can be varied from 25 - 75%. Two independent outputs are available, a 50Ω source that supplies pulses with a rise and fall time of 5 ns at 5V peak amplitude and a 600Ω source which supplies pulses with a rise and fall tme of 70 ns at 30 V peak amplitude. The instrument can be operated as a freerunning genenrator or, it can be synchronized with externalsignals.

The basic generating loop consists of the current sources, the ramp capacitor, the Schmitt triggerand the current switching circuit as shown in the fig

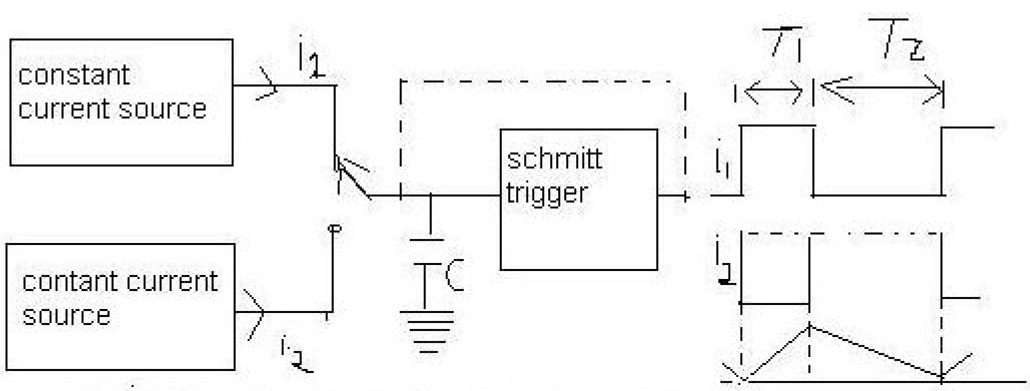


fig. Basic generating loop,

The upper current source supplies a constant current to the capacitor and the capacitor voltage increases linearly. When the positive slope of the ramp voltage reaches the upper limit set by theinternal circuit components, the Schmitt trigger changes state. The trigger circuit output becomesnegative and reverses the condition of the current switch. The capacitor discharges linearly, controlled by the lower current source.

When the negative ramp reaches a predetermined lower level, the Schmitt trigger switches back to its original state. The entire process is then repeated.

The ratio i1/i2 determines the duty cycle, and is controlled by symmetry control. The sum of i1 and i2 determines the frequency. The size of the capacitor is selected by the multiplier switch. The unit is powered by an intenal supply that provides regulated voltages for all stages of the instrument.

The precautionary measures to be taken in a signal generator application:-

A signal generator is an instrument, which can produce various types of wave forms such as sine wave, square wave, triangular wave, saw tooth wave, pulse trains etc. As it can generate a variety of waveforms it is widely used in applications like electronic troubleshooting anti development, testing the performance of electronic equipments etc. In such applications a signal generator is used to provide known test conditions (i.e., desired signals of known amplitude and frequency

Hence, the following precautionary measures should be taken while using a signal generator for an application.

1. The amplitude and frequency of the output of the signal generator should be made stable and well known.

2. There should be provision for controlling the amplitude of signal generator output from very small to relatively large values.

3. The output signal of generator should not contain any distortion and thus, it should possess very low harmonic contents.

4. Also, the output of the signal generator should be less spurious.

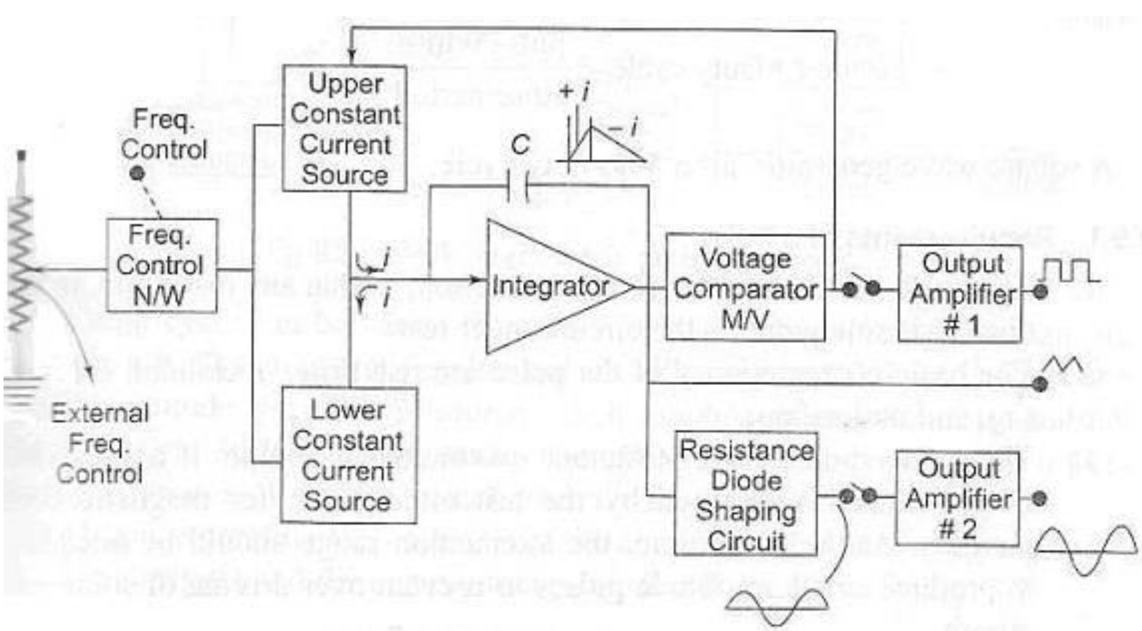
FUNCTION GENERATOR

A function generator produces different waveforms of adjustable frequency. The common outputwaveforms are the sine, square, triangular and saw tooth waves. The frequency may be adjusted, from a fraction of a Hertz to several hundred kHz lie various outputs of the generator can bemade available at the same time. For example, the generator can provide a square wave to testthe linearity of a rectifier and simultaneously provide a saw tooth to drive the horizontal deflection amplifier of the CRO to provide a visual display.

Capability of Phase Lock the function generator can be phase locked to an external source. One function generator can be used to lock a second function generator, and the two output signals can be displaced in phase by adjustable amount. In addition, the fundamental frequency of one generator can be phase locked to a harmonic of another generator, by adjusting the amplitude and phase of the harmonic; almost any waveform can be generated by addition.

The function generator can also be phase locked to a frequency standard and its output waveforms will then have the same accuracy and stability as the standard source.

The block diagram of a function generator:



The block diagram of a function generator is illustrated in fig. Usually the frequency is controlled by varying the capacitor in the LC or RC circuit. In the instrument the frequency is controlled by varying the magnitude of current which drives the integrator. The instrument produces sine, triangular and square waves with a frequency range of 0.01 Hz to 100 kHz.

The frequency controlled voltage regulates two current sources. The upper current source supplies constant current to the integrator whose output voltage increases linearly with time, according to the equation of the output signal voltage. An increase or decrease in the currentincreases or decreases the slope of the output voltage and hence controls the frequency. The voltage comparator multi-vibrator changes states at a pre-determined maximum level of the integrator output voltage. This change cuts off the upper current supply and switches on the lower current supply. The lower current source supplies a reverse current to the integrator, so that its output decreases linearly with time. When the output reaches a pre-determined minimum level, the voltage comparator again changes state and switches on the Lower current source. The output of the integrator is a triangular waveform whose frequency is determined by the magnitude of the current supplied by the constant current sources. The comparator output delivers a square wave voltage of the same frequency.

$e = - 1/C \int idt$

The resistance diode network alters the slope of the triangular wave as its amplitude changes and produces a sine wave with less than 1% distortion.

Arbitrary Waveform Generator, AWG

The waveforms produced by arbitrary waveform generators, AWGs can be either repetitive or sometimes just a single-shot. If the AWG waveform is only a single shot, then a triggering mechanism is needed to trigger the AWG and possibly the measuring instrument.

The AWG is able to generate an arbitrary waveform defined by a set of values, i.e. "waypoints" entered to set the value of the waveform at specific times. They can make up a digital or even an analogue waveform.

As a result an arbitrary waveform generator is a form of test equipment that is able to produce virtually any waveshape that is required.

Arbitrary Waveform Generator techniques

There are a number of ways of designing arbitrary waveform generators. They are based around digital techniques, and their design falls into one of two main categories:

- ***Direct Digital Synthesis, DDS:*** This type of arbitrary waveform generator is based around the DDS types of frequency synthesizer, and sometimes it may be referred to as an Arbitrary Function Generator, AFG.
- ***Variable-clock arbitrary waveform generator*** The variable clock arbitrary function generator is the more flexible form of arbitrary waveform generator. These arbitrary waveform generators are generally more flexible, although they do have some limitations not possessed by the DDS versions. Sometimes these generators are referred to as just arbitrary waveform generators, AWGs rather than arbitrary function generators.
- ***Combined arbitrary waveform generator*** This format of AWG combines both of the other forms including the DDS and variable clock techniques. In this way the advantages of both systems can be realised within a single item of test equipment.

Arbitrary waveform generator resolution and speed:

Two of the main specifications for an arbitrary waveform generator are their resolution and also the speed. These two parameters determine the precision with which the waveform can be reproduced. They are governed by different elements within the arbitrary waveform generator circuit.

The amplitude resolution is governed by the resolution of the digital to analogue converter (D/A or D2A). This is described in terms of the number of bits. A 12 bit resolution provides 4096 amplitude steps.

The speed of the arbitrary waveform generator is also very important. The maximum repetition rate for the waveform is governed by two factors: the length of the waveform in terms of the number of samples required to simulate the waveform and the maximum clock frequency. For example if the arbitrary waveform generator had a maximum clock frequency of 25 MHz and the waveform had 1000 points, then the maximum repetition rate would be 25 kHz. If a higher repetition rate was required, then it would be necessary to decrease the number of samples as it would not be possible to increase the clock frequency in the arbitrary waveform generator!

Arbitrary waveform generator applications:

AWGs are used in many applications where specialised waveforms are required. These can be within a whole variety of sectors of the electronics industry.

To give a view of some of the AWG applications, it is possible for DDS-based arbitrary waveform generators is to create signals with precisely controlled phase offsets or ratio-related frequencies. This enables the generation of signals like polyphase sine waves, I-Q constellations, or simulation of signals from geared mechanical systems such as jet engines. Complex channel-channel modulations are also possible.

The arbitrary waveform generator may not be the most widely used of items of test instrumentation, but they can be immensely useful in a variety of applications. Modern arbitrary waveform generators are very flexible and can be used to create very specific waveforms for use in testing a variety of applications.

Direct digital synthesizer, DDS technology lends itself to being used within arbitrary waveform generators, AWGs. Those AWGs that use DDS technology are often referred to as arbitrary function generators, or AFGs.

The reason for being called arbitrary function generators is that they often appear as an extension of the function generator test instruments that are available.

Arbitrary waveform generators using direct digital synthesis technology are able to benefit from the technology, while not adding unwanted additional complexity and cost. DDS technology has developed considerably in recent years and this makes them a very attractive option to form the basis of a waveform generator. As a result arbitrary function generators are relatively widely used.

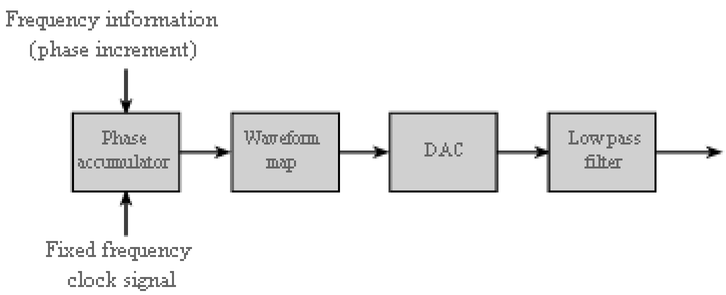
Arbitrary function generator basics:

As mentioned, this type of arbitrary waveform generator is based around the DDS types of frequency synthesizer, and sometimes it may be referred to as an Arbitrary Function generator, AFG.

The arbitrary function generator uses integrated circuits intended for direct digital frequency synthesizers, but enables an arbitrary waveform generator circuit to be created relatively easily and for an economic price.

To look at how an arbitrary function generator works, it is necessary to look at the operation of a direct digital synthesizer.

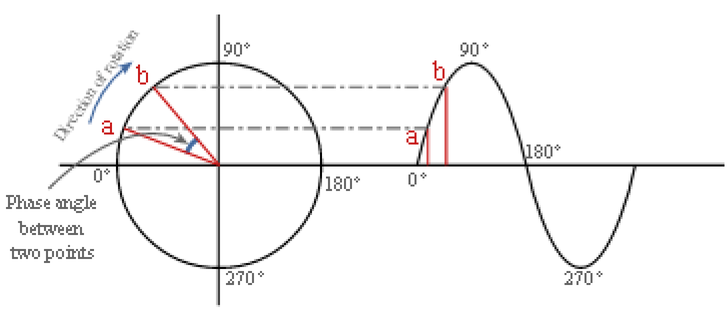
This circuit operates by storing the points of a waveform in digital format, and then recalling them to generate the waveform. These points can be on any form of repetitive waveform that is required. The rate at which the DDS completes one waveform governs the frequency. The basic block diagram of the DDS based arbitrary waveform generator is shown below.



DDS frequency synthesizer as used in an arbitrary function generator, AFG

The operation of the DDS within the arbitrary function generator can be envisaged by looking at the way that phase progresses over the course of one cycle of the waveform.

The phase is often depicted as a line of phasor rotating around a circle. As the phase advances around the circle, this corresponds to advances in the waveform. The faster is progresses, the sooner it completes a cycle and the hence the higher the frequency.



Phase angle of points on a sine wave

The direct digital synthesizer operates by storing various points of the required waveform in digital format in a memory. These can then be recalled to generate the waveform as they are required.

To simulate the phase advances a phase accumulator is used. This takes in phase increment information, and clock pulses from a clock. For each clock pulse, the phase will advance a certain amount. The greater the increment, the larger the phase advance, and hence the higher the frequency generated.

At each clock pulse the phase information is presented to the memory and the relevant location is accessed, proving the waveform information for that particular phase angle.

It can be seen that any waveform can be loaded into the memory; although a sine wave is shown on the diagram, the actual waveform could be anything.

While it is possible to load certain preset waveforms into the memory, it is also possible to load user generated ones in as well. These make the test instrument an arbitrary waveform generator or arbitrary function generator rather than a standard function generator.

Advantages and disadvantages of AFG:

While the arbitrary function generator or DDS based version of the arbitrary waveform generator, has many advantages, there are also some disadvantages that should also be taken into account when choosing what type of signal generator to use.

Arbitrary function generator advantages

- ***Sub Hz frequency resolution:*** By using a long word length phase accumulator in the phase accumulator of the DDS, it is possible to achieve sub-Hertz frequency resolution levels.
- ***Down sampling:*** Waveforms are automatically truncated by sampling to allow repetition rates above the clock frequency.
- ***Digital modulation:*** It is possible to add digital modulation words to the phase accumulator to provide a means of providing digital modulation.

Arbitrary function generator disadvantages

- ***Waveform jitter:*** Waveform jitter is an issue with arbitrary function generators because frequencies are up-sampled or down-sampled and this results in missing samples and hence jitter. Only frequencies equal to the clock frequency divided by the waveform length and its sub multiples are not sampled and therefore they do not suffer from this problem

- ***Single waveform capability:*** It is only possible to generate a single waveform at a time because memory segmentation and waveform sequencing is not possible using a DDS arbitrary function generator

The arbitrary function generator is the ideal instrument where a variety of programmed waveforms are required without the added flexibility and complexity of the more expensive variable clock arbitrary waveform generator. For most laboratory applications, the arbitrary function generator is an ideal choice.

Unit 3

Oscilloscopes

Introduction:

In studying the various electronic, electrical networks and systems, signals which are functions of time, are often encountered. Such signals may be periodic or non periodic in nature. The device which allows, the amplitude of such signals, to be displayed primarily as " function of time, is called **cathode ray** oscilloscope, commonly known as C.R.O. The CR.O gives the visual representation of the time varying signals. The oscilloscope has become an universal instrument and is probably most versatile tool for the development of electronic circuits and systems. It is an integral part of electronic laboratories.

The oscilloscope is, in fact, a voltmeter. Instead of the mechanical deflection of a metallic pointer as used in the normal voltmeters, the oscilloscope uses the movement of an electron beam against a fluorescent screen, which produces the movement of a visible spot. The movement of such spot on the screen is proportional to the varying magnitude of the signal, which is under measurement.

Basic Principle:

The electron beam can be deflected in two directions : the horizontal or x-direction and the vertical or y-direction. Thus an electron beam producing a spot can be used to produce two dimensional displays, Thus CRO. can be regarded as a fast x-y plotter. The x-axis and y-axis can be used to study the variation of one voltage as a function of another. Typically the x-axis of the oscilloscope represents the time while the y-axis represents variation of the input voltage signal. Thus if bhe input voltage signal applied to the y-axis of CRO. is sinusoidally varying and if x-axis represents the time axis, then the spot moves sinusoidally, and the familiar sinusoidal waveform can be seen on the screen of the oscilloscope. The oscilloscope is so fast device that it can display the periodic signals whose time period is as small as microseconds and even nanoseconds. The CRO. Basically operates on voltages, but it is possible to convert current, pressure, strain, acceleration and other physical quantities into the voltage using transducers and obtain their visual representations on the CRO.

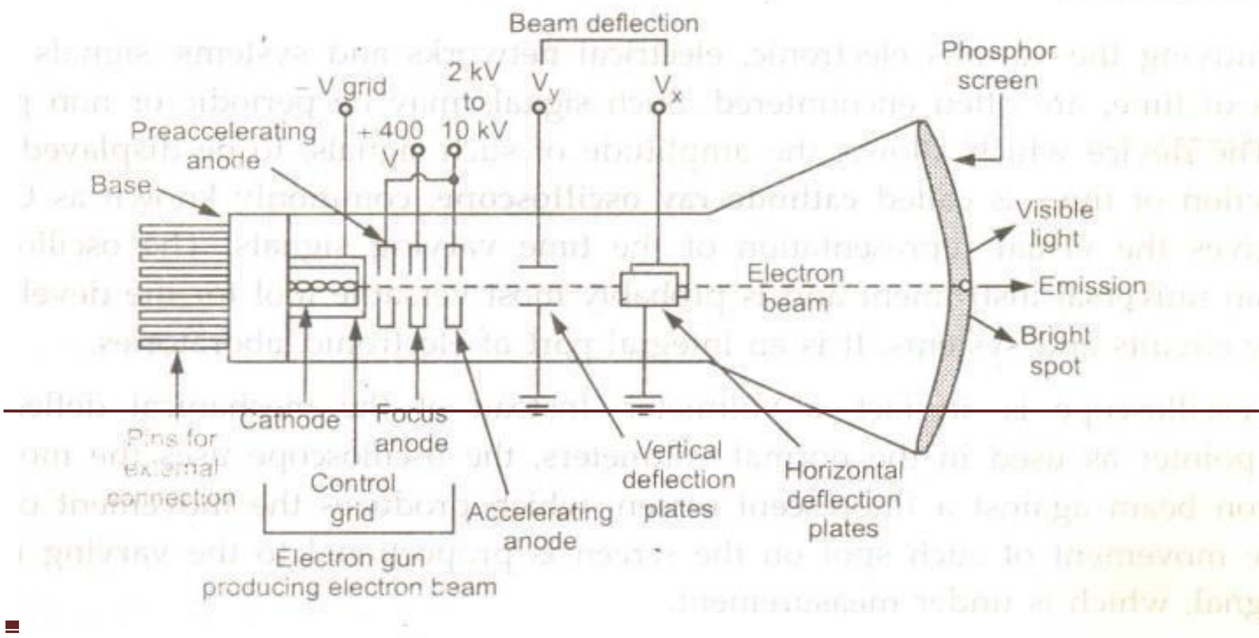
Cathode Ray Tube (CRT):

The cathode ray tube (CRT) is the heart of the CR.O. the CRT generates the electron beam, ,accelerates the beam, deflects the beam and also has a screen where beam becomes visible ,as a spot. The main parts of the CRT are:

i) Electron gun ii) Deflection system iii) Fluorescent screen

iv) Glass tube or envelope v) Base

A schematic diagram of CRT, showing its structure and main components is shown in the Fig.



Electron Gun:

The electron gun section of the cathode ray tube provides a sharply focused electron beam directed :towards the fluorescent-coated screen. This section starts from theql1ally heated cathode, limiting the electrons. The control grid is give!! negative potential with respect to cathode dc. This grid controls the number of electrons in the beam, going to the screen.

The momentum of the electrons (their number x their speed) determines the intensity, or brightness, of the light emitted from the fluorescent screen due to the electron bombclrdl1lent. The light emitted is usually of the green colour. Because the electrons are negatively charged, a repulsive force is created by applying a negative voltage to the control grid (in CRT, voltages applied to various grids are stated with respect to cathode, which is taken as common point). This negative control voltage can be made variable.

Deflection System:

When the electron beam is accelerated it passes through the deflection system, with which beam can be positioned anywhere on the screen. The deflection system of the cathode-ray-tube consists of two pairs of parallel plates, referred to as the vertical and horizontal deflection plates. One of the plates' in each set is connected to ground (0 V), To the other plate of each set, the external deflection voltage is applied through an internal adjustable gain amplifier stage, To apply the deflection voltage externally, an external terminal, called the Y input or the X input, is available.

As shown in the Fig. , the electron beam passes through these plates. A positive voltage applied to the Y input terminal (Vy) Causes the beam to deflect vertically upward due to the attraction forces, while a negative voltage applied to. the Y input terminal will cause the electron beam to deflect vertically downward, due to the repulsion forces. When the voltages are applied simultaneously to vertical and horizontcl1 deflecting plates, the electron beam is deflected due to the resultant-of these two voltages.

Fluorescent Screen:

The light produced by the screen does not disappear immediately when bombardment by electrons ceases, i.e., when the signal becomes zero. The time period for which the trace remains on the screen after the signal becomes zero is known as "persistence". The persistence may be *jS* short as a few microsecond, or as long as tens of seconds and minutes.

Long persistence traces are used in the study.. of transients. Long persistence helps in the study of transients since the trace is still seen on the screen after the transient has disappeared.

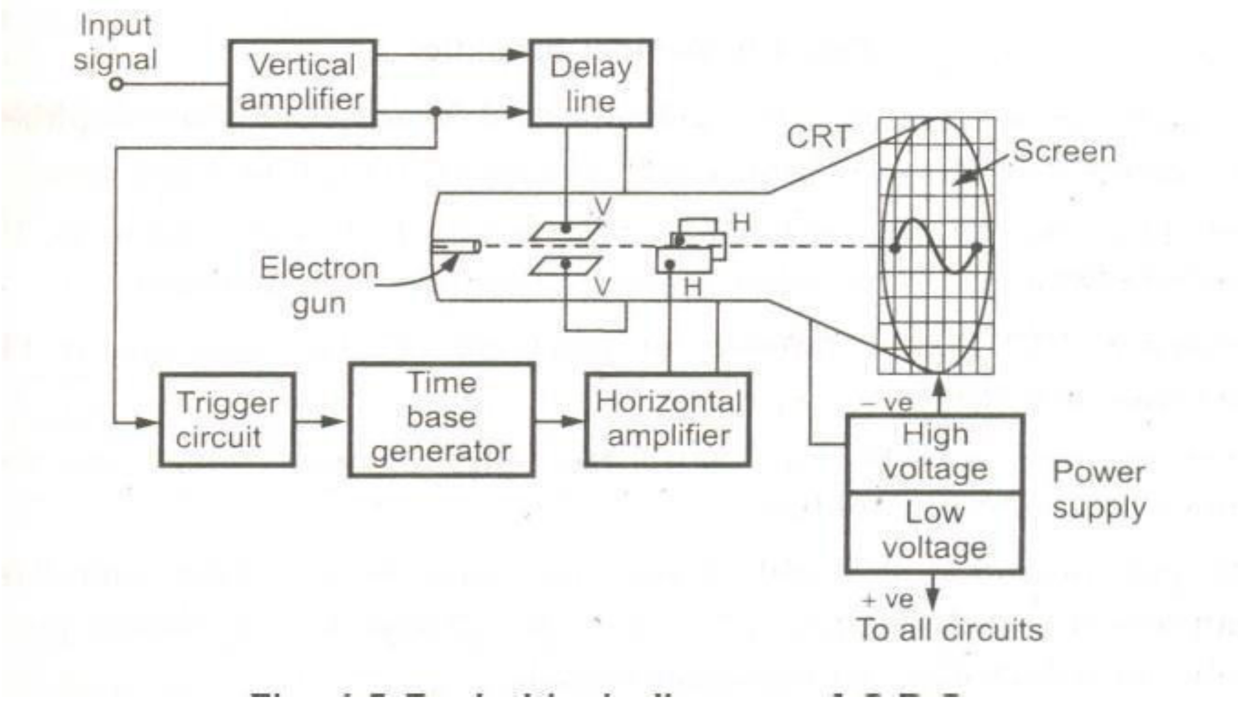
Phosphor screen characteristics:

Many phosphor materials having different excitation times and colours as well as different phosphorescence times are available. The type PI, P2, PI1 or P3I are the short persistence phosphors and are used for the general purpose oscilloscope

Medical oscilloscopes require a longer phosphor decay and hence phosphors like P7 and P39 are preferred for such applications. Very slow displays like radar require long persistence phosphors to maintain sufficient flicker free picture. Such phosphors are P19, P26 and, P33.

The phosphors P19, P26, P33 have low burn resistance. The phosphors PI, P2, P4, P7, Pll have medium burn resistance while PIS, P3I have high burn resistance.

Block diagram of simple oscilloscope:

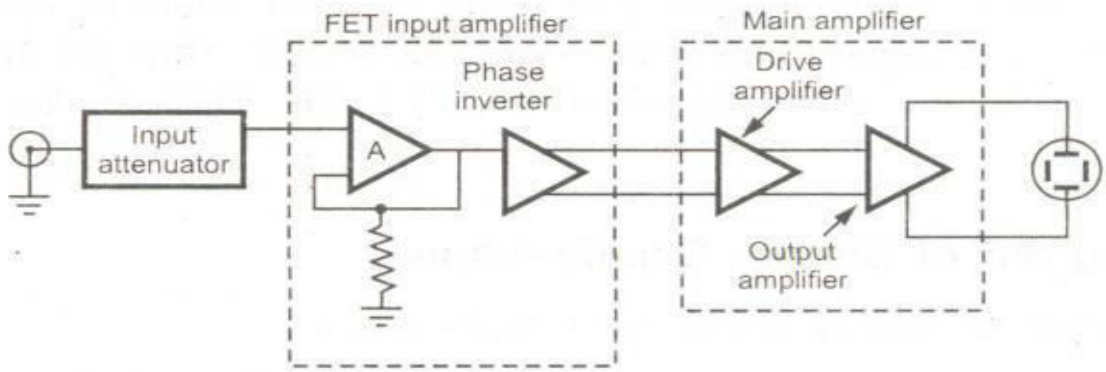


CRT:

This is the cathode ray tube which is the heart of CR.O. It is' used to emit the rlectrons required to strike the phosphor screen to produce the spot for the visual display of the signals.

Vertical Amplifier:

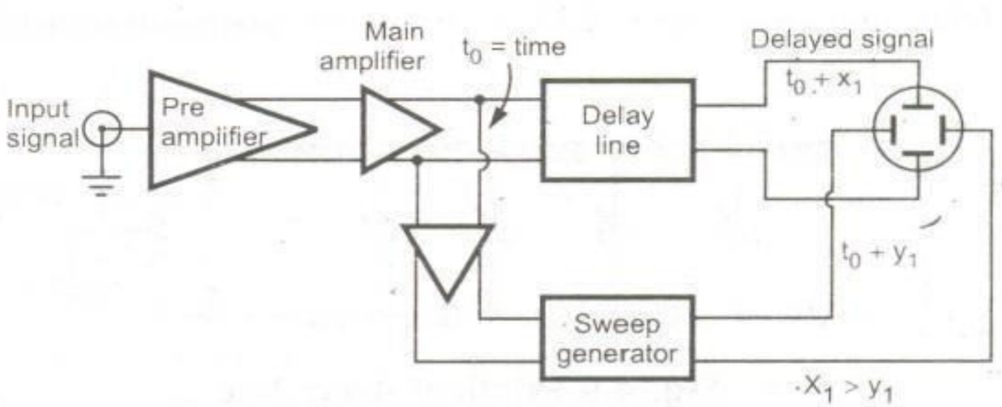
The input signals are generally not strong to provide the measurable deflection on the screen. Hence the vertical amplifier. stage is used Jo amplify the input signals. The amplifier stages used are generally wide band amplifiers so as to pass faithfully the entire band of frequencies to be measured. Similarly it contains the attenuator stages as well. The attenuators are used when very high voltage signals are to be examined, to bring the signals within the proper range of operation.



It consists of several stages with overall fixed sensltivity. The amplifier can be designed for stability and required bandwidth very easily due to the fixed gain. The input stage colrtsists of an attenuator followed by FET source follower. It has vel' high input impedance required to isolate the amplifier from the attenuator. It is followed by BJT emitter follower to match the output impedance of FET output With input of phase inverter. The phase inverter provides two antiphase output signals which are required to operate the push pull output amplifier. The push pull operation has advantages like better hum voltage cancellation, even harmonic suppression especially large 2nd harmonic, greater power output per tube and reduced number of defocusing and nonlinear effects.

Delay line:

The delay line is used to delay the signal for some time in the verticClI sections. When the delay line is not used, the part of the signal gets lost. Thus the input signal is not applied directly to the vertical plates but is delClyed bv some time using a delay line cu-cuit as shown in the Fig.



If the trigger pulse is picked off at a time t = to after the signal has passed through the main amplifier then signal is delayed by XI nanoseconds while sweep takes YI nanoseconds to reach. The design of delay line is such that the delay time XI is higher than the time YI' Generally XI is 200. nsec while tl;1.eYI is 80 ns, thus the sweep starts well in time and no part of the signal is lost. There are two types of delay lines used in CR.O. which are:

i) Lumped parameter delay line

ii) Distributed parameter delay line

Trigger circuit:

It is necessary that horizontal deflection starts at the same point of the input vertical signal, each time it sweeps. Hence to synchronize horizontal deflection with vertical deflection a synchronizing or triggering circuit is used. It converts the incoming signal into the triggering pulses, which are used for the synchronization.

Time base generator:

The time base generator is used to generate the sawtooth voltage, required to deflect the beam in the horizontal section. This voltage deflects the spot at a constant time dependent rate. Thus the x-axis' on the screen can be represented as time, which, helps to display and analyse the time varying signals.

LISSAJOUS FIGURES:

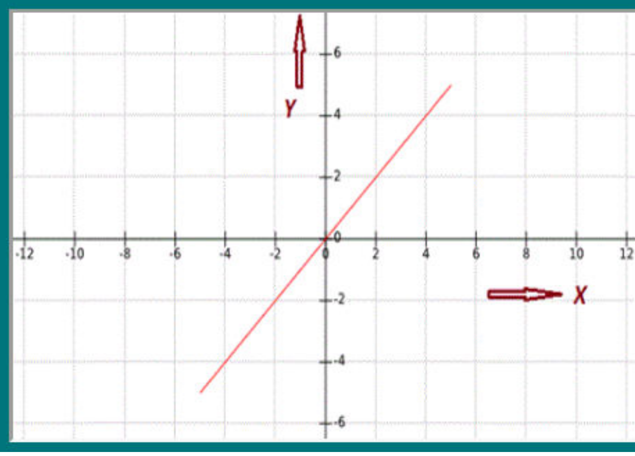
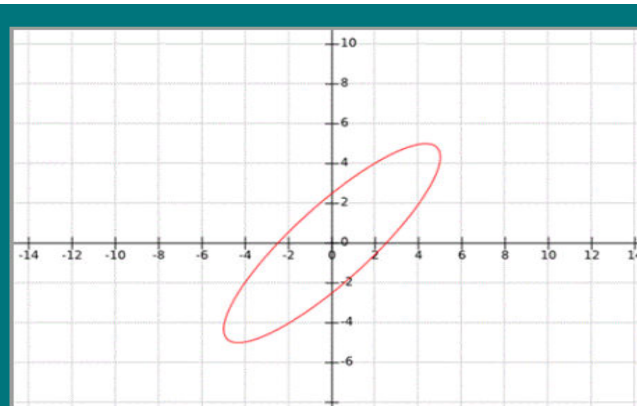
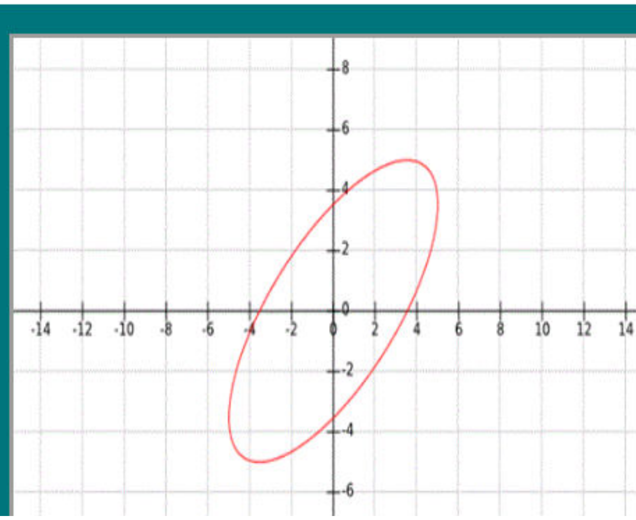
When both pairs of the deflection plates (horizontal deflection plates and vertical deflection plates) of CRO (**Cathode Ray Oscilloscope**) are connected to two sinusoidal voltages, the patterns appear at CRO screen are called the **Lissajous pattern**.

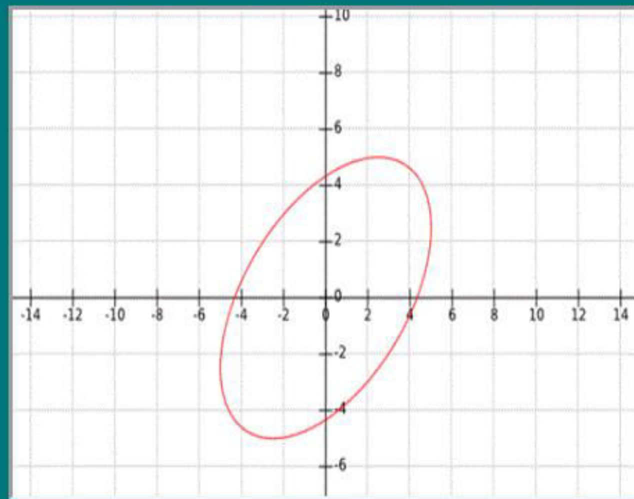
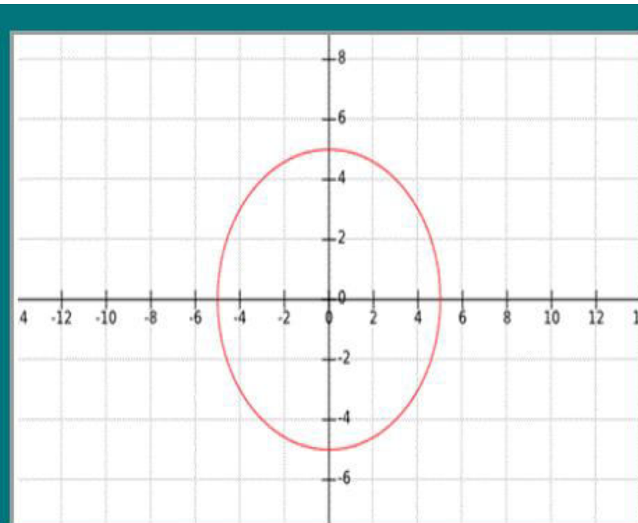
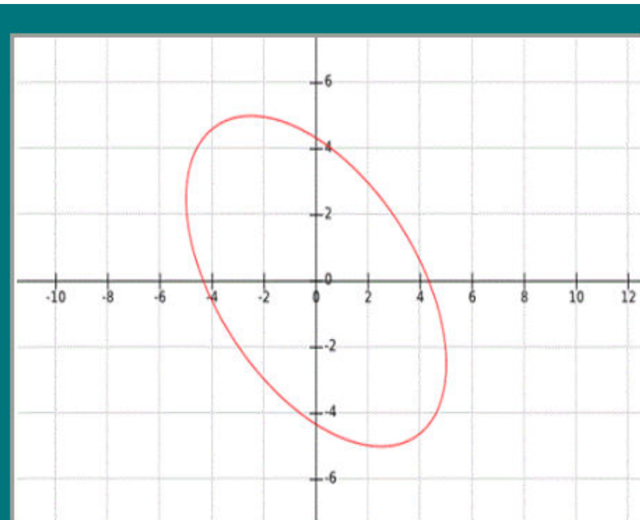
Shape of these **Lissajous pattern** changes with changes of phase difference between signal and ration of frequencies applied to the deflection plates (traces) of **CRO**. Which makes these **Lissajous patterns** very useful to analysis the signals applied to deflection plated of CRO. These lissajous patterns have two Applications to analysis the signals. To calculate the phase difference between two sinusoidal signals having same frequency. To determine the ratio frequencies of sinusoidal signals applied to the vertical and horizontal deflecting plates.

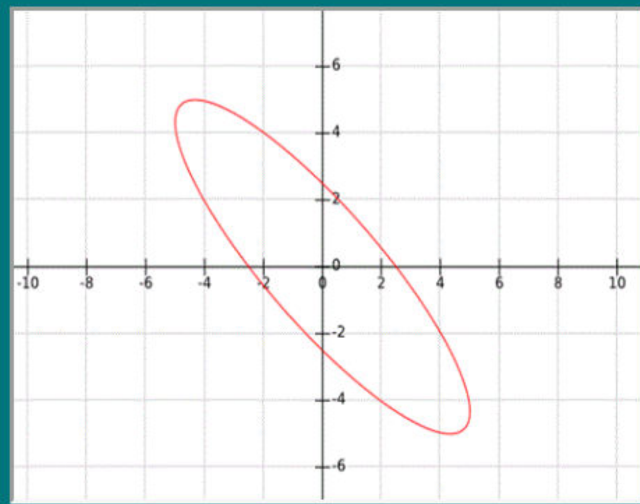
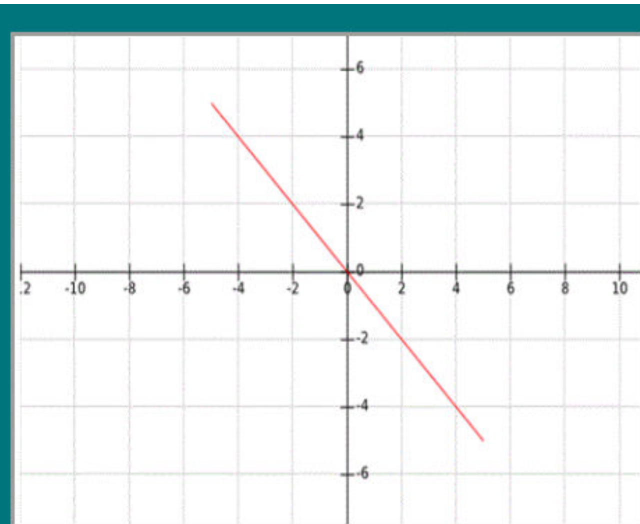
Calculation of the phase difference between two Sinusoidal Signals having same frequency

When two sinusoidal signals of same frequency and magnitude are applied two both pairs of deflecting plates of **CRO**, the Lissajous pattern changes with change of phase difference between signals applied to the CRO.

For different value of phase differences, the shape of Lissajous patterns is shown in figure below,

<table>
<tr><th>SL No.</th><th>Phase angle difference 'ø'</th><th>Lissajous Pattern appeared at CRO Screen</th></tr>
<tr><td>1</td><td>0º & 360º</td><td>
</td></tr>
<tr><td>2</td><td>30º or 330º</td><td>
</td></tr>
<tr><td>3</td><td>45º or 315º</td><td>
</td></tr>
</table>

4	60^o or 300^o	
5	90^o or 270^o	
6	120^o or 240^o	

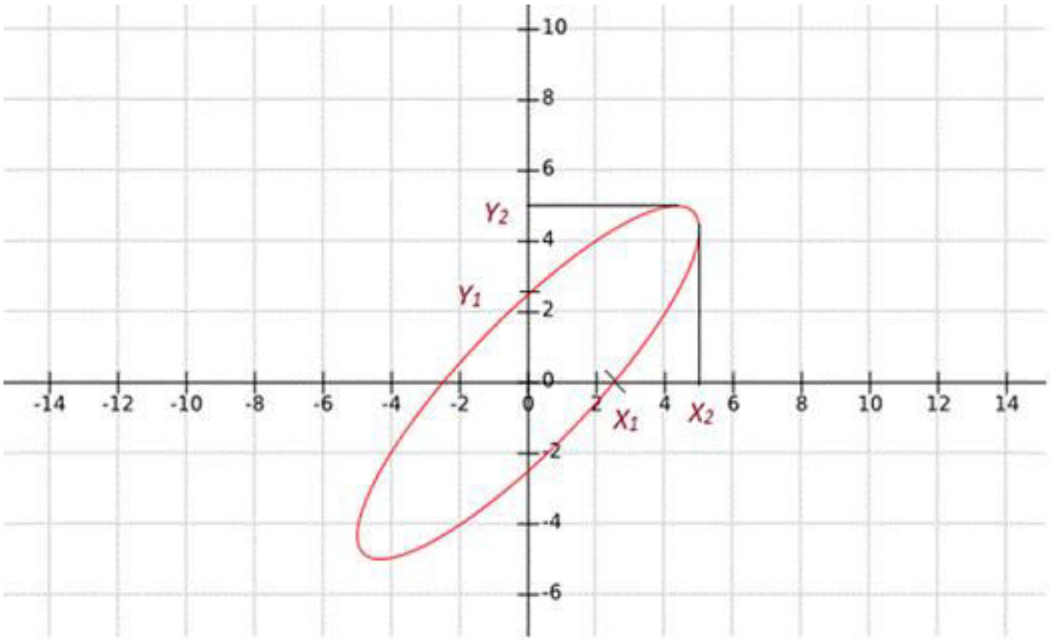
7	150^o or 210^o	
8	180^o	

There are two cases to determine the phase difference ø between two signals applied to the horizontal & vertical plates,

Case - I: When, $0 < ø < 90^o$ or $270^o < ø < 360^o$: -

As we studied above it clear that when the angle is in the range of $0 < ø < 90^o$ or $270^o < ø < 360^o$, the Lissajous pattern is of the shape of Ellipse having major axis passing through origin from first quadrant to third quadrant:

Let's consider an example for $0 < ø < 90^o$ or $270^o < ø < 360^o$, as shown in figure below,



In this condition the phase difference will be,

$$\emptyset = sin^{-1}\left(\frac{x_1}{x_2}\right) = sin^{-1}\left(\frac{y_1}{y_2}\right)$$

Another possibility of phase difference,

$$\emptyset' = 360^0 - \emptyset$$

From Above given Lissajous pattern

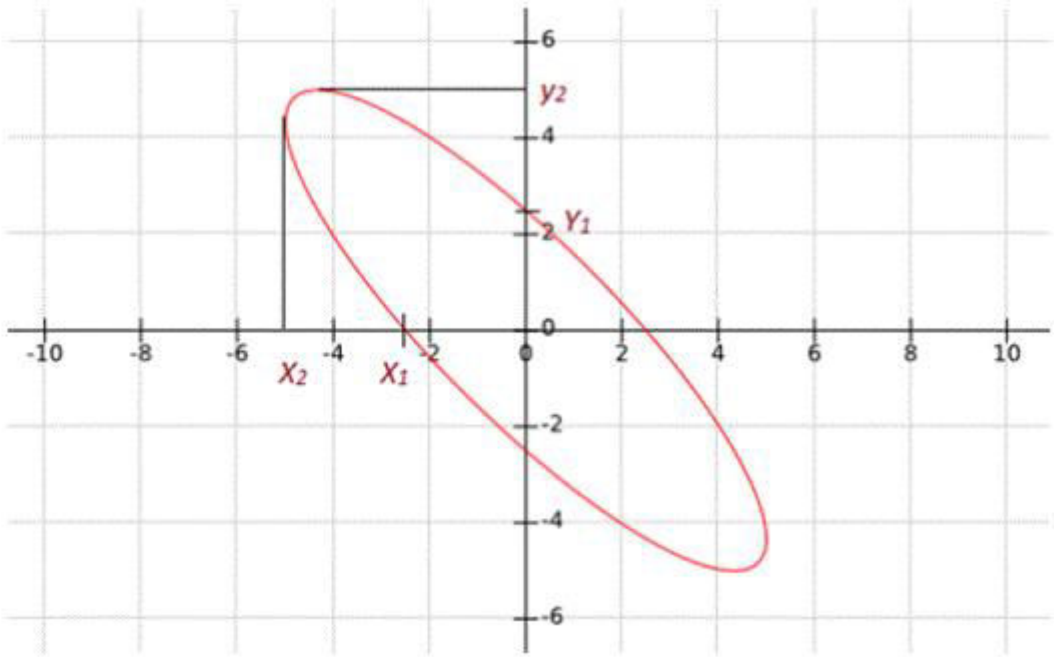
$$x_1 = 2.25\ \&\ x_2 = 4.5$$

$$\text{Hence, } \emptyset = sin^{-1}\left(\frac{x_1}{x_2}\right) = sin^{-1}\left(\frac{2.25}{4.5}\right) = 30^0$$

Another Possibility of Phase Difference,

$$\emptyset' = 360^0 - \emptyset = 360^0 - 30^0 = 330^0$$

Case - II: When $90^0 < ø < 180^0$ or $180^0 < ø < 270^0$



As we studied above it Clear that when the angle is in the range of $0^o < ø < 90^o$ or $270^o < ø < 360^o$, the Lissajous Pattern is of the shape of Ellipse having major axis passing through origin from second quadrant to fourth quadrant:

Let's consider an example for When, $90^o < ø < 180^o$ or $180^o < ø < 270^o$, as shown in figure below, In this condition the phase difference will be,

$$\emptyset = 180^0 - sin^{-1}\left(\frac{x_1}{x_2}\right) = 180^0 - sin^{-1}\left(\frac{y_1}{y_2}\right)$$

Another possibility of phase difference,

$$\emptyset' = 360^0 - \emptyset$$

From Above given Lissajous pattern

$$x_1 = 2.25 \;\&\; x_2 = 4.5$$

$$\text{Hence, } \emptyset = 180^0 - sin^{-1}\left(\frac{x_1}{x_2}\right) = 180^0 - sin^{-1}\left(\frac{2.25}{4.5}\right) = 180^0 - 30^0 = 150^0$$

Another Possibility of Phase Difference,

$$\emptyset' = 360^0 - \emptyset = 360^0 - 150^0 = 210^0$$

To determine the ratio of frequencies of signal applied to the vertical and horizontal deflecting plates:

To determine the ratio of frequencies of signal by using the Lissajous pattern, simply draw arbitrary horizontal and vertical line on lissajous pattern intersecting the Lissajous pattern. Now count the number of horizontal and vertical tangencies by Lissajous pattern with these horizontal and vertical line.

Then the ratio of frequencies of signals applied to deflection plates,

$$\frac{\omega_y}{\omega_x} = \frac{f_y}{f_x} = \frac{Number\ of\ horizontal\ tangencies}{Number\ of\ vertical\ tangencies}$$

Or,

$$\frac{\omega_y}{\omega_x} = \frac{f_y}{f_x} = \frac{(Number\ of\ intersections\ of\ lissajous\ pattern\ with\ horizontal\ line)}{(Number\ of\ intersections\ of\ lissajous\ pattern\ with\ horizontal\ line)}$$

Oscilloscope probes:

Oscilloscopes are widely used for test and repair of electronics equipment of all types. However it is necessary to have a method of connecting the input of the oscilloscope to the point on the equipment under test that needs monitoring.

To connect the scope to the point to be monitored it is necessary to use screened cable to prevent any pick-up of unwanted signals and in addition to this the inputs to most oscilloscopes use coaxial BNC connectors. While it is possible to use an odd length of coax cable with a BNC connector on one end and open wires with crocodile / alligator clips on the other, this is not ideal and purpose made oscilloscope probes provide a far more satisfactory solution.

Oscilloscope probes normally comprise a BNC connector, the coaxial cable (typically around a metre in length) and what may be termed the probe itself. This comprises a mechanical clip arrangement so that the probe can be attached to the appropriate test point, and an earth or ground clip to be attached to the appropriate ground point on the circuit under test.

Care should be taken when using oscilloscope probes as they can break. Although they are robustly manufactured, any electronics laboratory will consider oscilloscope probes almost as "life'd" items that can be disposed of after a while when they are broken. Unfortunately the fact that they are clipped on to leads of equipment puts a tremendous strain on the mechanical clip arrangement. This is ultimately the part which breaks.

X1 and X10 oscilloscope probes

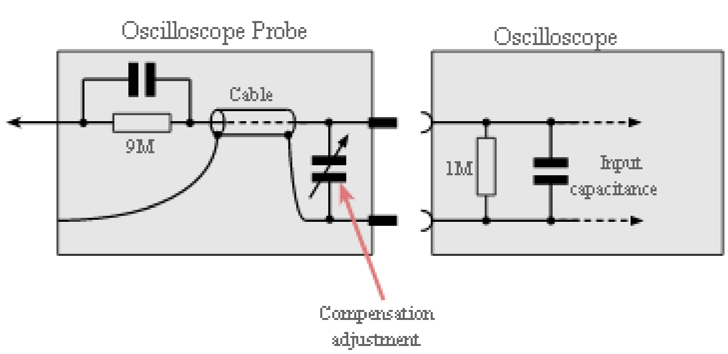
There are two main types of passive voltage scope probes. They are normally designated X1 and X10, although 1X and 10X are sometimes seen. The designation refers to the factor by which the impedance of the scope itelf is multiplied by the probe.

The X1 probes are suitable for many low frequency applications. They offer the same input impedance of the oscilloscope which is normally 1 MΩ. However for applications where better accuracy is needed and as frequencies start to rise, other test probes are needed.

To enable better accuracy to be achieved higher levels of impedance are required. To achieve this attenuators are built into the end of the probe that connects with the circuit under test. The most common type of probe with a built in attenuator gives an attenuation of ten, and it is known as a X10 oscilloscope probe. The attenuation enables the impedance presented to the circuit under test to be increased by a factor of ten, and this enables more accurate measurements to be made.

As the X10 probe attenuates the signal by a factor of ten, the signal entering the scope itself will be reduced. This has to be taken into account. Some oscilloscopes automatically adjust the scales according to the probe present, although not all are able to do this. It is worth checking before making a reading.

The 10X scope probe uses a series resistor (9 M Ohms) to provide a 10 : 1 attenuation when it is used with the 1 M Ohm input impedance of the scope itself. A 1 M Ohm impedance is the standard impedance used for oscilloscope inputs and therefore this enables scope probes to be interchanged between oscilloscopes of different manufacturers.



Oscilloscope probe circuit

The scope probe circuit shown is a typical one that might be seen - other variants with the variable compensation capacitor at the tip are just as common.

In addition to the X1 and X10 scope probes, X100 probes are also available. These oscilloscope probes tend to be used where very low levels of circuit loading are required, and where the high frequencies are present. The difficulty using the is the fact that the signal is attenuated by a factor of 100.

X10 oscilloscope probe compensation

The X10 scope probe is effectively an attenuator and this enables it to load the circuit under test far less. It does this by decreasing he resistive and capacitive loading on the circuit. It also has a much higher bandwidth than a traditional X1 scope probe.

The x10 scope probe achieve a better high frequency response than a normal X1 probe for a variety of reasons. It does this by decreasing the resistive and capacitive loading on the The X10 probe can often be adjusted, or compensated, to improve the frequency response.

Typical oscilloscope probe

For many scope probes there is a single adjustment to provide the probe compensation, although there can be two on some probes, one for the LF compensation and the other for the HF compensation.

Probes that have only one adjustment, it is the LF compensation that is adjusted, sometimes the HF compensation may be adjusted in the factory.

To achieve the correct compensation the probe is connected to a square wave generator in the scope and the compensation trimmer is adjusted for the required response - a square wave.

Compensation adjustment waveforms for X10 oscilloscope probe.

As can be seen, the adjustment is quite obvious and it is quick and easy to undertake. It should be done each time the probe is moved from one input to another, or one scope to another. It does not hurt to check it from time to time, even if it remains on the same input. As in most laboratories, things get borrowed and a different probe may be returned, etc . .

A note of caution: many oscilloscope probes include a X1/X10 switch. This is convenient, but it must be understood that the resistive and capacitive load on the circuit increase significantly in the X1 position. It should also be remembered that the compensation capacitor has no effect when used in this position.

As an example of the type of loading levels presented, a typical scope probe may present a load resistance of 10MΩ along with a load capacitance of 15pF to the circuit in the X10 position. For the

X1 position the probe may have a capacitance of possibly 50pF plus the scope input capacitance. This may end up being of the order of 70 to 80pF.

Other types of probe

Apart from the standard 1X and 10X voltage probes a number of other types of scope probe are available.

- ***Current probes:*** It is sometimes necessary to measure current waveforms on an oscilloscope. This can be achieved using a current probe. This has a probe that clips around the wire and enables the current to be sensed. Sometimes using the maths functions on a scope along with a voltage measurement on another channel it is possible to measure power,
- ***Active probes:*** As frequencies rise, the standard passive probes become less effective. The effect of the capacitance rises and the bandwidth is limited. To overcome these difficulties active probes can be used. They have an amplifier right at the tip of the probe enabling measurements with very low levels of capacitance to be made. Frequencies of several GHz are achievable using active scope probes.
- ***Differential scope probes:*** In some instances it may be necessary to measure differential signals. Low level audio, disk drive signals and many more instances use differential signals and these need to be measured as such. One way of achieving this is to probe both lines of the differential signal using one probe each line as if there were two single ended signals, and then using the oscilloscope to add then differentially (i.e. subtract one from the other) to providethedifference.

  Using two scope probes in this way can give rise to a number of problems. The main one is that single ended measurements of this nature do not give the required rejection of any common mode signals (i.e. Common Mode Rejection Ratio, CMMR) and additional noise is likely to be present. There may be a different cable length on each probe that may lead to a time differences and a slight skewing between the signals.

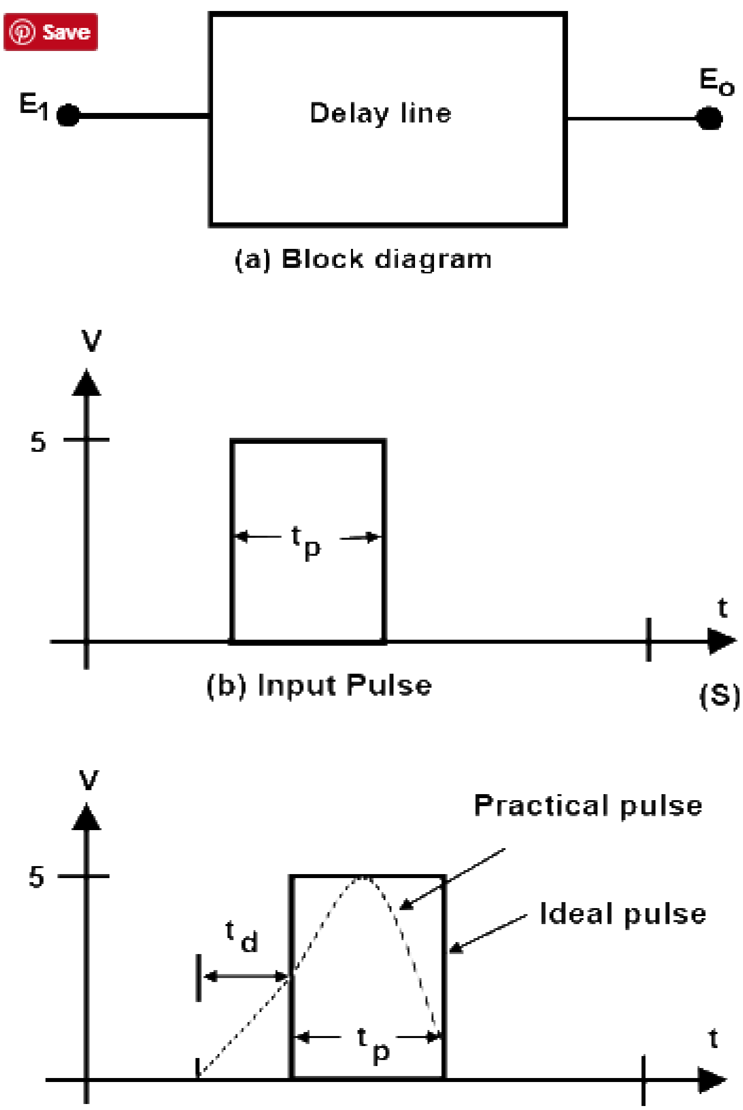
  To overcome this a differential probe may be used. This uses a differential amplifier at the probing point to provide the required differential signal that is then passed along the scope probe lead to the oscilloscope itself. This approach provides a far higher level of performance.

- ***High voltage probes:*** Most standard oscilloscope voltage probes like the X1 or X10 are only specified for operation up to voltages of a few hundred volts at most. For operation higher than this a proper high voltage probe with specially insulated probe is required. It also will step down the voltage for the input to the scope so that the test instrument is not damaged by the high voltage. Often voltage probes may be X50 or X100.

Delay Lines:

In many pulse circuit applications, it is necessary to delay a signal a certain amount of time called delay time. Delay circuits may be classified in two main groups: electronic circuits (e.g. mono stable multi vibrator and blocking oscillator etc.) and electromagnetic delay lines.

The electronic circuit. operate on the principle of being in a stable until triggered by a pulse. Then they shift to a quasi stable state for a period of time determined usually by the time constant or RC circuit. After this period of time, the circuit returns against to its stable state until triggered again by another pulse. If some means is provided to have a pulse generated at a time the circuit returns to its stable state, a time delay circuit results. The output pulse is not necessarily an exact reproduction of the input pulse. Its amplitude and width are dependent upon the circuit producing it and not on the amplitude and width of the input pulse.



(a) Block diagram

(b) Input Pulse

(C) Delay output pulse

A delay line is required to delay a pulse for predetermined period of time constant distorting it. The block input to it is shown in figure 1 (a). The pulse input to it is shown in the figure 1 (b). The output is shown in figure 1 (c) observed that the delay line simply delays to input pulse by time t_d as shown in figure 1 (c).

Types of Delay Lines

There are two types of delay lines used in switching circuits the electronic types and electromagnetic type.

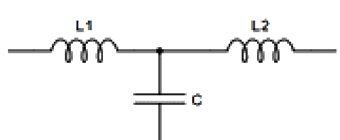
Electromagnetic Delay Lines

Electromagnetic delay lines may be further classified as distributed parameter or lumped parameter delay lines. The distributed parameter line more closely approaches a transmission line, whereas the lumped parameter line resembles a filter. Delay ranging from a few nanoseconds to hundreds of micro-seconds are obtainable with electromagnetic lines. For longer delays acoustic delay lines and storage devices are used. Acoustic delay lines employ acoustic wave propagation and electro-mechanical transducers at the input and output.

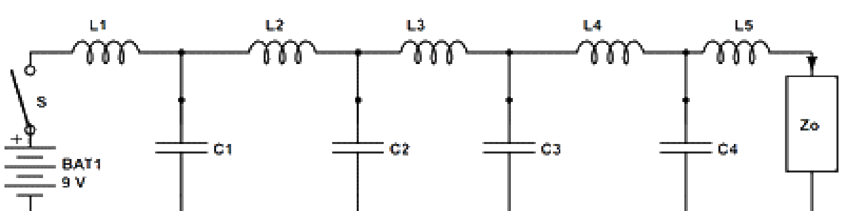
Among the common applications for electromagnetic delay lines are air navigation system such as DME (distance measuring equipment) using electrical information related to time. A cathode ray oscilloscope for observing fast waveforms may also use a electromagnetic delay line to delay the input signal to the vertical plates until the sweep is started. If the delay line were not included, the initial portion of the waveform might not be visible on the scope face. Other applications of delay line occur in distributed amplifiers, in pulse coders and decoders, in precise time measurement, in television and in digital computer systems.

Lumped Parameter Delay Lines

In its simplest form the lumped parameter consist of a number of inductors and capacitors are similar in value. The inductors are connected in series and the capacitors are connected through the junction between the inductors to the ground. Lumped parameter delay lines can be tapped at several points to give a series of delay and this property is useful i many applications. The products are of the order of 0.01 to 100,000 µ sec. In addition, lumped parameter delay lines are used as pulse forming networks (PFN) is such applications as radar modulators.



(a) Equivalent Circuit



(b) Four delay line sections

Figure 2 shows an equivalent circuit of the loss less transmission line. In such a circuit C is inter-conductor capacitance and L is the total equivalent inductance figure 2 shows the transmission lie divided into several equal lengths. A step input applied at V_i takes a certain finite time t_d to read the load Z_0 t_d is called the delay time.

Let us consider the circuit of figure 2 as the step input is applied, capacitor C_1start charging through (L_1/2). When C_1 gets completely charged, C_2 starts charging. After C_2 has charged C_3 starts charging, and so on. Hence the pulse reaches Z_0 after all the capacitors have been charged. The time that the total capacitance required to fully charged is the delay time t_d. Suppose the constant current I_0 is following from the source V_i when capacitors C_1,C_2, C_3 and so on are being charged then.

$I_0 = V_i/Z_i$

Where I_0 is the constant current that flows from the input source which capacitance is being charged. E_i is the amplitude of the step voltage V and Z_i is the input impedance of the lumped parameter delay lines measured in ohm.

If the load Z_i become equal to the input impedance Z_i then the load current will also be I_0 after the pulse has reached at the load. Under such condition, there will be a constant current drain equal to I_0 from the source V first charging the capacitance and then flowing through the load. In such a case, Z_0 is called as characteristic impedance of the delay line. It is true that the characteristic impedance Z_0 of transmission line is equal to its input impedance when the line is infinitely ling. It is the property of Z_0 that a delay line is terminated in its characteristics impedance to save from the reflections or line oscillation.

For the lumped parameter delay line shown in figure 2 the characteristic impedance is

$$Z_o = \sqrt{\frac{L}{C}}\ \Omega$$

$$t_d = \sqrt{LC}\,\Omega$$

$$f_c = \left(\frac{1.1}{\pi}\right)\left(\frac{1}{\sqrt{LC}}\right)\ hertz$$

Where Z_0 is the characteristic impedance, L is the inductance, C is the capacitance, t_d is the delay time of f_c is the upper cut off frequency.

The circuit shown in figure 2 (a) is low pass equivalent circuit. It attenuates signals that have frequencies above upper cut off frequency f_c.

Distributed Parameter Delay Lines

A transmission line may be conductive that serves to guide energy from one point (input) to another (output). Normally the transmission lines consists of pair of straight conducting wires or coaxial cables. Any linear conductor possesses both inductive and resistance.

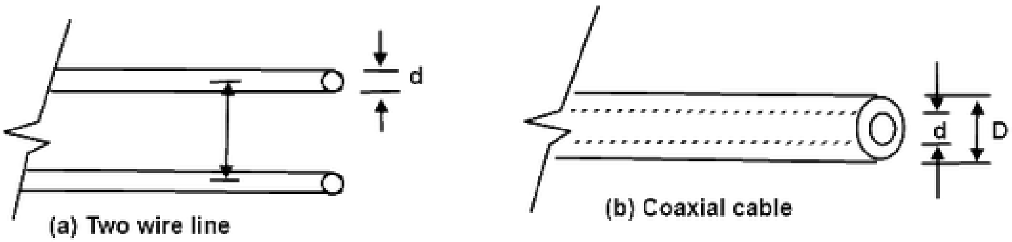


Figure 3: Distributed parameter delay lines

Distributed parameter delay line are shown in figure figure 3. In a two wire transmission line the two wires are separated by a dielectric and therefore, distributed capacitance is also present. Because no capacitor is perfect, a dielectric leakage resistance must also be considered. The equivalent circuit for a short length of line, therefore must include all these elements with resistance R and inductance L in series, and dielectric leakage conductance G and capacitive reactance in shunt. A long line may be considered as made up a number of similar sections distributed along the length of the line. A transmission line is termed loss less or distortion less if

---

R = G = 0

Loss less lines are performed as delay lines because they do not introduce attenuation to the impressed signal. Further infinite lines are simulated by finite lines terminated in its characteristic impedance Z_0.

The characteristics impedance of the line also depends on L and C and is given by:

$$Z_o = \sqrt{\frac{L}{C}} \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots 1$$

The L and C of a transmission line are both functions of the geometry of the cross-section of the line. For example, a two wire transmission line made up of wires diameter d, which are separated by a distance of D (measured between wire centers), will have an inductance and capacitance given by:

$$L = 0.921 \log\frac{2D}{d}\ uh/m \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots 2$$

$$and \quad C = \frac{12.06 \in r}{\log\frac{2D}{d}}\ pF/m \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots 3$$

Where ϵ_r is the relative dielectric constant.

For air ϵ_r = subtracting equation 2 in equation 3 and 4 we have.

$$T_d = \sqrt{LC}$$

$$= \sqrt{\left(9.21 \times 10^{-7} log\frac{2}{d}\right)\left(1.206 \times 10^{-11} \times \frac{\in r}{log\frac{2D}{d}}\right)}$$

$$= 3.33 \times 10^{-9}\sqrt{\in r}\frac{sec}{m} \quad \ldots\ldots\ldots\ldots\ldots 4$$

$$= Z_0 = \sqrt{\frac{L}{C}}$$

$$= \sqrt{\frac{7.64 \times 10^4}{\in r}} log\frac{2}{d} = \frac{276}{\sqrt{\in r}} log\frac{2D}{d}$$

Observe that the time delay is independent of the diameter of the spacing between them. In fact the time delay depends only upon the magnetic permeability (μ) and dielectric constant (ϵ) of the insulating material between the conductors.

Since

$$T = \sqrt{\mu \in}$$

$$\left[Hint.\ \mu = \mu_o; permeability\ of\ air, \quad \in r = \frac{\in}{\in_o}\ and\ \sqrt{\mu_o \in_o} = 3.33 \times 10^{-9}\right]$$

For a coaxial transmission line with linear inner conductor of radius d, outer conductor with diameter D, and the insulating material with relative dielectric constant ϵ_r,

$$L = 0.46 \log\frac{D}{d}\ \mu h/m \ldots\ldots\ldots\ldots\ldots.\ 6$$

$$C = \frac{24.1 \in r}{log\frac{2D}{d}}\ pF/m \ldots\ldots\ldots.7$$

It can be shown that the characteristic impedance is

$$Z_0 = \frac{138}{\sqrt{\epsilon_r}} \log\frac{D}{d} \ldots\ldots\ldots\ldots.8$$

And time delay is

$$T = 3.33 \times 10^{-9}\sqrt{\in_r}\ \sec/m$$

Uses of Delay Lines

Delay lines are used in distributed amplifiers, cathode ray oscilloscope pulse coder and decoder, measuring instrument, Radar, T.V. system and computers.

Applications of CRO:

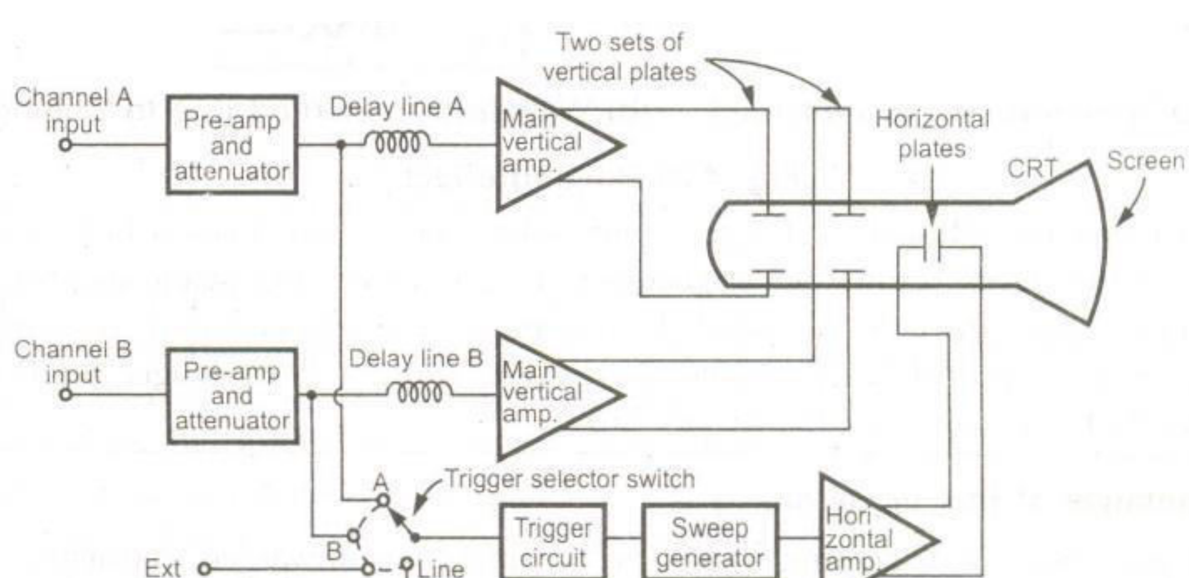
1. **Measurement of voltage** – Voltage waveform will be made on the oscilloscope screen. From the screen of the CRO, the voltage can be measured by seeing its amplitude variation on the screen.
2. **Measurement of current** – Current waveform will be read from the oscilloscope screen in the similar way as told in above point. The peak to peak, maximum current value can be measured from the screen.
3. **Measurement of phase** – Phase measurement in cro can be done by the help of Lissajous pattern figures. Lissajous figures can tell us about the phase difference between two signals. Frequency can also be measured by this pattern figure.
4. **Measurement of frequency** – Frequency measurement in cathode ray oscilloscope can be made with the help of measuring the time period of the signal to be measured.

Special Purpose Oscilloscopes:

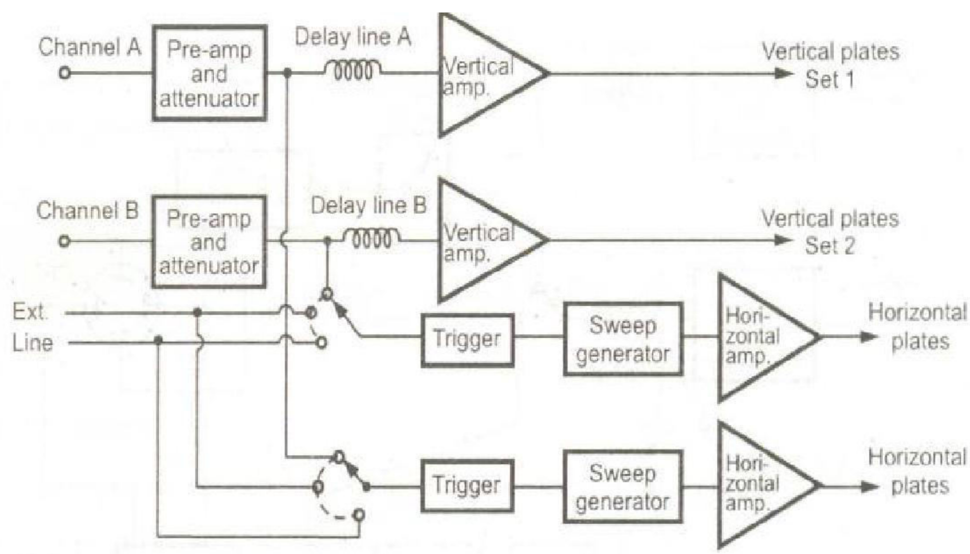
Dual Beam Oscilloscope:

Another method of studying two voltages simultaneously on the screen is to u special cathode ray tube having two separate electron guns generating two separate beami Each electron beam has its own vertical deflection plates.

But the two beams are deflected horizontally by the common set of horizontal plate\ The time base circuit may be same or different. Such an oscilloscope is called **Dual** Beam Oscilloscope.



The oscilloscope has two vertical deflection plates and two separate channels A and B for the two separate input signals. Each channel consists of a preamplifier and an attenuator. A delay line, main vertical amplifier and a set of vertical deflection plates together forms a single channel. There is a single set of horizontal plates and single time base circuit. The sweep generator drives the horizontal amplifier which inturn drives the plates. The' horizontal plates sweep both the beams across the screen at the same rate. The sweep generator can be triggered internally by the channel A signal or .channel B signal. Similarly it' can also be triggered from an external signal or line frequency signal. This is possible with the help of trigger selector switch, a front panel control. Such an oscilloscope may have separate timebase circuit for separate channel. This allows different sweep rates for the two channels but increases the size and weight of the oscilloscope.

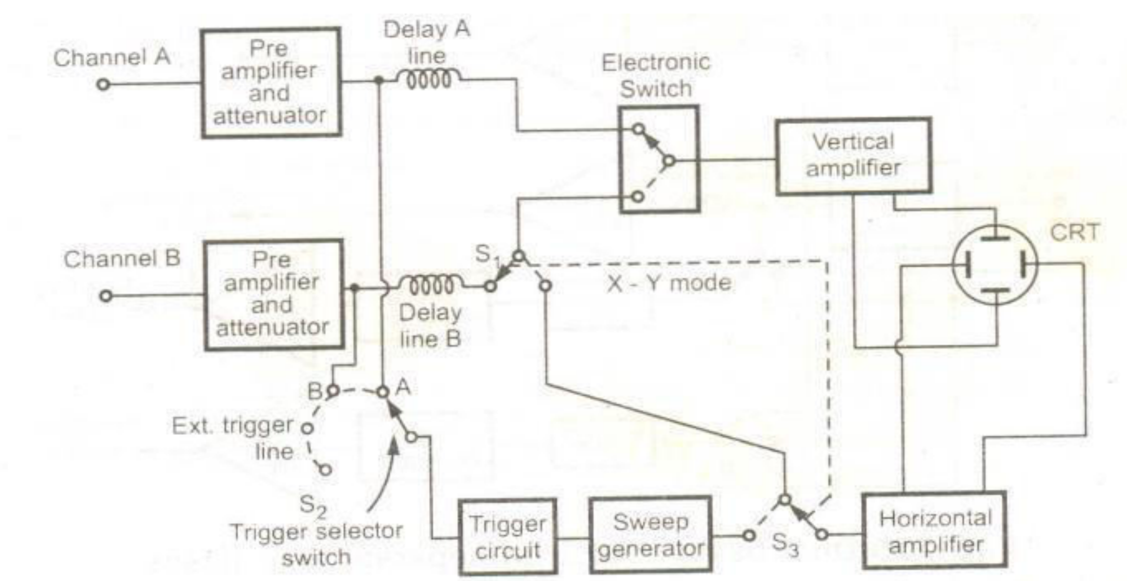


Dual beam CRO with separate time bases

Dual trace oscilloscope:

The comparison of two or more voltages is very much ,necessary in the analysis and study of many electronic circuits and systems. This is possible by using more than one oscilloscope but in such a case it is difficult to trigger the sweep of each oscilloscope precisely at the same time. A common and less costly method to solve this problem is to use dual trace or multitrace oscilloscopes. In this method, the same electron beam is used to generate two traces which can be deflected from two independent vertical sources. The methods are used to generate two independent traces which the alternate sweep method and other is chop method.
The block diagram of dual trace oscilloscope is shown in the Fig

There are two separate vertical input channels A and B. A separate preamplifier and -attenuator stage exists for each channel. Hence amplitude of each input can be individually controlled. After preamplifier stage, both the signals are fed to an electronic switch. The switch has an ability to pass one channel at a time via delay line to the vertical amplifier. The time base circuit uses a trigger selector switch 52 which allows the circuit to be triggered on either A or B channel, on line frequency or on an external signal. The horizontal amplifier is fed from the sweep generator or the B channel via switch 5! and 51. The X-Y mode means, the oscilloscope operates from channel A as the vertical signal and the channel B as the horizontal signal. Thus in this mode very accurate X-Y measurements can be done.



Sampling Time Base:

The time base circuit of the sampling oscilloscope is different than the conventional oscilloscope. The time base of sampling oscilloscope has two functions:

i) To move the dots across the screen

ii) To generate the sampling command pulses for the sampling circuit.

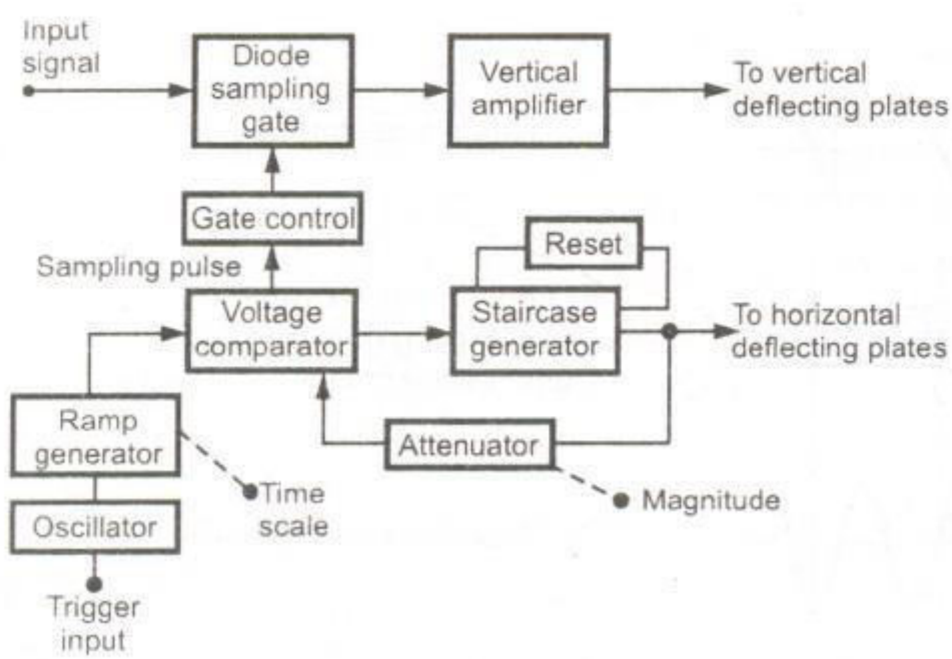
It consists of synchronous circuit, which determines the sampling rate and establishes a reference point in time with respect to the input signal. The time base generates a triggering pulse which activates the oscillator to generate a ramp voltage. Similarly it generates a stair case waveform. The ramp generation is based on the output of the synchronizing circuit.

Both the ramp as well as staircase waveforms are applied to a voltage comparator. This comparator compares the two voltages and whenever these two voltages are equal, it generates a samppng pulse. This pulse then momentarily bias the diodes of the sampling gate in the forward direction and thus diode switch gets closed for short duration of time.

The capacitor charges but for short time hence, it can charge to only a small percentage of the input signal value at that instant. This voltage is amplified by the vertical amplifier and then applied to the vertical deflecting plates. This is nothing but a sample. At the same time, the comparator gives a signal to the staircase generator to advance through one step. This is applied to horizontal deflecting plates, thus during each step of the stair case waveform, the spot moves across the screen. Thus the sampling time base is called a staircase-ramp generator in case of a sampling oscillosope.

Block diagram of Sampling Oscilloscope:

The block diagram of sampling oscilloscope is shown in the Fig.



The input signal is applied to the diode sampling gate. At the start of each sampling cycle a trigger inpu t pulse is generated which activates the blocking oscillator. The oscillator output is given to the ramp generator which generates the linear ramp signal. Since the sampling must be synchronized with the input signal freq\,lency, the signal is delayed in the vertical amplifier.

The staircase generator produces a staircase waveform which is applied to an attenuator. The attenuator controls the magnitude of the staircase signal and then it is applied to a voltage comparator. Another input to the voltage comparator is the output of the ramp generator. The voltage comparator compares the two signals and produces the output pulse when the two voltages are equal. This is nothing but a sampling pulse which is applied to sampling gate through the gate control circuitry.

This pulse opens the diode gate and sample is taken in. This sampled signal is then applied to the vertical amplifier and the vertical deflecting plates. The output of the staircase generator is also applied to the horizontal deflecting plates.

During each step of staircase the spot moves on the screen. The comparator output advances the staircase output through one step. After certain number of p\llses about thousand or so, the staircase generator resets. The sm,lIler the size of the steps of the staircase generator, larger is the number of samples and higher *is* the resolution of the image.

Analog storage oscilloscope:

The conventional cathode ray tube has the persistence of the phosphor ranging from a Few millisecond to several seconds. But sometimes it is necessary to retain the image for much 'longer periods, upto several hours. It requires storing of a waveform for a certain duration,' independent of phosphor persistence. Such a retention property helps to display the waveforms of very low frequency.

Mainly two types of storage techniques are used in cathode ray tubes which are:

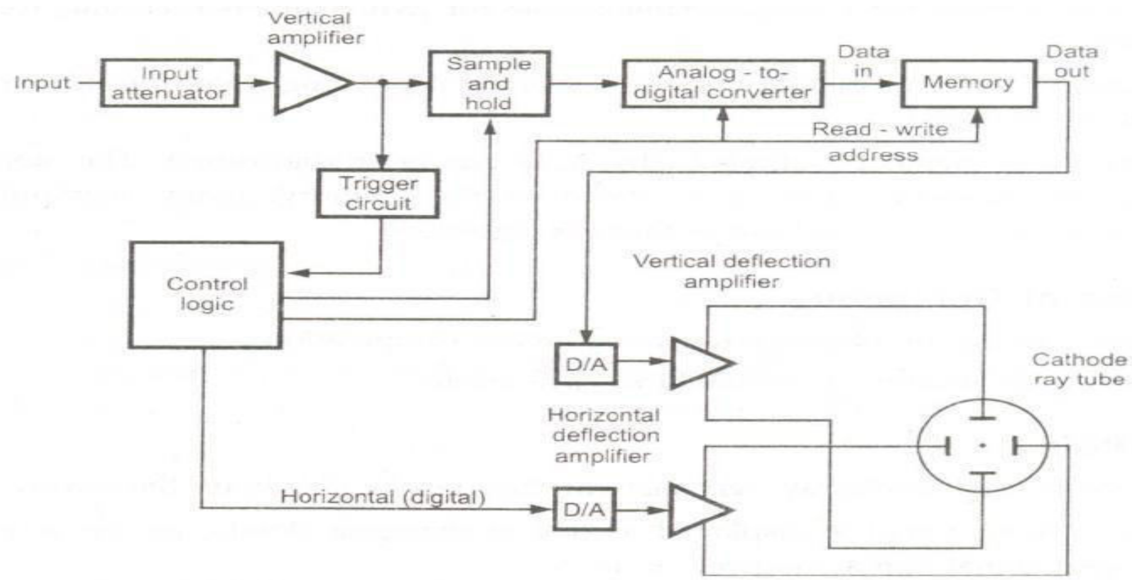
i) Mesh storage and ii) Phosphor storage

Digital Storage Oscilloscope:

In this digital storage oscilloscope, the waveform to be stored is digitised ,md then stored in a digital memory. The conventional cathode ray tube is used in this oscilloscope hencethe cost is less. The power to be applied to memory is small and can be supplied by small battery. Due to this the stored image can be displayed indefinitely as long ,15 power is supplied to memory. Once the waveform is digitised then it can be further loaded into thecomputer and can be ana lysed in detail.

Block Diagram:

The block diagram of digital storage oscilloscope is shown in the Fig.



As done in all the oscilloscopes, the input signal is applied to the amplifier and attenuator section. The oscilloscope uses same type of amplifier and attenuator circuitry as used in the conventional oscilloscopes. The attenuated signal is then applied to the vertical amplifier.

The vertical input, after passing through the vertical amplifier, is digitised by an analog to digital converter to create a data set that is stored in the memory. The data set is processed by the microprocessor and then sent to the display.

To digitise the analog signal, analog to digital *(A/D)* converter is used. The output of the vertical amplifier is applied to the *AID* converter section. The main requirement of *A/D* converter in the digital storage oscilloscope is its speed, while in digital voltmeters accuracy and resolution were the main requirements. The digitised output needed only in the binary form and not in BCD. The successive approximation type of *AID* converter is most oftenly used in the digital storage oscilloscopes.

Modes of operation:

The digital storage oscilloscope has three modes of operation:

1. Roll mode ii) Store mode iii) Hold or save mode.

Roll mode:

This mode is used to display very fast varying signals, clearly on the screen. The fast varying signal is displayed as if it is changing slowly, on the screen. In this mode, the input signal is not triggered at all.