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SCHOOL OF ELECTRICAL AND ELECTRONICS

DEPARTMENT OF ELECTRONICS AND INSTRUMENTATION

UNIT – I

ELECTRONIC MEASUREMENT AND INSTRUMENTATION – SIC 1305

MEASURING INSTRUMENTS AND WAVEFORM GENERATORS

I REVIEW OF AC AND DC VOLTMETER AND AMMETER

The action of the most common dc meter is based on the fundamental principle of the motor. The motor action is produced by the flow of a small current through a moving coil, which is positioned in the field of a permanent magnet. This basic moving coil system is called the D'Arsonval galvanometer.

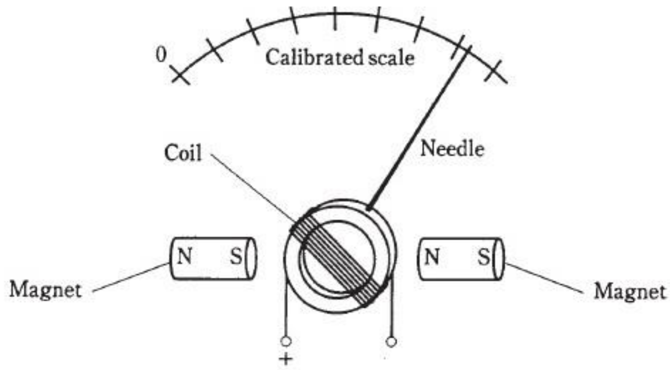


Fig. 1: The D'Arsonval Principle.

It employs a spring-loaded coil through which the measured current flows. The coil (rotor) is in nearly homogeneous field of permanent magnet and moves in rotary fashion. The amount of rotation is proportional to the amount of current flowing through the coil. A pointer attached to the coil indicates the position of the coil on a scale calibrated in terms of current or voltage. It responds to dc current only, and has an almost linear calibration. The magnetic shunt that varies the field strength is used for calibration.

The construction of a D'Arsonval galvanometer is shown below along with brief description of the its parts:

1) The moving coil:

It is the current carrying element. It is either rectangular or circular in shape and consists of number of turns of fine wire. This coil is suspended so that it is free to turn about its vertical axis of symmetry. It is arranged in a uniform, radial, horizontal magnetic field in the air gap between pole pieces of a permanent magnet and iron core. The iron core is spherical in shape if the coil is circular but is cylindrical if the coil is rectangular. The iron core is used to provide a flux path of low reluctance and therefore to provide strong magnetic field for the coil to move in. this increases the deflecting torque and hence the sensitivity of the galvanometer. The length of air gap is about 1.5mm. In some galvanometers the iron core is omitted resulting in of decreased value of flux density and the coil is made narrower to decrease the air gap. Such a galvanometer is less sensitive, but its moment of inertia is smaller on account of its reduced radius and consequently a short periodic time.

2) Damping:

There is a damping torque present owing to production of eddy currents in the metal former on which the coil is mounted. Damping is also obtained by connecting a low resistance across the galvanometer terminals. Damping torque depends upon the resistance and we can obtain critical damping by adjusting the value of resistance.

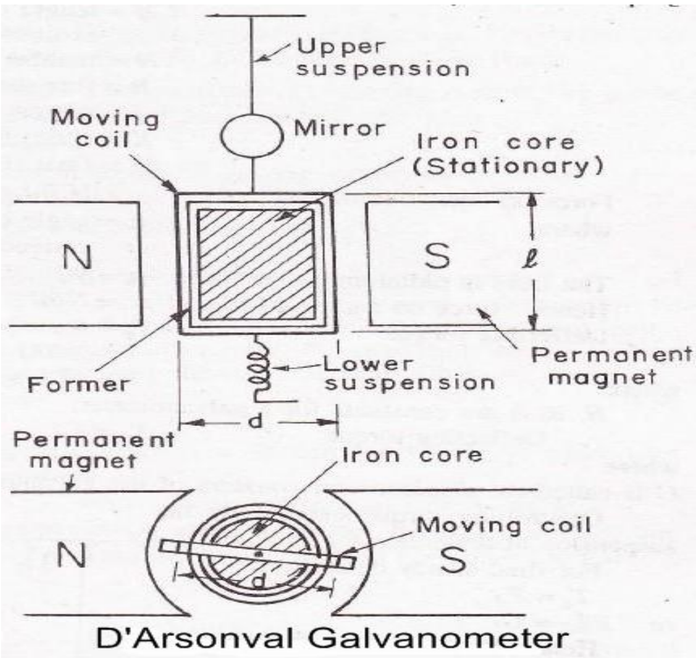


Fig. 2 A D'Arsonval Galvanometer.

3) Suspension:

The coil is supported by a flat ribbon suspension which also carries current to the coil. The other current connection in a sensitive galvanometer is a coiled wire. This is called the lower suspension and has a negligible torque effect. This type of galvanometer must be leveled carefully so that the coil hangs straight and centrally without rubbing the poles or the soft iron cylinder. Some portable galvanometers which do not require exact leveling have " taut suspensions" consisting of straight flat strips kept under tension for at the both top and at the bottom.

The upper suspension consists of gold or copper wire of nearly (0.012-5)mm or (0.02-5) mm diameter rolled into the form of a ribbon. This is not very strong mechanically; so that the galvanometers must he handled carefully without jerks. Sensitive galvanometers are provided with coil clamps to the strain from suspension, while the galvanometer is being moved.

4) Indication:

The suspension carries a small mirror upon which a beam of light is cast. The beam of light is reflected on a scale upon which the deflection is measured. This scale is usually about 1 meter away from the instrument, although ½ meter may be used for greater compactness.

5) Zero Setting:

A torsion head is provided for adjusting the position of the coil and also for zero setting.

Advantages:

1)They can be modified with the help of shunts and resistance to cover a wide range of currents and voltages.

2)They display no hysteresis.

3)Since operating fields of such instruments are very strong, they are not significantly affected by stray magnetic fields.

Disadvantages:

1) Some errors may set in due to ageing of control springs and the permanent magnet.

2)Friction due to jewel-pivot suspension.

1.1 DC AMMETER

A PMMC galvanometer constitutes the basic movement of a dc ammeter. Since the coil winding of a basic movement is small and light it can carry only very small currents. When large currents are to be measured, it is needed to bypass major part of the current via a resistance called as shunt as shown below:

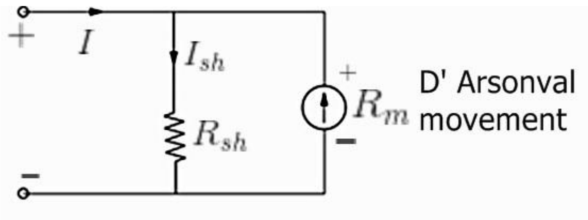


Fig. 3 Basic DC ammeter.

Since the shunt resistance is in parallel with the meter movement, the voltage drop across the shunt and movement must be the same.

Therefore, $V_{sh}=V_m$

ie., $I_{sh}.R_{sh}=I_m.R_m$,

$$R_{sh}=\frac{Im.Rm}{\text{Ish}}$$

For each required value of full scale meter current, we can determine the value of shunt resistance.

1.1.1 Multirange Ammeters:

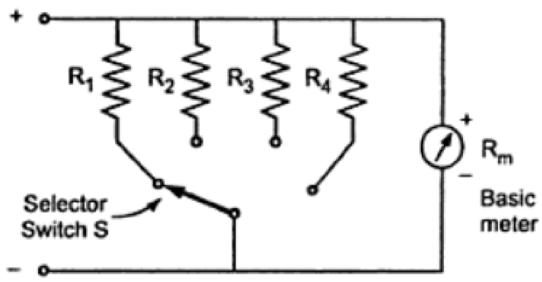


Fig. 4 Multirange Ammeters

The current range of the dc ammeter may be further extended by a number of shunts selected by a range switch. Such a meter is called multirange ammeter. It is shown above.

The circuit has four shunts R_1, R_2, R_3, R_4 which can be placed in parallel with the movement to give four different current ranges. Switch 'S' is a multiposition switch having low contact resistance and high current carrying capacity since its contacts are in series with low resistance shunts.

Shunts are used for the extension of range of Ammeters. So a good shunt should have the following properties:-
1- The temperature coefficient of shunt should be low
2- Resistance of shunt should not vary with time
3- They should carry current without excessive temperature rise
4- They should have thermal electromotive force with copper
'Manganin' is used for DC shunt and 'Constantan' as AC shunt.

1.1.2Extending Ammeter ranges:

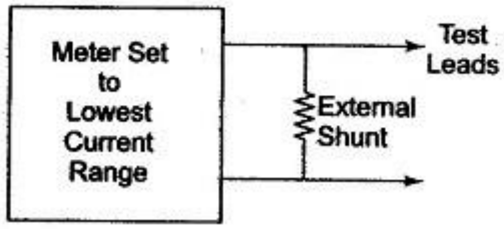


Fig.5 Extending of Ammeters

The range of the ammeter can be extended to measure high current values by using external shunts connected to the basic meter movement (usually the lowest current range) as given in the figure above. Note that the range of the basic meter movement canned be lowered. (for example if a 100μA movement with 100 scale division is used to measure 1μA, then the meter will deflect by only one division. Hence ranges lower than the basic range are not practically possible)

1.2 AC Ammeter

An ideal ammeter senses the current flowing through it while maintaining zero voltage across its terminals. This implies that the meter must have zero internal resistance and behaves like a short circuit when connected to the circuit under test.

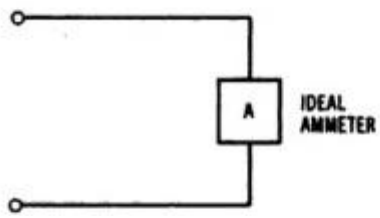


Fig.6 Ideal ammeter has zero internal resistance

A more realistic model of a real ammeter includes the internal resistance of the meter. If the internal resistance of the circuit is less compared with the circuit under test, then there will be minimal effect on the circuit being measured and the meter will approximate the ideal case.

An AC ammeter is similar to DC ammeter except that it measures the current of AC (not DC) waveforms. The same ideal and real (non-zero resistance) ammeter applies to this case of AC ammeter. All of the considerations discussed under AC voltmeters apply to AC ammeters as well.

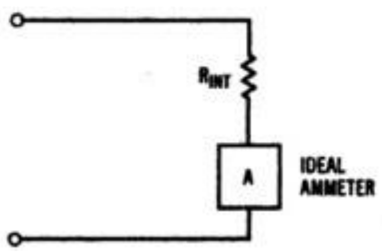


Fig.7 A realistic ammeter has a non-zero internal resistance

1.3 DC Voltmeter

The basic PMMC meter can be converted into a DC voltmeter by adding a series resistor known as multiplier as shown below:

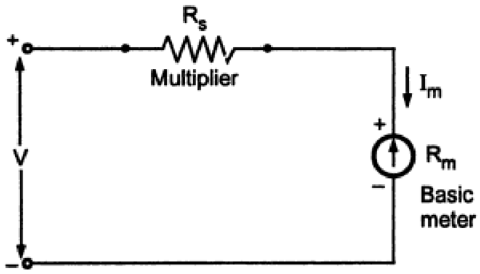


Fig.8 Basic DC voltmeter

The function of the series resistor is to limit the current through the movement so that the current does not exceed the full scale deflection value. A dc voltmeter measures the potential difference between two points in a dc circuit or a circuit component. To measure the potential difference between two points in a dc circuit or a circuit component, dc voltmeter is always connected across them with proper polarity.

The value of the multiplier required is calculated as follows.

I_m=full scale deflection current of the movement (I_{fsd})

R_m=Internal resistance of the movement

R_s=multiplier resistance

V=full range voltage of the instrument.

from the circuit, $V=I_m(R_s+R_m)$

$$R_s=(V/I_m)-Rm.$$

The multiplier limits the currents through the movement, so as to not exceed the value of the full scale deflection I_{fsd}.. The above equation is used to further extend the range of DC voltmeter.

1.3.1 Multirange Voltmeter:

As in case of an ammeter. to obtain a multirange ammeter, a number of shunts were connected across the movement with a multi-position switch. Likewise a dc voltmeter can be converted into a multirange voltmeter by joining a number of resistors (multipliers) along with a range switch to provide a greater number of workable ranges.

Figure shows the multirange voltmeter using a four position switch.

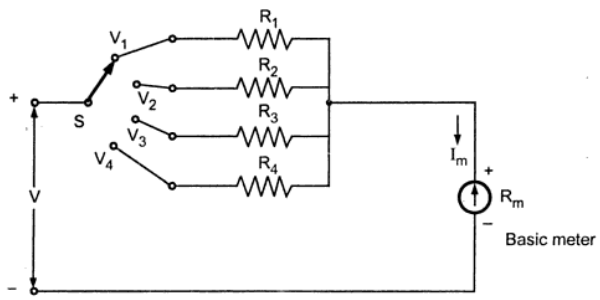


Fig.9 Multirange Voltmeter

1.3.2 Extending Voltmeter ranges:

The range of a voltmeter can be extended to measure high voltages by using a high voltage probe or by using an external multiplier resistor as shown below:

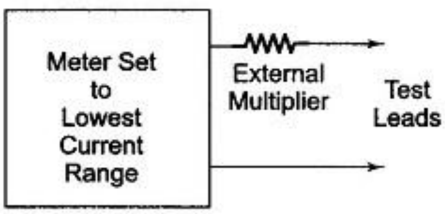


Fig.10 Extending Voltmeter ranges.

In most meters the basic movement is used on the lowest current range.

1.4 AC VOLTMETER

In electronic ac voltmeters input signal is firstly rectified and then supplied to the dc amplifier, as shown in figure. Sometimes signal is firstly amplified by ac amplifier and then rectified before supplying it to dc meter, as shown in figure.

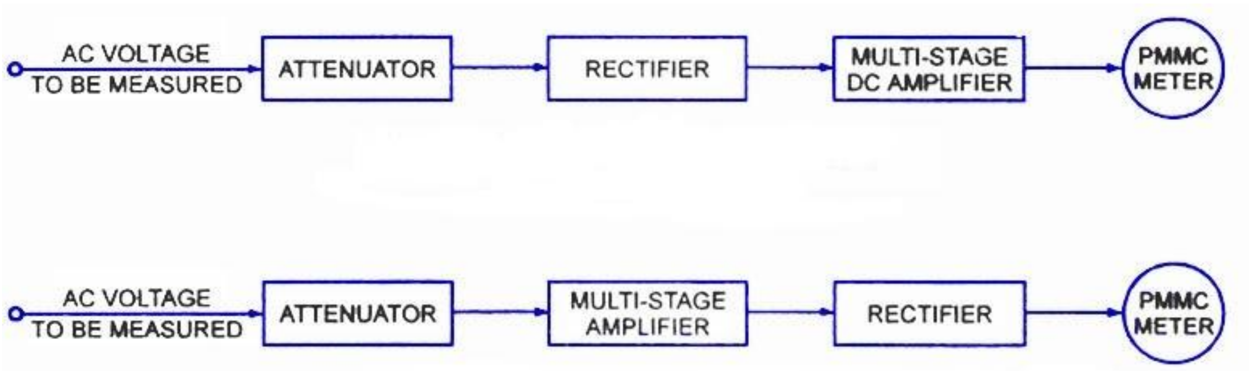


Fig.11 AC Voltmeters

Normally ac voltmeters are average responding type and the meter is calibrated in terms of the rms values for a sine wave. Since most of the voltage measurements involve sinusoidal waveform so this method of measuring rms value of ac voltages works satisfactorily and is less expensive than true rms responding voltmeters. However, in case of measurement of non-sinusoidal waveform voltage, this meter will give high or low reading depending on the form factor of the waveform of the voltage to be measured.

Circuit diagram of an average reading ac voltmeter using semi-conductor diode is shown in figure. The diode conducts during the positive half cycle and does not conduct during the -ve half cycle, as shown in figure.

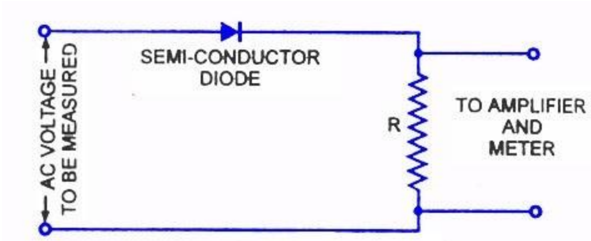


Fig.12 Average reading AC Voltmeter

The average current through the meter will be given by the expression

$I_{av} = V_{av} / 2R = 0.45 * [V_{rms}/R]$

V_{rms} is the effective or rms value of applied voltage and 1.11 is the form factor of sinusoidal wave. R is multiplied by 2 because the voltmeter operates on half-wave rectification. It is to be worthnoting that this instrument can be used to indicate dc voltages but in such a case the instrument readings will have to be multiplied by 2 x 1.11, that is, as the diode conducts all the time. Main advantages associated with these voltmeters are that they are simple in construction, have high input impedance, low power consumption and uniform scale. Main disadvantage of these voltmeters is that these operate in audio-frequency range. In radio-frequency range, distributed capacitance of the high resistance R introduces error in the readings.

Another disadvantage of such a voltmeter is that due to non-linear volt-ampere characteristic for lower voltage the readings of the voltmeter at lower voltage are not correct.

1.4.1 Multirange AC Voltmeter:

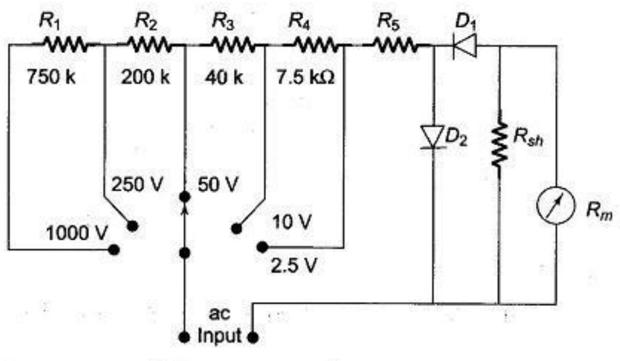


Fig.13 Multirange AC Voltmeters

Resistances R_1, R_2, R_3, R_4 form a chain of multipliers for voltage ranges of 1000V, 250V, 50V and 10V respectively.

On the 2.5V range, resistance R_5 acts as a multiplier and corresponds to the multiplier Rs as shown in the figure. R_{sh} is the meter shunt and acts to improve the rectifier operation.

1.5 Ohmmeter

The ohmmeter means that it is an instrument which measures resistance of a quantity. Resistance in the electrical sense means the opposition offered by a substance to the current flow in the device. Every device has a resistance, it may be large or small and it increases with temperature for conductors, however for semiconducting devices the reverse is true. There are many types of ohmmeters available such as

1. Series ohmmeter.
2. Shunt ohmmeter.
3. Multi range ohmmeter.

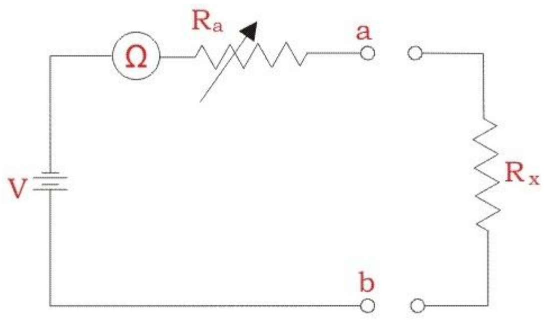


Fig.14 Ohmmeter

The instrument is connected with a battery, a series adjustable resistor and an instrument which gives the reading. The resistance to be measured is connected at terminal ab. When the circuit is completed by connecting output resistance, the circuit current flows and so the deflection is measured.
When the resistance to be measured is very high then current in the circuit will be very small

and the reading of that instrument is assumed to be maximum resistance to be measured. When resistance to be measured is zero then the instrument reading is set to zero position which gives zero resistance.

1.5.1 Series type Ohmmeter:

The series type ohmmeter consists of a current limiting resistor R_1, Zero adjusting resistor R_2, EMF source E, Internal resistance of D'Arsonval movement R_m and the resistance to be measured R. When there is no resistance to be measured, current drawn by the circuit will be maximum and the meter will show a deflection. By adjusting R_2 the meter is adjusted to a full scale current value since the resistance will be zero at that time. The co-responding pointer indication is marked as zero. Again when the terminal AB is opened it provides very high resistance and hence almost zero current will flow through the circuit. In that case the pointer deflection is zero which is marked at very high value for resistance measurement.

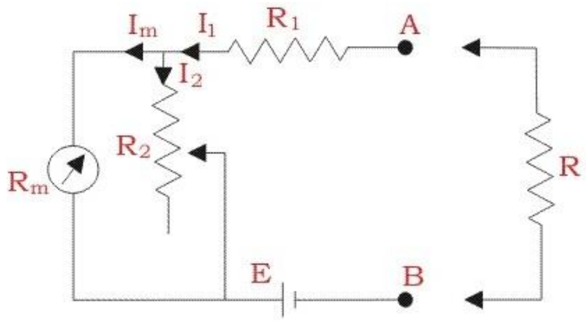


Fig.15 Series type ohmmeter.

So a resistance between zeros to a very high value is marked and hence can be measured. So, when a resistance is to be measured, the current value will be somewhat less than the maximum and the deflection is recorded and accordingly resistance is measured. This method is good but it posses certain limitations such as the decrease in potential of the battery with its use so adjustment must be made for every use. The meter may not read zero when terminals are shorted, these types of problem may arise which is counteracted by the adjustable resistance connected in series with the battery.

1.5.2 Shunt type Ohmmeter:

In this type of meters we have a battery source and an adjustable resistor is connected in series with the source. We have connected the meter in parallel to the resistance which is to be measured. There is a switch by the use of which we can on or off the circuit.

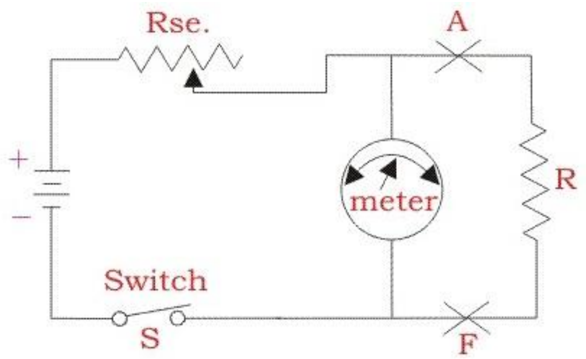


Fig.16 Shunt type Ohmmeter.

The switch is opened when it is not in use. When the resistance to be measured is zero, the terminals A and F are shorted so the current through the meter will be zero. The zero position of the meter denotes the resistance to be zero. When the resistance connected is very high, then a small current will flow through the terminal AF and hence full scale current is allowed to flow through the meter by adjusting the series resistance connected with the battery. So, full scale deflection measures very high resistance. When the resistance to be measured is connected between A and F, The pointer shows a deflection by which we can measure the resistance values. In this case also, the battery problem may arise which can be counteracted by adjusting the resistance. The meter may have some error due to its repeated use also.

1.5.3 Multirange Ohmmeter:

This instrument provides the reading up to a very wide range. In this case we have to select the range switch according to our requirement. An adjuster is provided so that we can adjust the initial reading to be zero. The resistance to be measured is connected in parallel to the meter.

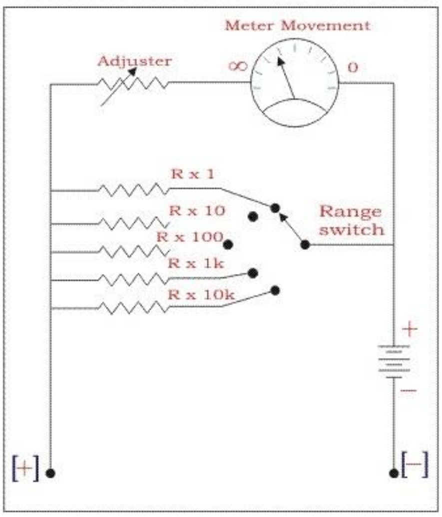


Fig. 17 Multirange Ohmmeter.

The meter is adjusted so that it shows full scale deflection when the terminals in which the resistance connected is full scale range connected through the range switch. When the resistance is zero or short circuit, there is no current flow through the meter and hence no deflection. Suppose we have to measure a resistance under 1 ohm, then the range switch is selected at 1 ohm range at first. Then that resistance is connected in parallel and the corresponding meter deflection is noted. For 1 ohm resistance it shows full scale deflection but for the resistance other that 1 ohm it shows a deflection which is less than the full load value and hence resistance can be measured. This is the most suitable method of all the ohmmeters as we can get accurate reading in this type of meters. So this meter is most widely used now days.

II.ELECTRONIC MULTIMETERS

A multimeter is basically a PMMC meter. To measure dc current the meter acts as an ammeter with a low series resistance. Range changing is accomplished by shunts in such a way that the current passing through the meter does not exceed the maximum rated value. A multimeter consists of an ammeter, voltmeter and ohmmeter combined with a function switch to connect the appropriate circuit to the D'Arsonval movement.

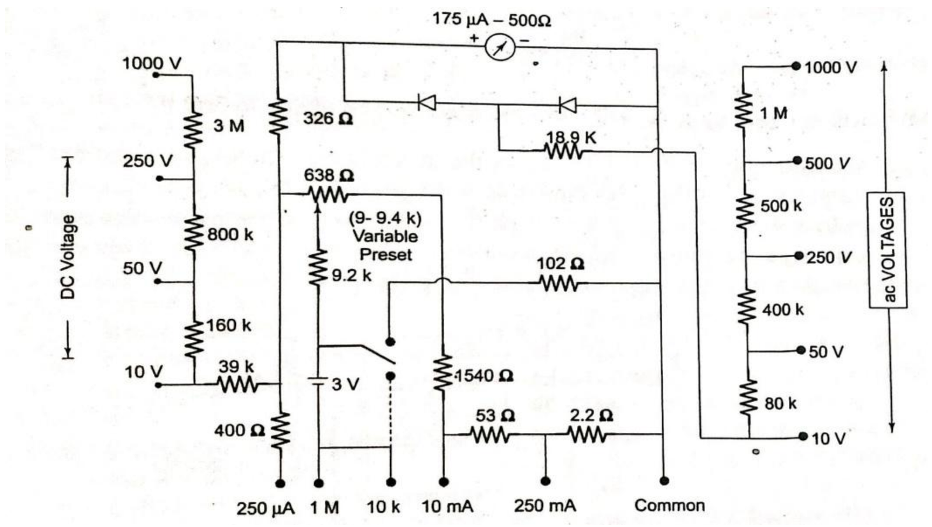


Fig. 18 Diagram of a multimeter

The above figure shows the circuit diagram of a multimeter used as a microammeter, ac voltmeter, dc voltmeter, dc milliammeter and an ohmmeter.

2.1 Microammeter & DC ammeter

Following figure shows the circuit of a multimeter used as a microammeter as well as its use as a dc ammeter.

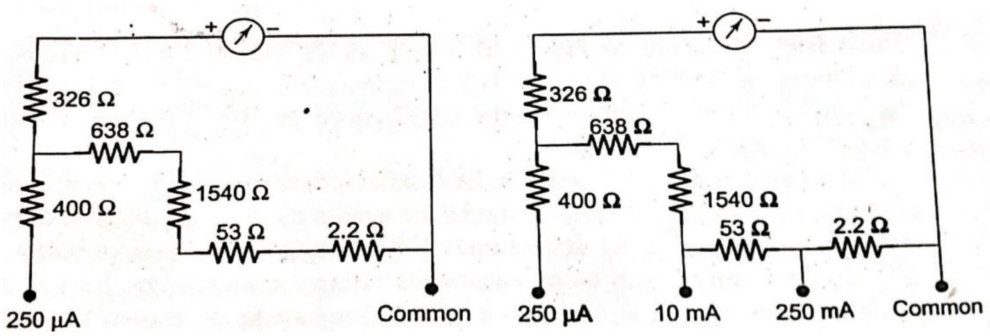


Fig. 19 Multimeter as a microammeter and as a dc ammeter.

2.2 DC Voltmeter:

Following figure shows the use of multimeter as a dc voltmeter.

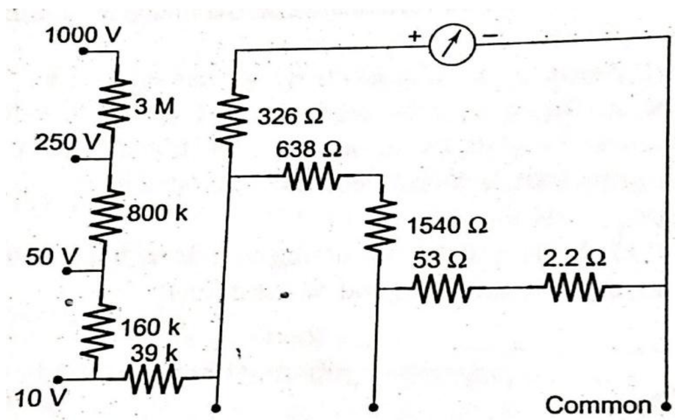


Fig. 20 DC voltmeter section of the multimeter.

2.3 AC Voltmeter:

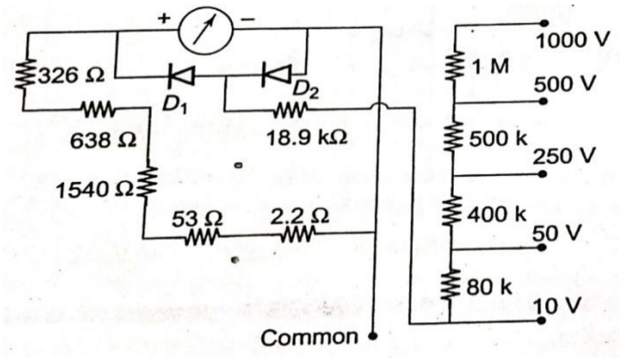


Fig.21 AC voltmeter section of a multimeter.

Figure shows the ac voltmeter section of a multimeter. To measure ac voltage, the output voltage is rectified by a half wave rectifier before the current passes through the meter. Across the meter, the other diode serves as protection. The diode conducts when a reverse voltage appears across the diodes, so that current bypasses the meter in the reverse direction.

2.4 Ohmmeter:

Referring to the following figure, which shows the ohmmeter section of a multimeter, in the 10k range the 102 Ω resistance is connected in parallel with the total circuit resistance and in the 1M Ω range the 102 Ω resistance is totally disconnected from the circuit.

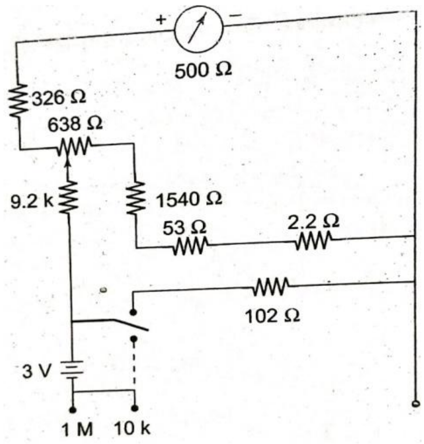


Fig. 22: Ohmmeter section of the multimeter.

Therefore on the 1M Ω range, the half scale deflection is 10k. Since on the 10k range, the 102 Ω resistance is joined across the total resistance, therefore in this range the half scale deflection is 100 Ω. The measurement of resistance is done by applying a small voltage installed within the meter. For the 1M Ω range, the internal resistance is 10k Ω. ie., value at the mid scale.

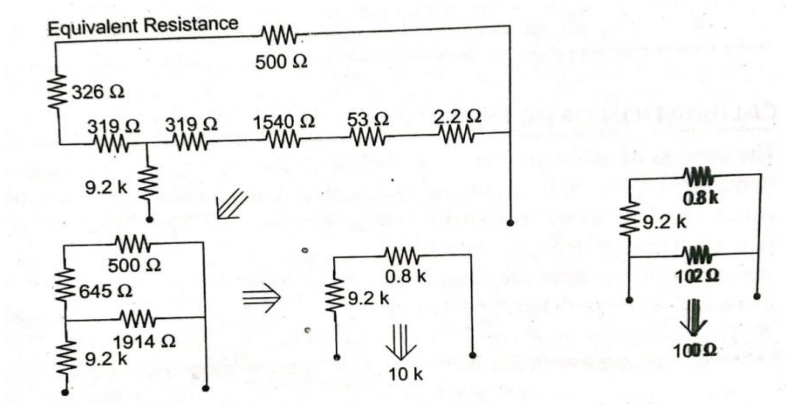


Fig. 23 Equivalent resistance on the 1M Ω range and half scale deflection

The range of the ohmmeter can be changed by connecting the switch to a suitable shunt resistance. By using different values of shunt resistance, different ranges can be obtained. By increasing the battery voltage and using a suitable shunt, the maximum values which the ohmmeter reads can be changed.

III TYPES OF VOLTMETER

3.1 DIFFERENTIAL TYPE VOLTMETER

The differential voltmeter technique is one of the most common and accurate methods of measuring unknown voltages. In this technique, the voltmeter is used to indicate the difference between known and unknown voltages. ie., an unknown voltage is compared with a known voltage.

Following figure shows the basic circuit of a differential voltmeter based on the potentiometric method, hence the name potentiometric voltmeter.

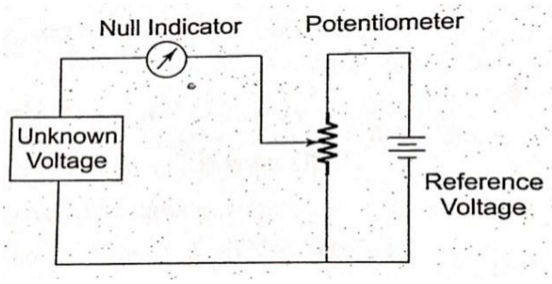


Fig. 24 Basic Differential voltmeter.

In this method, the potentiometer is varied until the voltage across it equals the unknown voltage, which is indicated by the null indicator reading zero. Under null conditions, the meter draws current from neither the reference source nor the unknown voltage source and hence the differential voltmeter represents an infinite impedance to the unknown source (The null meter serves as an indicator only). To detect small differences the meter movement must be sensitive but it need not be calibrated, since only zero has to be indicated. The reference source used is usually a 1V dc standard source or a zener controlled precision supply. A high voltage reference supply is used for measuring high voltages.

The usual practice, however is to employ voltage dividers or attenuators across an unknown source to reduce the voltage. The input voltage divider has a relatively low input impedance , especially for unknown voltages much higher than the reference standard. The attenuation will have a loading effect and the input resistance of voltmeter is not infinity when an attenuator is used.

In order to measure ac voltages, the ac voltages must be converted into dc by incorporating a precision rectifier circuit. A block diagram of such ac differential voltmeter is shown below:

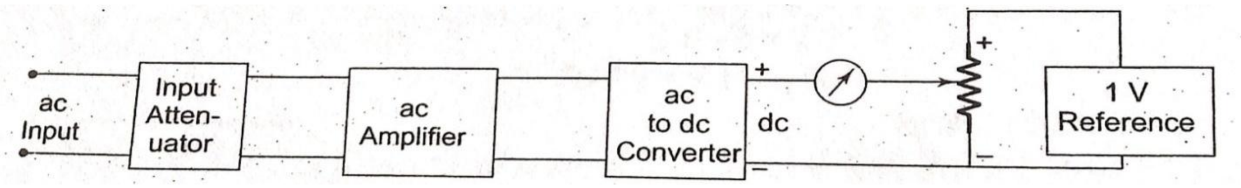


Fig. 25 Block diagram of an ac differential voltmeter.

3.2 True RMS Voltmeter

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.

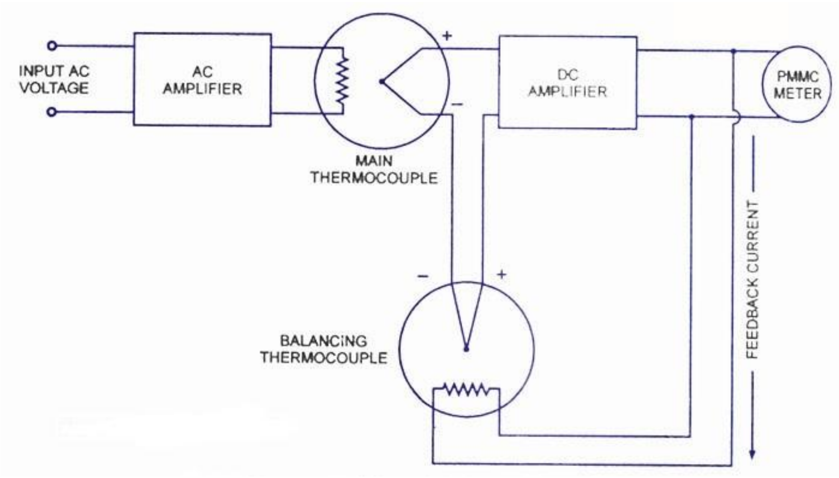


Fig. 26 True RMS Voltmeter

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 thermo-couple 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.

3.3 Vector Voltmeter**

The vector voltmeter is basically a new type of amplitude and phase measuring device. It uses two samplers to sample the two waves whose amplitude and relative phase are to be measured. It measures the voltages at two different points in the circuit and also measures the phase difference between these voltages at these two points. In this voltmeter, two RF signals of the same fundamental frequency (1MHz to GHz) are converted to two IF signals. The amplitudes, waveforms and the phase relations of IF signals are same as that of RF signals. Thus, the fundamental components of the IF signal have the same amplitude and phase relationships as the fundamental components RF signals. These fundamental components are filtered from the IF signals and are measured by a voltmeter and phase meter.

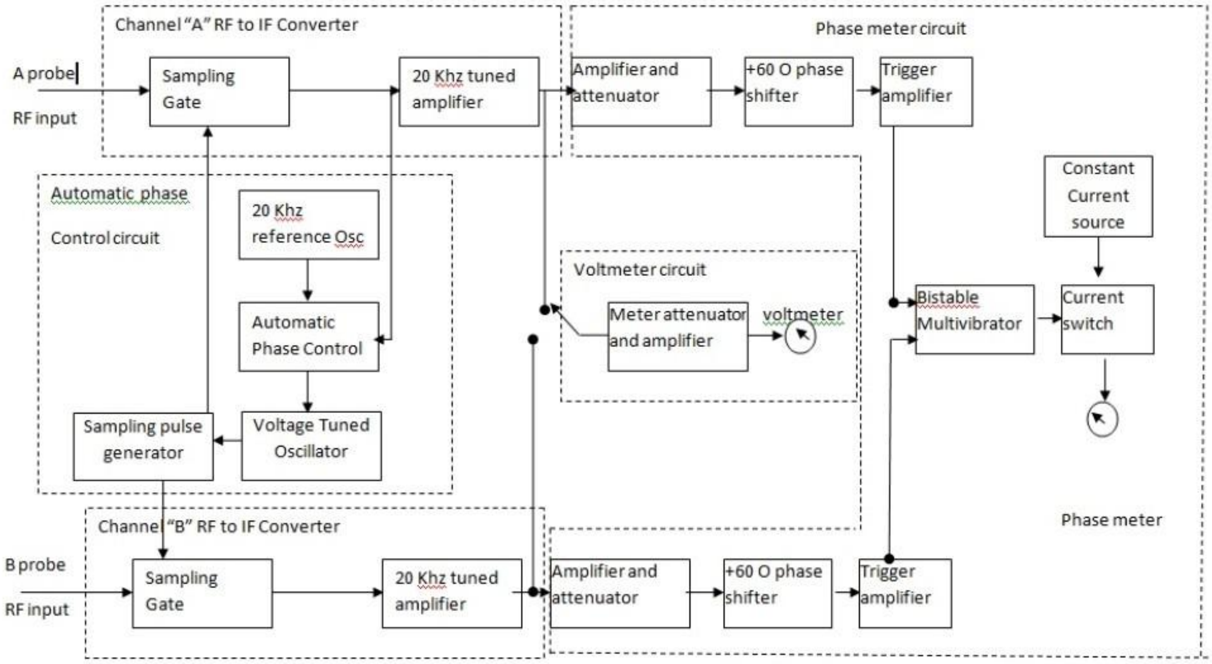


Fig. 27 Block diagram of Vector Voltmeter

The vector voltmeter is useful in VHF applications and are used as :

1)Insertion loses

2)Complex impedance of mixers

3)S parameters of transistors

4) Radio frequency distortion

5) Amplitude modulation index

6) Amplifier gain and phase shift

7) Filter transfer functions

8) Two port network parameters.

The block diagram shows the instrument consists of five major sections. Two IF-RF converters, an automatic phase control section, a phase meter circuit, a voltmeter circuit. The RF-IF converters and the phase control sections produce two 20KHz sine waves with the same amplitudes and phase relationships as the fundamental components of the RF signals applied to channels A and B. The phase meter section continuously monitor these two 20KHz sine waves and indicates the phase angle between them. The voltmeter section can be switched to channel A or channel B to provide a meter display of the amplitude.

Each RF-IF converter consists of a sampler and a tuned amplifier. The sampler produces a 20 KHz replica of the input RF waveform, and the tuned amplifier extracts the 20KHz fundamental component from this waveform replica. The phase control unit is rather a sophisticated circuit that generates the sampling pulses for both RF-IF converters and automatically controls the pulse rate to produce 20KHz IF signals. The sampling pulse rate is controlled by a voltage tuned oscillator (VTO) for which the tuning voltage is supplied by an automatic phase control section. This section locks the IF signal of channel A to a 20KHz reference oscillator. To get initial locking, the phase control section applies a ramp voltage to the VTO. The ramp voltage sweeps the sampling rate until channel A IF is 20KHz and in phase with the reference oscillator.

To determine the phase difference between the two IF signals, the tuned amplifiers are followed by the phase meter circuit. Each channel is first amplified and then limited resulting in square wave signals at the inputs to the IF phase shifting circuits. The circuit in channel A shifts the phase of the square wave signal by $+60^0$ the circuit in channel B shifts the phase of its signal by -120^0 Both phase shifts are accomplished by a combination of capacitive networks and inverting and non-inverting amplifiers whose vector-sum outputs provide the desired phase shift. The outputs of the phase shift circuits are amplified and clipped, producing square waveforms and applied to trigger amplifiers. These circuits convert the square wave input signals to positive spikes with very fast rise times. The bistable multivibrator is triggered by the pulses from both the channels, Channel A is joined to the set input of the MV, channel B is joined to the reset input of the MV. If the initial phase shift between the RF signals at the probes was 0^0, the trigger pulses into the multivibrator are 180^0 out of phase owing to the action of the phase shift circuits. The MV then produces a square wave output voltage which is symmetrical about zero.

The instrument contains a power supply section, which is not shown in the block diagram. The power supply generates all the needed supply voltages for the various sections of the instrument.

IV.SIGNAL GENERATORS

4.1 Audio generator

The main requirement of sine wave signal generator in instrumentation and measurement system is amplitude stability and frequency stability. The audio frequency signal generator uses RC network for controlled phase shift.

In the wein bridge oscillator circuit shown below, the bridge components are R_1, R_2, R_3, C_1, C_2. The operational amplifier together with R_3 and R_4 forms a non-inverting amplifier and R_1-R_2, C_1-C_2 forms the feedback network.

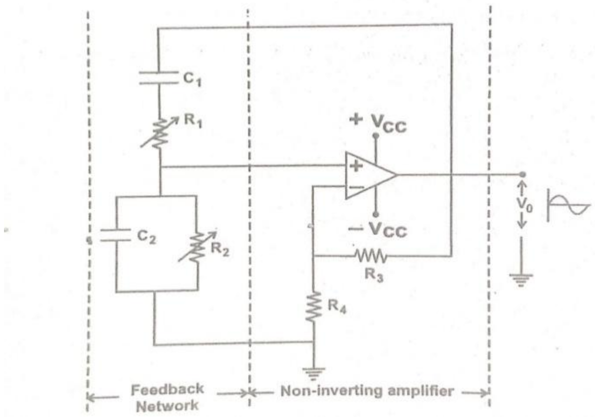


Fig. 28: Wein bridge oscillator

Analysis of the bridge shows the balance condition:

$$\frac{R_3}{R_4} = \frac{R_1}{R_2} + \frac{C_2}{C_1}\,;\ \frac{R_3}{R_4} = 2$$

$$f = \frac{1}{2\pi\sqrt{R_1 R_2 C_1 C_2}}$$

$$R_1 = R_2 = R\,;\ C_1 = C_2 = C$$

$$\boxed{f = \frac{1}{2\pi RC}}$$

The wein bridge oscillator can be tuned by varying either the resistances or capacitances or both. Generally only capacitor tuning is done for variable frequency of signal in one range of frequency. The positive is given by two RC network as shown in above circuit. This creates oscillations.

Advantages:

1) Stable and simple operation

2) Low distortion

3) Good amplitude stability.

4) Relatively easily achievable audio frequency variation.

Audio frequency sine and square wave generator:

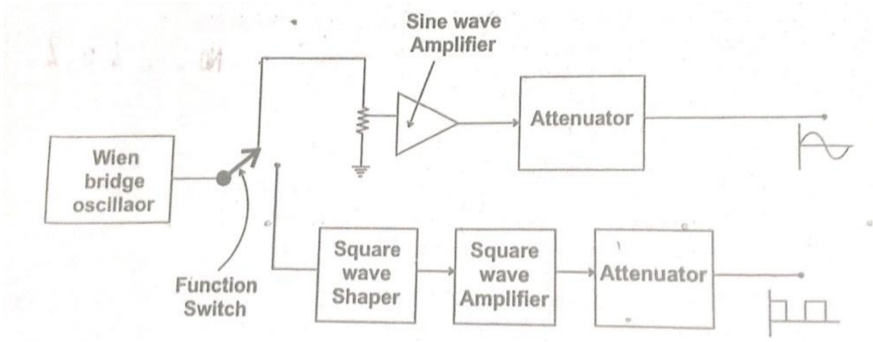


Fig. 29 Audio frequency generator.

The wein bridge oscillator is the main part of the sine-square wave generator. The wein bridge oscillator is used to generate a sine wave. The frequency of oscillations can be changed by varying the capacitance in the oscillator. The frequency can be changed insteps by switching in resistors of different values.

The output of the Wein bridge oscillator goes to the function switch. The function switch directs the oscillators output either to the sine wave amplifier or to the square wave shaper. The output is varied by means of an attenuator. The instrument generates a frequency ranging from 10Hz to 1MHz.

4.2 FUNCTION GENERATORS

Function generators are items of test equipment that are able to generate a variety of simple repetitive waveforms. Straightforward signal generators such as RF signal generators or simple audio oscillators focus on producing a good sine waves, but in many cases other waveforms are needed. In addition to producing sine waves, function generators may typically produce other repetitive waveforms including sawtooth and triangular waveforms, square waves, and pulses. Another feature included on many function generators is the ability to add a DC offset. Often some of the low end function generators may only operate up to frequencies of possibly around 100 kHz as the various shaped waveforms are normally only needed at lower frequencies. However many other more comprehensive function generators are able to operate at much higher frequencies, often up to 10 or 20 MHz.

Function generator controls

In addition to a selection of the basic waveforms that are available, other controls on the function generator may include:

- ***Frequency:*** As would be expected, this control alters the basic frequency at which the waveform repeats. It is independent of the waveform type.
- ***Waveform type :*** This enables the different basic waveform types to be selected:

1. Sine wave

2. Square wave
3. Triangular wave

- ***DC offset:*** This alters the average voltage of a signal relative to 0V or ground.
- ***Duty cycle:*** This control on the function generator changes the ratio of high voltage to low voltage time in a square wave signal, i.e. changing the waveform from a square wave with a 1:1 duty cycle to a pulse waveform, or a triangular waveform with equal rise and fall times to a sawtooth.

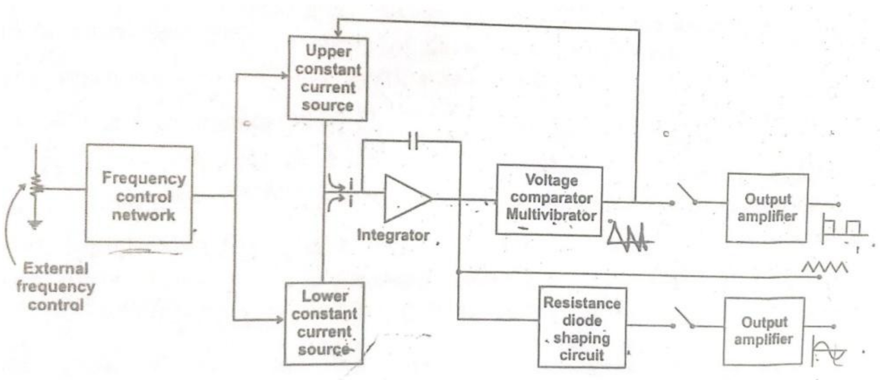


Fig. 30 Function generator

In the function generator the frequency is controlled by varying the magnitude of the current which drives the integrator. The frequency control voltage regulates two current sources namely the upper current source and the lower current source. The upper current source supplies constant current to the integrator. The output of the integrator linearly increases with respect to time.. If the current, charging the capacitor increases or decreases, the slope of the output voltage increases or decreases. the voltage comparator multivibrator circuit changes the state of the network when the output voltage of the integrator equals the maximum predetermined upper level. because of this change in state, the upper current source is removed and the lower current source is switched ON. This lower current source supplies opposite current to the integrator circuit. The output of integrator decreases linearly with time.

When the output voltage reaches a predetermined level on the negative slope of the output waveform, the voltage comparator multivibrator again changes the condition of the network by switching OFF the lower current source. The integrator output voltage has a triangular waveform. The frequency of this triangular waveform is determined by the magnitudes of the currents supplied by the upper and lower current sources. The output of the integrator is passed to the comparator and we can get square waveform. The triangular waveform is converted to sine wave by using diode resistance shaping circuit.

Features of the function generator:

1) The frequency range is 0.01Hz to 100Khz

2) Can produce various waveforms like sine, sawtooth, triangular, square wave, etc.,

3) The accuracy is within +1% or -1% in low frequency range.

4) The distortion is less than 1% of the sine wave.

5) Can be phase locked to another external signal source.

6) A continuous adjustable dc offset is available between -5V to +5V.

4.3 Pulse generators

A pulse generator is either an electronic circuit or a piece of electronic test equipment used to generate rectangular pulses. Pulse generators are used primarily for working with digital circuits, related function generators are used primarily for analog circuits. The fundamental difference between a square wave generator and a pulse generator depends upon the duty cycle. The duty cycle is defined as the ratio of average value of the pulse over one cycle to the peak value. It is also defined as ratio of the pulse width to the period of one cycle.

$$\text{Duty cycle} = \frac{PULSE\ WIDTH}{PULSE\ PERIOD}$$

$$\text{Duty cycle of a square wave} = \frac{\frac{1}{2}peak\ value}{peak\ value} = 0.5$$

Thus square wave produces an output voltage with equal ON and OFF periods, their duty cycle is 0.5 or 50% as the frequency of oscillations is varied. consider a general pulse as shown in figure below:

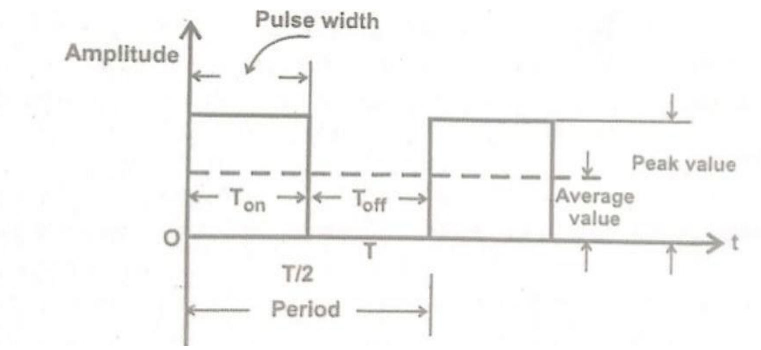


Fig. 31 Pulse waveform.

$$\text{Duty cycle} = \frac{Ton}{T} = \frac{Ton}{Ton+Toff}$$

Pulse characteristics and terminology:

The characteristics of pulse is shown below:

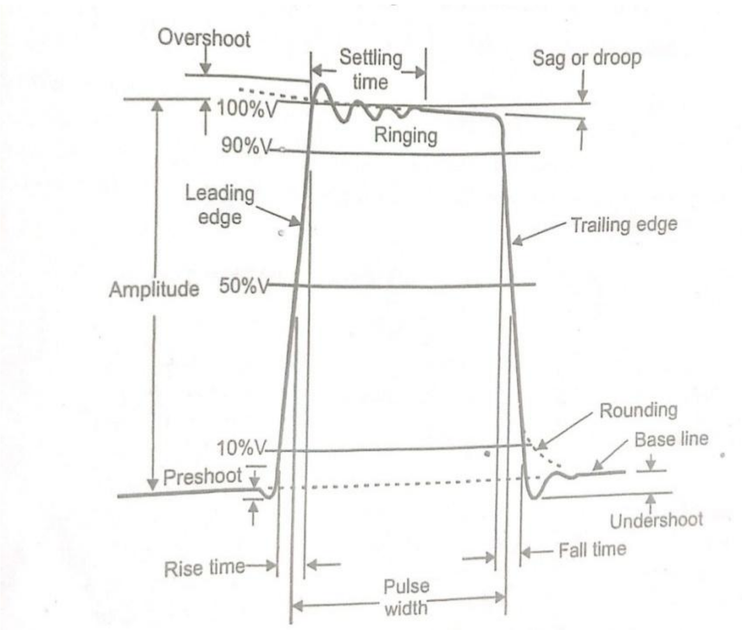


Fig.32 Pulse Characteristics.

1) Pulse rise and fall times:

Pulse rise time is the time needed for the pulse to go from 10% to 90% of its amplitude. The fall time is the time needed for the trailing edge to go from 90% to 10% These times are called leading edge and trailing edge transition times.

2) Linearity:

Linearity of a pulse is the deviation of an edge from the straight line drawn through the 10% and 90% amplitude points , expressed as percentage of pulse amplitude.

3) Pulse preshoot:

Its the deviation prior to reaching the base line at the start of the pulse.

4)Overshoot:

Overshoot is the maximum height immediately following the edge.

5) Undershoot:

A distortion of the base value immediately following a falling edge.

6) Ringing:

The positive and negative peak distortion, excluding overshoot or undershoot, on the pulse top or baseline.

7) Droop or Sag:

Occurs when the peak value gradually decreases during the pulse.

8) Pulse repetition rate (PRR):

The rate at which pulses are produced

9) Settling time:

The time required for the signal to decrease to a given percentage typically 1% to 5% of its peak value.

Pulse generator:

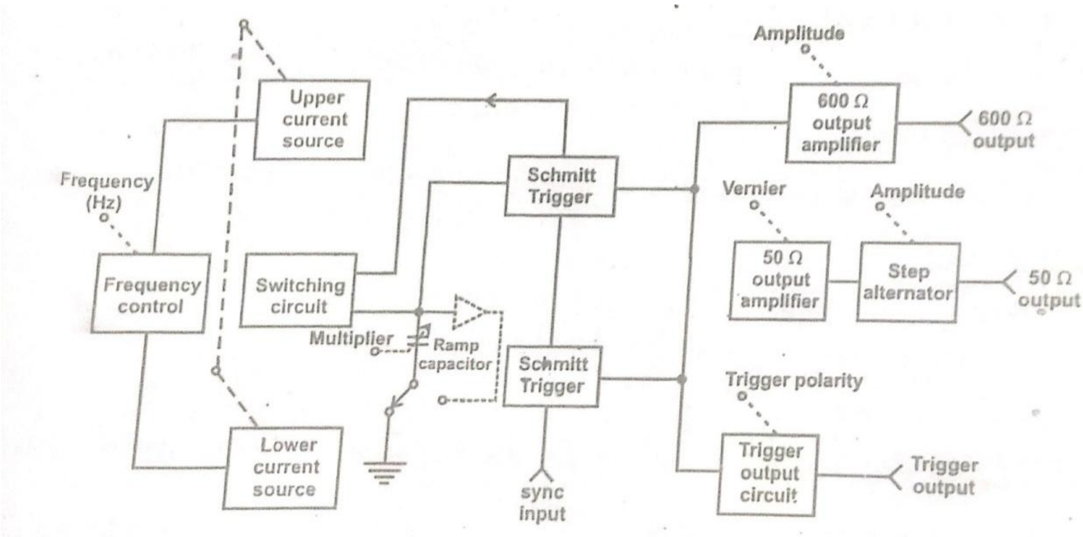


Fig. 33 Block diagram of Pulse generator.

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 time of 70 ns at 30 V peak amplitude. The instrument can be operated as a free running generator or, it can be synchronized with external signals.

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 the internal circuit components, the Schmitt trigger changes state. The trigger circuit output becomes negative and reverses the condition of the current switch. The capacitor discharges linearly, controlled by the lower 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 i_1/i_2 determines the duty cycle, and is controlled by symmetry control. The sum of i_1 and i_2 determines the frequency. The size of the capacitor is selected by the multiplier switch. The unit is powered by an internal supply that provides regulated voltages for all stages of the instrument.

4.4 RF Signal generator

Radio frequency signal generators (RF signal generators) are a particularly useful item of test equipment widely used in RF microwave design and test applications. These microwave and RF signal generators come in a variety of forms and with a host of facilities and capabilities. In order to gain the most from any RF signal generator or microwave signal generator, it is necessary to have an understanding of its operation and the capabilities it possesses.

Types of RF signal generator:

It is possible to design radio frequency signal generators in a variety of ways. Also with developments that have been made in electronics circuitry over the years, different techniques have evolved. It can be said that there are two forms of signal generator that can be used:

- ***Free running RF signal generators:*** These signal generators are rarely used these days as their frequency tends to drift. However they do have the advantage that the signal produced is very clean and does not have the level of noise (phase noise) either side of the main signal that is present on some other radio frequency signal generators.

  Some signal generators used a form of frequency locked loop to provide a means of adding some frequency stability while still retaining the very low levels of phase noise. Again, these are not common these days because the performance of RF signal generators using frequency synthesizer technology has considerably improved.

- ***Synthesized radio frequency signal generators:*** Virtually all radio frequency signal generators used today employ frequency synthesizers. Using this technique enables frequencies to be entered directly from a keypad, or via remote control and it also enables the output signal to be determined very accurately. The accuracy being dependent upon either an internal reference oscillator that can have a very high degree of accuracy, or the signal can be locked to an external frequency reference which can be exceedingly accurate.

  There are two main techniques that are used within synthesized RF signal generators:

  - *Phase locked loop synthesizer:* Phase locked loop synthesizers are used within most RF signal generators as they enable signals to be generated over a wide range of frequencies with a relatively low level of spurious signals. Phase locked loop synthesizer technology is well developed and enables high performance RF signal generators to be produced using them.
  - *Direct Digital Synthesizer, DDS:* Direct digital synthesis techniques may be used in RF signal generators. They enable very fine frequency increments to be achieved relatively easily. However the maximum limit of a DDS is normally much lower than the top frequencies required for the signal generator, so they

are used in conjunction with phase locked loops to give the required frequency range.

RF signal generator operation
In order to understand the operation of a generic microwave or RF signal generator it is useful to understand what is included in terms of a basic block diagram.

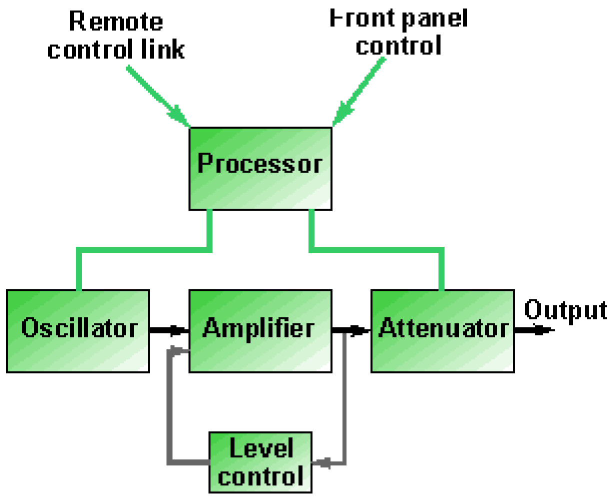


Fig. 34 Block diagram of RF signal generator.

The diagram shows a very simplified block diagram for an RF / Microwave signal generator.

From this, it can be seen that the generator has a few major blocks within it:

- ***Oscillator:*** The most important block within the RF signal generator is the oscillator itself. This can be any form of oscillator, but today it would almost certainly be formed from a frequency synthesizer. This oscillator would take commands from the controller and be set to the required frequency.

- ***Amplifier:*** The output from the oscillator will need amplifying. This will be achieved using a special amplifier module. This will amplify the signal, typically to a fixed level. It would have a loop around it to maintain the output level accurately at all frequencies and temperatures.

- ***Attenuator:*** An attenuator is placed on the output of the signal generator. This serves to ensure an accurate source impedance is maintained as well as allowing the generator level to be adjusted very accurately. In particular the relative power levels, i.e. when changing from one level to another are very accurate and represent the accuracy of the attenuator. It is worth noting that the output impedance is less accurately defined for the highest signal levels where the attenuation is less.

- ***Control:*** Advanced processors are used to ensure that the RF and microwave signal generator is easy to control and is also able to take remote control commands. The processor will control all aspects of the operation of the test equipment.

V FREQUENCY SYNTHESIZER

A frequency synthesizer is an electronic circuit that generates a range of frequencies from a single reference frequency. Frequency synthesizers are used in many modern devices such as radio receivers, televisions, mobile telephones, radiotelephones, walkie-talkies, CB radios, cable television converter boxes satellite receivers, and GPS systems. A frequency synthesizer may use the techniques of frequency multiplication, frequency division, direct digital synthesis, frequency mixing, and phase-locked loops to generate its frequencies. The stability and accuracy of the frequency synthesizer's output are related to the stability and accuracy of its reference frequency input. Consequently, synthesizers use stable and accurate reference frequencies, such as those provided by crystal oscillators.

Most frequency synthesizers are based around a phase locked loop or PLL.The PLL uses the idea of phase comparison as the basis of its operation. From the block diagram of a basic loop shown below, it can be seen that there are three basic circuit blocks, a phase comparator, voltage controlled oscillator, and loop filter. A reference oscillator is sometimes included in the block diagram, although this is not strictly part of the loop itself even though a reference signal is required for its operation.

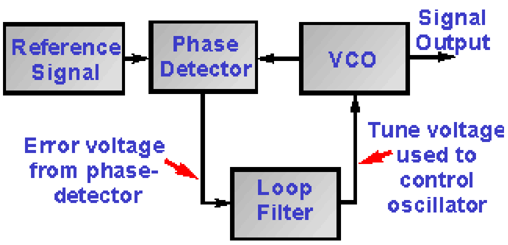


Fig. 35 A Phase locked loop

The phase locked loop, PLL, operates by comparing the phase of two signals. The signals from the voltage controlled oscillator and reference enter the phase comparator. Here a third signal equal to the phase difference between the two input signals is produced.

The phase difference signal is then passed through the loop filter. This performs a number of functions including the removal of any unwanted products that are present on this signal. Once this has been accomplished it is applied to the control terminal of the voltage controlled oscillator. This tune voltage or error voltage is such that it tries to reduce the error between the two signals entering the phase comparator. This means that the voltage controlled oscillator will be pulled towards the frequency of the reference, and when in lock there is a steady state error voltage. This is proportional to the phase error between the two signals, and it is constant. Only when the phase between two signals is changing is there a frequency difference. As the phase difference remains constant when the loop is in lock this means that the frequency of the voltage controlled oscillator is exactly the same as the reference.

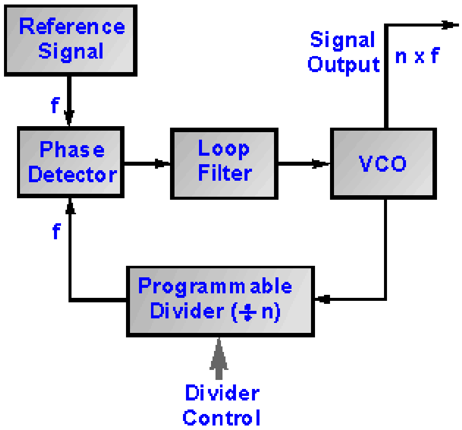


Fig.36 Digital Frequency Synthesizer

This is the concept that is at the root of most single loop synthesizers. It involves placing a digital divider in the loop between the voltage controlled oscillator. This means that the voltage controlled oscillator frequency will be divided by the division ratio of the divider, e.g. n, and the VCO will run at n times the phase comparison frequency. By changing the division ratio of the divider, the output frequency of the oscillator can be changed. This makes the frequency synthesizer programmable. These digital frequency synthesizers are ideal for many applications on their own. They perform well where the differences between channels are relatively high. Where virtual continuous tuning using steps of 1 Hz or 10Hz may be needed, this requires very high division ratios and this can degrade the phase noise performance and give rise to other issues.

VI.NOISE GENERATORS

The simplified block diagram of a random noise generators is shown below:

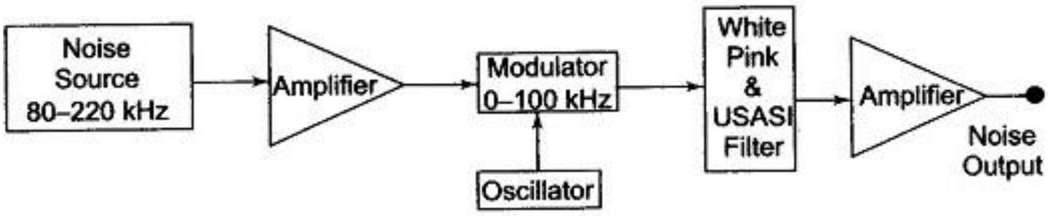


Fig.37 Random noise generator.

This instrument offers the possibility of using a single measurement to indicate performance over a wide range of frequency band, instead of many measurements at frequency at a time. The spectrum of random noise covers all frequencies and its referred to as White noise. That is the noise having equal power density at all frequencies. The power density spectrum tells us how the energy of a signal is distributed in frequency, but it does not specify the signal uniquely, nor does it tell us very much about how the amplitude of the signal varies with time. it contains no phase information. The method of generating noise is usually to use a semi conductor noise diode, which delivers frequencies in a band roughly extending from 80 — 220 kHz. The output from the noise diode is amplified and heterodyned down to the audio frequency band by means of a balanced symmetrical modulator. The filter arrangement controls the bandwidth and supplies an output signal in three spectrum choices, white noise, pink noise and Usasi noise.

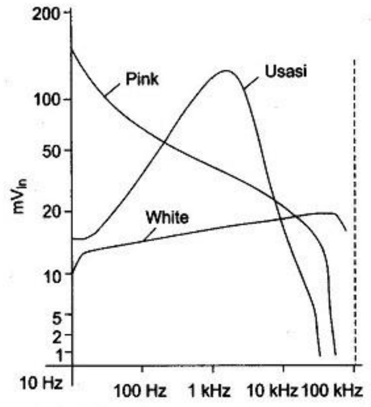


Fig.38 Random Noise generator.

From the above figure we can observe that white noise is flat from 20 Hz to 25 kHz and has an upper cut-off frequency of 50 kHz with a cut-off slope of —12 dbs/ octave. Pink noise is so called because the lower frequencies have a larger amplitude, similar to red light. Pink noise has a voltage spectrum which is inversely proportional to the square root of frequency and is used in bandwidth analysis. Usasi noise ranging simulates the energy distribution of speech and music frequencies and is used for testing audio amplifiers and loud speakers.

6.1 ARBITRARY WAVEFORM GENERATORS

The arbitrary waveform generator(AWG) comes as close as possible to being a universal signal source. Waveforms can be created analytically with great precision using equations, or captured using digitizers or digital oscilloscopes and replayed. Additionally, modular AWG's offer compact size and highly integrated compatibility with their host computers making them ideal for automated test systems.

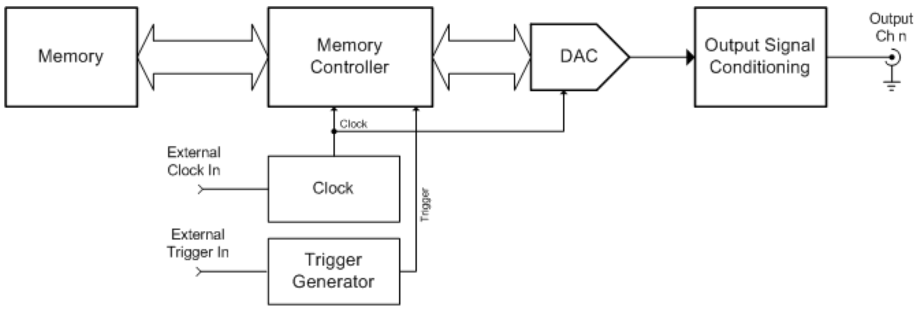


Fig.39 Arbitrary Waveform Generators.

Arbitrary waveform generators (AWG's) are digital signal sources that operate very much like a digitizer in reverse. Where a digitizer samples an analog waveform, digitizes it and then stores it in its acquisition memory, the AWG has a numeric description of the waveform stored in waveform memory. Selected samples of the waveform are sent to a digital to analog converter (DAC) and then, with appropriate filtering and signal conditioning, are output as an analog waveform. Figure contains a conceptual block diagram of an AWG. The waveform, in numeric form, is loaded into the waveform memory. Like the acquisition memory in a digitizer this memory has to be capable of being clocked at the highest sampling rate supported by the AWG. When commanded, the contents of the waveform memory are sent to the DAC where the digital values are converted into an analog voltage. Some DAC's allow additional interpolation to reach a higher update rate at the output than supplied by the waveform memory. The memory controller keeps track of the elements of each waveform component in the waveform memory, and any associated links, and outputs them in the correct order. In addition, in order to save memory space, it can loop on repetitive elements so that these elements need be listed only once in the waveform memory. The DAC output is harmonic rich and requires filtering. This is accomplished on the output stage which filters and conditions the signal by adjusting gains and offsets to meet the user's waveform specification.

The timing of the waveform is controlled by a clock which has provision for using an internal or an external clock source. Synchronization is maintained by the trigger generator which causes the waveform to be output or advanced based on a user specified event. Trigger events can be internal, external, or linked to another modular AWG or digitizer. The actual implementation of the elements above vary with specific models but all AWG's have similar elements.

6.2 PATTERN GENERATOR (VIDEO PATTERN GENERATOR)

A pattern generator provides video signals directly, and with RF modulation, on standard TV channels for alignment, testing and servicing of TV receivers. The output signal is designed to produce simple geometric patterns like vertical and horizontal bars, checkerboard, cross-hatch, dots, etc.

These patterns are used for linearity and video amplifier adjustment. In addition to this, an FM sound signal is also provided in pattern generators for aligning sound sections of the receiver.

A simplified functional block diagram of a pattern cum sound signal generator is shown below:

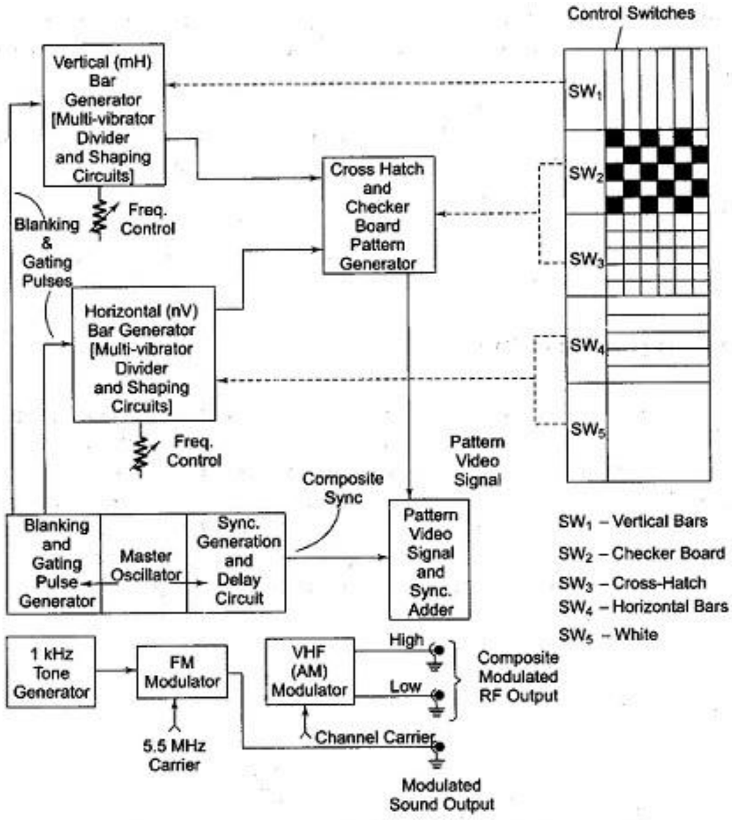


Fig.40 Simplified block diagram of a pattern cum sound signal generator.

The generator employs two stable chains of multivibrators, dividers and pulse shaping circuits, one below the line frequency to produce a series of horizontal bars, and another above 15625 Hz to produce vertical bars. The signals are modified into short duration pulses, which when fed to the video section of the receiver along with the sync pulse train, produce fine lines on the screen. Multivibrators produce a square wave video signal at m times the horizontal frequency to provide m vertical black and white bars. After every m cycles, the horizontal blanking pulse triggers the multivibrators for synchronising the bar signal on every line. A control on the front panel of the Video Pattern Generator enables variation of multivibrators frequency to change the number of bars.

Similarly, square wave pulses derived either from 50 Hz mains of from the master oscillator are used to trigger another set of multivibrator to generate square wave video signals that are n times the vertical frequency. On feeding the video amplifier these produce horizontal black and white bars. The number of horizontal bars can also be varied by a potentiometer that controls the switching rate of the corresponding multivibrator. (The bar pattern signal is combined with the sync and blanking pulses in the video adder to produce composite video signals before being fed to the modulator). The provision of switches in the signal path of the two multivibrators enables the generation of various patterns. If both mH and nV switches are off, a blank white raster is produced. With only the mH switch on, vertical bars are produced, and with only the nV switch on, horizontal bars are generated. With both switches on, a cross-hatch pattern will be produced.

The horizontal bar pattern is used for checking vertical linearity. These bars should be equally spaced throughout the screen for linearity. Similarly, the vertical bar pattern can be used for checking and setting horizontal linearity. With the cross-hatch pattern formed by the vertical and horizontal lines, linearity can be adjusted more precisely, because any unequal spacing of the lines can be discerned.

Picture centering and aspect ratio can also be checked with the cross-hatch pattern by counting the number of squares on the vertical and horizontal sides of the screen. The Video Test Pattern Generator can also be used for detecting any spurious oscillations in the sweep generation circuits, interaction between the two oscillators, poor interlacing, and barrel and pin cushion effects. Modulated picture signals are available on limited channels for injecting into the RF section of the receiver. Similarly, an FM sound signal with a carrier frequency of 5.5 MHz ± 100 kHz, modulated by a 1 kHz tone, is provided for aligning sound IF and discriminator circuits. A 75/300 Ω VHF balun is usually available as a standard accessory with the Video Test Pattern Generator.

6.3WOBBLUSCOPE

This instrument combines a Sweep generator, a Marker generator, and an oscilloscope, as shown in below.

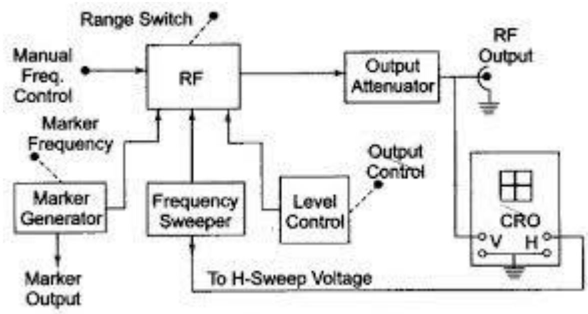


Fig.41 Block diagram of a Wobbluscope

It is a very useful unit for the alignment of RF, IF and video sections of a TV receiver. It may not have all the features of a high quality sweep generator but is an economical and compact piece of equipment specially designed for TV servicing. The oscilloscope usually has a provision for TV-V (vertical) and TV-H (horizontal) sweep modes. An RF output, down to 1 MHz, is available for video amplifier testing.

SCHOOL OF ELECTRICAL AND ELECTRONICS

DEPARTMENT OF ELECTRONICS AND INSTRUMENTATION

UNIT – II

ELECTRONIC MEASUREMENT AND INSTRUMENTATION – SIC 1305

UNIT-II

I.DATA CONVERTERS AND CONNECTORS

1.1 ADC AND DAC SPECIFICATIONS

Resolution:

The resolution of a DAC is the smallest change in the output of the DAC for any change in digital input.i.e. if a input to DAC changes one bit, how much analog output has changed in full scale deflection.

% resolution = [Step size / Full scale output (FSO)] * 100

In other way the resolution is the number of states into which the full scale output is divided. i.e if a 8 bit DAC can resolve the FSO up to 255 levels. Each level of output is called step size and for higher number of bits the resolution will be better.

% resolution = [1/(2^N-1) *100]

Normally the resolution will be in milli volts.

Accuracy

The Accuracy of a DAC is the difference between output practical analog output to the ideal expected output for a given digital input. The DAC is contains electronic components where the gain plays a major role which can introduce gain error in the output. Due to the the full scale output may differ compared to ideal one. For an example if a DAC of 10 V is said to have an accuracy of 0.01% there will be 10mv output deviation. The another factor which implicates the accuracy is the zero offset error i.e for a zero input the output of DAC reflects some offset value.

Conversion Speed

The conversion speed of the DAC is output analog value settling time period for a change in the digital input. This is also called settling time period of DAC. Normally it will be micro seconds and in some advanced micro controller DAC it may be nano seconds.

Monotonicity

The Digital to Analog Converter is said to be monotonic if its analog value is either increasing or equal to previous value for an LSB change in input digital signal.

Offset/ Zero scale error:

An input code of zero may be expected to give 0V output. A small offset may be present and the transfer characteristic does not pass through the origin.

Linearity:

The input-output characteristic of a D/A converter. zero offset and gain develop in the characteristic which passes through the origin and full scale points. But it is not sure that the intermediate points will always lie in the straight line. Avery small error in the weighting factor for a fraction LSB will cause non-linearity. Linearity can be expressed by deviation from the ideal line as a percentage or fraction of LSB. It is specified as ±LSB or ±1/2 LSB.

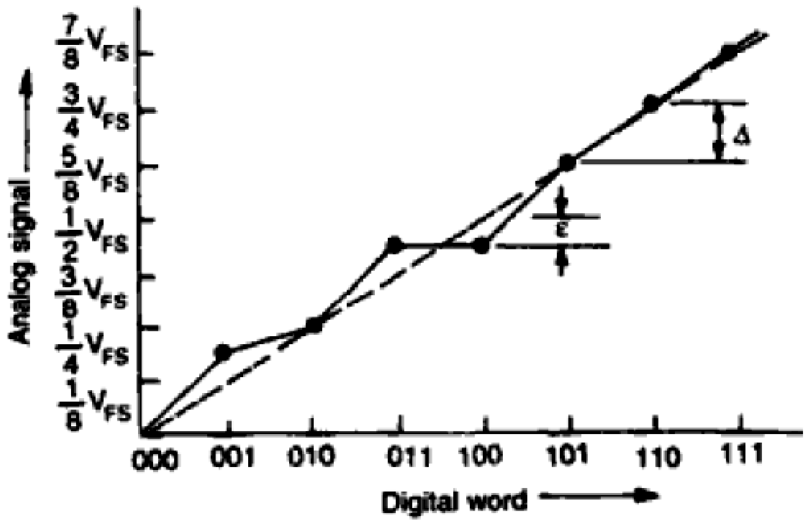


Fig.1: Linearity error for a 3-bit ADC

Settling time:

This is usually expressed as the time taken to settle within half LSB. Generally settling time will be 500ns.

Stability:

The ability of a DAC to produce a stable output all the time is called as Stability. The performance
of a converter changes with drift in temperature, aging and power supply variations. So all the parameters such as offset, gain, linearity error & monotonicity may change from the values specified in the datasheet. Temperature sensitivity defines the stability of a D/A converter.

1.2 Quantization error

Quantization is representing the sampled values of the amplitude by a finite set of levels, which means converting a continuous-amplitude sample into a discrete-time signal.

For any system, during its functioning, there is always a difference in the values of its input and output. The processing of the system results in an error, which is the difference of those values.

The difference between an input value and its quantized value is called a Quantization Error. A Quantizer is a logarithmic function that performs Quantization (rounding off the value). An analog-to-digital converter (ADC) works as a quantizer.

The following figure illustrates an example for a quantization error, indicating the difference between the original signal and the quantized signal.

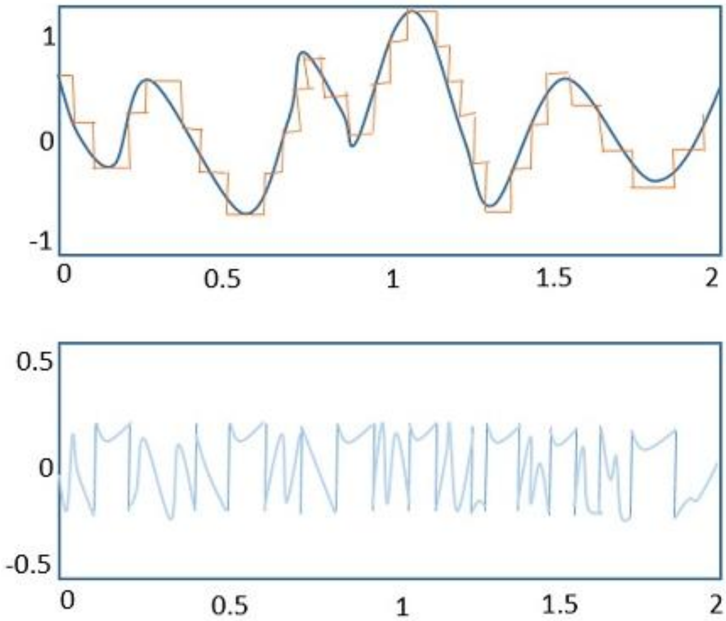


Fig.2 Quantization error.

2.TYPES OF ADC

2.1 FLASH TYPE ADC:

This is the simplest possible A/D converter. It is at the same time, the fastest and most expensive technique. Figure shows a 3 bit A/D converter. The circuit consists of a resistive divider network, 8 op-amp comparators and a 8-1ine to 3-1ine encoder (3-bit priority encoder). The Comparator and its truth table are shown in Figure below

Voltage input	Logic output X
$V_a > V_d$	$X = 1$
$V_a < V_d$	$X = 0$
$V_a = V_d$	Previous value

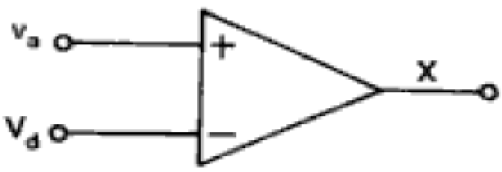


Fig.3 A comparator and its truth table

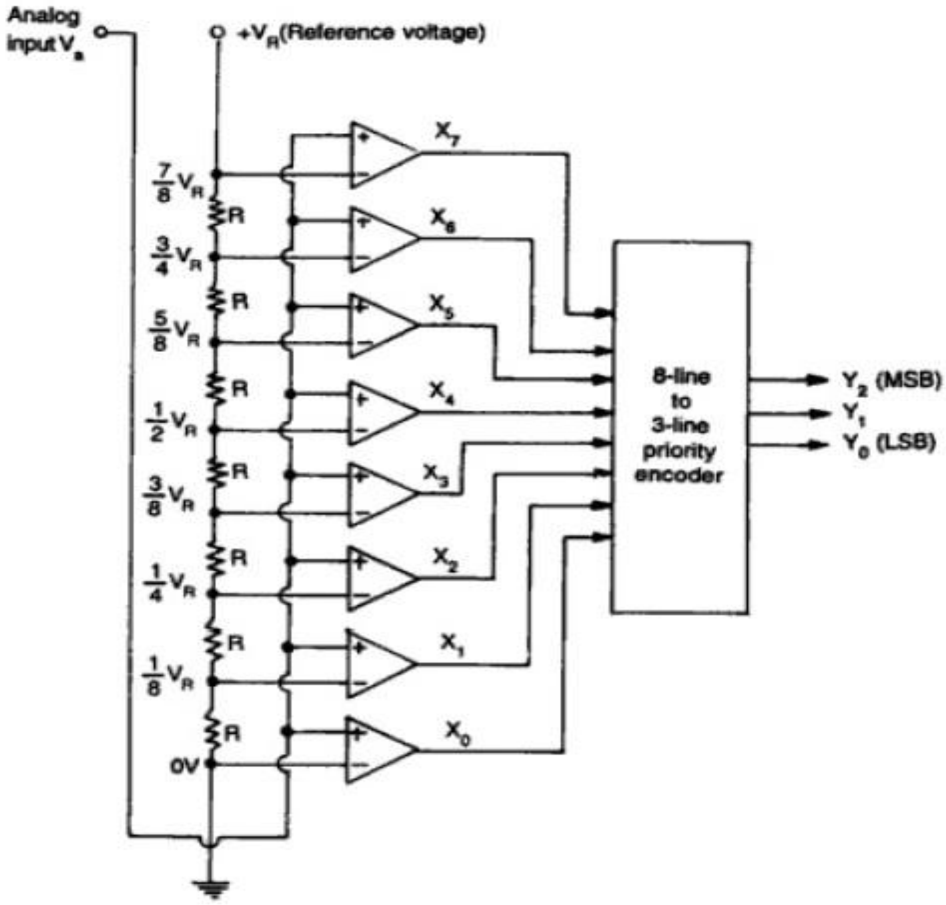


Fig.4 Flash type A-D converter

Table 1:Truth table for Flash type ADC

Input voltage V_a	X_7	X_6	X_5	X_4	X_3	X_2	X_1	X_0	Y_2	Y_1	Y_0
0 to $V_R/8$	0	0	0	0	0	0	0	1	0	0	0
$V_R/8$ to $V_R/4$	0	0	0	0	0	0	1	1	0	0	1
$V_R/4$ to 3 $V_R/8$	0	0	0	0	0	1	1	1	0	1	0
3 $V_R/8$ to $V_R/2$	0	0	0	0	1	1	1	1	0	1	1
$V_R/2$ to 5 $V_R/8$	0	0	0	1	1	1	1	1	1	0	0
5 $V_R/8$ to 3 $V_R/4$	0	0	1	1	1	1	1	1	1	0	1
3 $V_R/4$ to 7 $V_R/8$	0	1	1	1	1	1	1	1	1	1	0
7 $V_R/8$ to V_R	1	1	1	1	1	1	1	1	1	1	1

A small amount of hysteresis is built into the comparator to resolve any problems that might occur if both inputs were of equal voltage as shown in the truth table. From the Figure, at each node of the resistive divider, a comparison voltage is available. Since all the resistors are of equal value, the voltage levels available at the nodes are equally divided between the reference voltage V_R and the ground. The purpose of the circuit is to compare the analog input voltage V_a with each of the node voltages. The truth table for the flash type A/D converter is shown above.

Advantages of flash type A/D converter

- ✓ High speed simultaneous conversion
- ✓ Typical conversion time is 100 ns or less.

Disadvantages of flash type A/D converter

- ✓ The number of comparators required almost doubles for each added bit.
- ✓ Larger the value of n (number of bits), the more complex is the priority encoder.

2.2 COUNTER TYPE ADC

The counter type ADC is the basic form of ADC which is also called as ramp type ADC or stair case approximation ADC. This circuit consists of N-bit counter, DAC and comparator

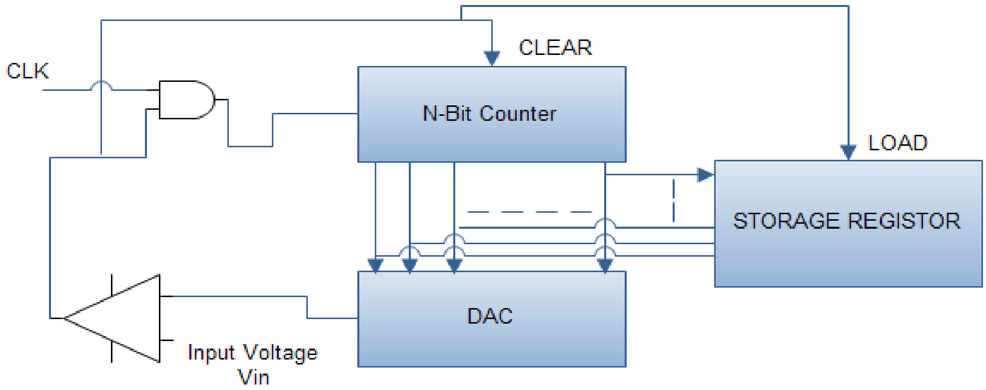


Fig.5 Counter type ADC

The N bit counter generates an n bit digital output which is applied as an input to the DAC. The analog output corresponding to the digital input from DAC is compared with the input analog voltage using an op-amp comparator. The op-amp compares the two voltages and if the generated DAC voltage is less, it generates a high pulse to the N-bit counter as a clock pulse to increment the counter. The same process will be repeated until the DAC output equals to the input analog voltage.

If the DAC output voltage is equal to the input analog voltage, then it generates low clock pulse and it also generates a clear signal to the counter and load signal to the storage resistor to store the corresponding digital bits. These digital values are closely matched with the input analog values with small quantization error. For every sampling interval the DAC output follows a ramp fashion so that it is called as Digital ramp type ADC. And this ramp looks like stair cases for every sampling time so that it is also called as staircase approximation type ADC.

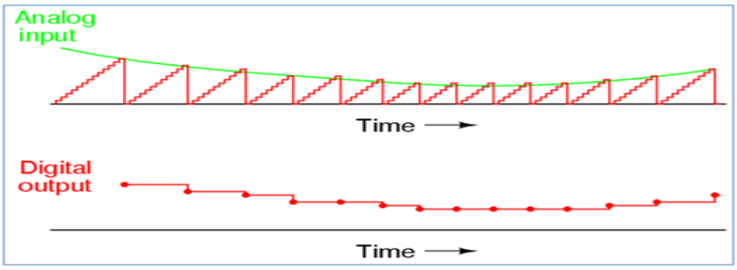


Fig.6 Digital output and analog input for a counter type ADC

Conversion time of ADC is the time taken by the ADC to convert the input sampled analog value to digital value. Here the maximum conversion of high input voltage for a N bit ADC is the clock pulses required to the counter to count its maximum count value. So

The maximum conversion of Counter type ADC is = (2^N-1) T

Where, T is the time period of clock pulse.

If N=2 bit then the $T_{max} = 3T$.

By observing the above conversion time of Counter type ADC it is illustrated that the sampling period of Counter type ADC should be as shown below.

$Ts >= (2^N-1)$ T

Advantages of Counter type ADC:

- Simple to understand and operate.
- Cost is less because of less complexity in design.

Disadvantages or limitations of Counter type of ADC:

- Speed is less because every time the counter has to start from ZERO.
- There may be clash or aliasing effect if the next input is sampled before completion of one operation.

2.3 SUCCESSIVE APPROXIMATION TYPE ADC

- A Successive Approximation Register (SAR) is added to the circuit
- Instead of counting up in binary sequence, this register counts by trying all values of bits starting with the MSB and finishing at the LSB.
- The register monitors the comparators output to see if the binary count is greater or less than the analog signal input and adjusts the bits accordingly

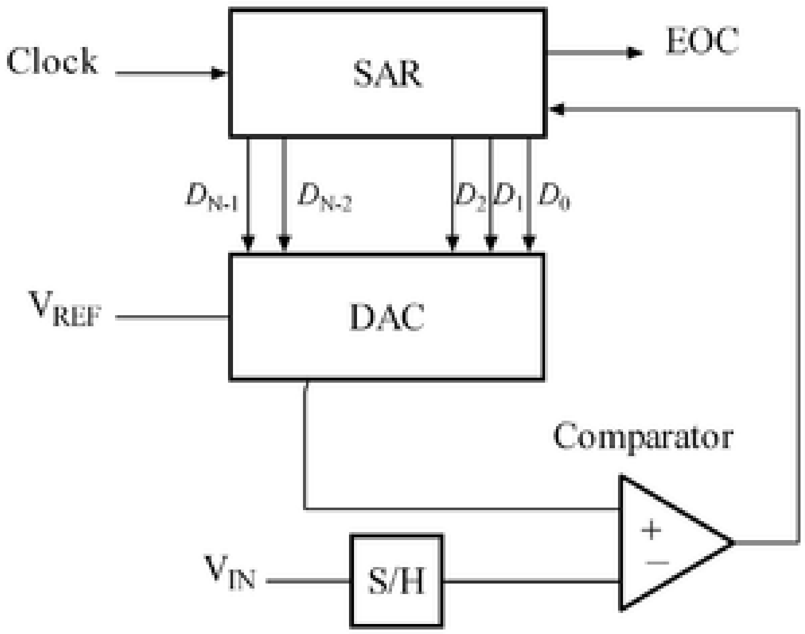


Fig.7 Successive Approximation ADC Circuit

Elements:

- DAC = Digital to Analog Converter
- EOC = End of Conversion
- SAR = Successive Approximation Register
- S/H = Sample and Hold Circuit
- V_{in} = Input Voltage
- Comparator
- V_{ref} = Reference Voltage

Algorithm

- Uses an n-bit DAC and original analog results
- Performs a binary comparison of V_{DAC} and V_{in}
- MSB is initialized at 1 for DAC
- If $V_{in} < V_{DAC}$ (V_{REF} / $2^{\wedge n=1}$) then MSB is reset to 0
- If $V_{in} > V_{DAC}$ (V_{REF} / $2^{\wedge n}$) Successive Bits set to 1 otherwise 0
- Algorithm is repeated up to LSB
- At end DAC in = ADC out

- N-bit conversion requires N comparison cycles

Example 1:

5-bit ADC, V_{in}=0.6V, V_{ref}=1V

Cycle 1 => MSB=1

SAR = $\underline{1}$ 0 0 0 0

$V_{DAC} = V_{ref}/2^{\wedge 1} = .5$ $V_{in} > V_{DAC}$ SAR unchanged = 1 0 0 0 0

- Cycle 2

SAR = 1 $\underline{1}$ 0 0 0

V_{DAC} = .5 +.25 = .75 $V_{in} < V_{DAC}$ SAR bit3 reset to 0 = 1 0 0 0 0

- Cycle 3

SAR = 1 0 $\underline{1}$ 0 0

V_{DAC} = .5 + .125 = .625 $V_{in} < V_{DAC}$ SAR bit2 reset to 0 = 1 0 0 0 0

- Cycle 4

SAR = 1 0 0 $\underline{1}$ 0

V_{DAC} = .5+.0625=.5625 $V_{in} > V_{DAC}$ SAR unchanged = 1 0 0 1 0

- Cycle 5

SAR = 1 0 0 1 $\underline{1}$

V_{DAC} = .5+.0625+.03125= $\underline{.59375}$

Table 2:Input vs Voltage

Bit	4	3	2	1	0
Voltage	.5	.25	.125	.0625	.03125

Advantages

- Capable of high speed and reliable
- Medium accuracy compared to other ADC types
- Good tradeoff between speed and cost
- Capable of output the binary number in serial (one bit at a time) format.
- High resolution

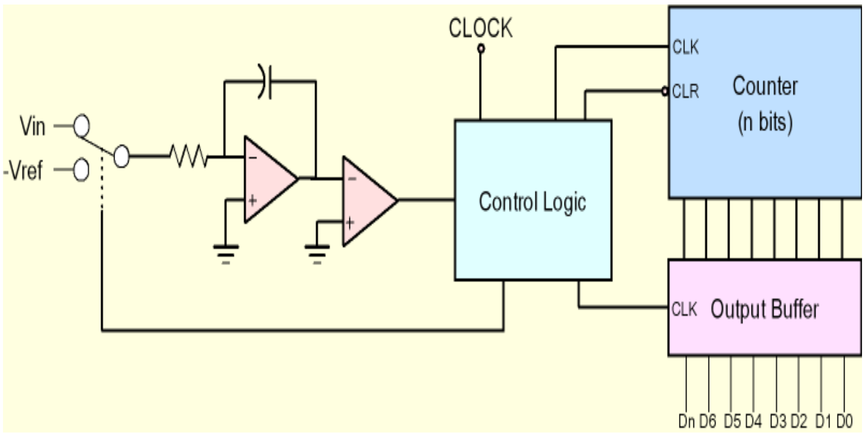
- No precision external components needed

Disadvantages

- Higher resolution successive approximation ADC's will be slower
- Speed limited.

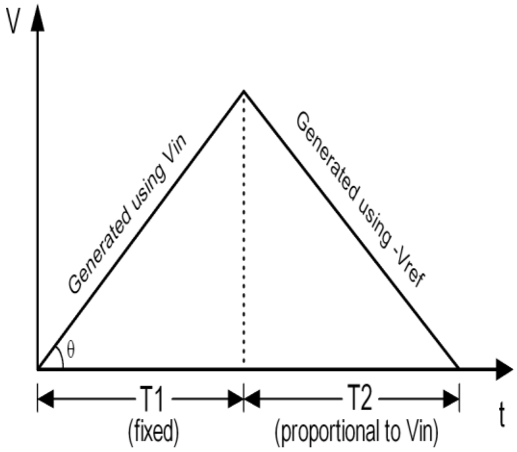
2.4 DUAL SLOPE ADC

- An unknown input voltage is applied to the input of the integrator and allowed to ramp for a fixed time period (t_u)
- Then, a known reference voltage of opposite polarity is applied to the integrator and is allowed to ramp until the integrator output returns to zero (t_d)
- The input voltage is computed as a function of the reference voltage, the constant run-up time period, and the measured run-down time period.
- The run-down time measurement is usually made in units of the converter's clock, so longer integration times allow for higher resolutions.
- The speed of the converter can be improved by sacrificing resolution



$$V_{in} = -V_{ref} \frac{t_d}{t_u}$$

Fig.8 Dual Slope ADC



$$\left(V(t) = V(0) + \frac{V_{in}\,t}{RC}\right)^{*}$$

Fig.9 Graph

Table 3:Comparison

Type	**Speed (relative)**	**Cost (relative)**
Dual Slope	Slow	Med
Flash	Very Fast	High
Successive Appox	Medium – Fast	Low
Sigma-Delta	Slow	Low

2.5 INTRODUCTION TO DELTA-SIGMA

A Delta Sigma (SD-ADC) has a modulator and a digital filter (also known as decimation filter) as shown in figure below. A modulator converts the input analog signal into digital bit streams (1s and 0s). One can observe a bit, either 1’b1 or 1’b0 coming at every clock edge of the modulator.
The decimation filter receives the input bit streams and, depending on the over sampling ratio (OSR) value, it gives one N-bit digital output per OSR clock edge. For example, if we consider OSR to be 64, then the Filter gives one N-bit output for every 64 clock edges (64 data outputs of the modulator). Here N is the resolution of the SD ADC.

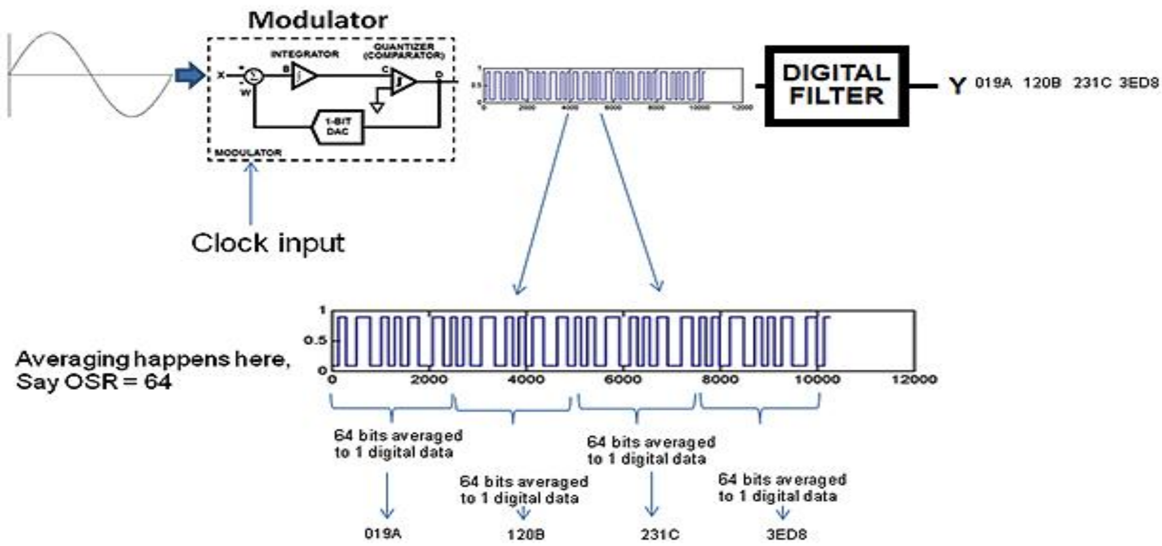


Fig.10 Delta Sigma ADC

In a conventional ADC, an analog signal is integrated, or sampled, with a sampling frequency and subsequently quantized in a multi-level quantizer into a digital signal. This process introduces quantization error noise. The first step in a delta-sigma modulation is delta modulation. In delta modulation the change in the signal (its delta) is encoded, rather than the absolute value. The result is a stream of pulses, as opposed to a stream of numbers as is the case with PCM. In delta-sigma modulation, the accuracy of the modulation is improved by passing the digital output through a 1-bit DAC and adding (sigma) the resulting analog signal to the input signal, thereby reducing the error introduced by the delta-modulation. Primarily because of its cost efficiency and reduced circuit complexity, this technique has found increasing use in modern electronic components such as DACs, ADCs, frequency synthesizers, switched-mode power supplies and motor controllers. Both ADCs and DACs can employ delta-sigma modulation. A delta-sigma ADC first encodes an analog signal using high-frequency delta-sigma modulation, and then applies a digital filter to form a higher-resolution but lower sample-frequency digital output. On the other hand, a delta-sigma DAC encodes a high-resolution digital input signal into a lower-resolution but higher sample-frequency signal that is mapped to voltages, and then smoothed with an analog filter. In both cases, the temporary use of a lower-resolution signal simplifies circuit design and improves efficiency.

In brief, because it is very easy to regenerate pulses at the receiver into the ideal form transmitted. The only part of the transmitted waveform required at the receiver is the time at which the pulse occurred. Given the timing information the transmitted waveform can be reconstructed electronically with great precision. In contrast, without conversion to a pulse stream but simply transmitting the analog signal directly, all noise in the system is added to the analog signal, permanently reducing its quality. Each pulse is made up of a step up followed after a short interval by a step down. It is possible, even in the presence of electronic noise, to recover the timing of these steps and from that regenerate the transmitted pulse stream almost noiselessly. Then the accuracy of the transmission process reduces to the accuracy with which the transmitted pulse stream represents the input waveform.

Delta-sigma modulation converts the analog voltage into a pulse frequency and is alternatively known as Pulse Density modulation or Pulse Frequency modulation. In general, frequency may vary smoothly in infinitesimal steps, as may voltage, and both may serve as an analog of an infinitesimally varying physical variable such as acoustic pressure, light intensity, etc. The substitution of frequency for voltage is thus entirely natural and carries in its train the transmission advantages of a pulse stream. The different names for the modulation method are the result of pulse frequency modulation by different electronic implementations, which all produce similar transmitted waveforms.

The ADC converts the mean of an analog voltage into the mean of an analog pulse frequency and counts the pulses in a known interval so that the pulse count divided by the interval gives an accurate digital representation of the mean analog voltage during the interval. This interval can be chosen to give any desired resolution or accuracy. The method is cheaply produced by modern methods; and it is widely used.

3.TYPES OF DAC

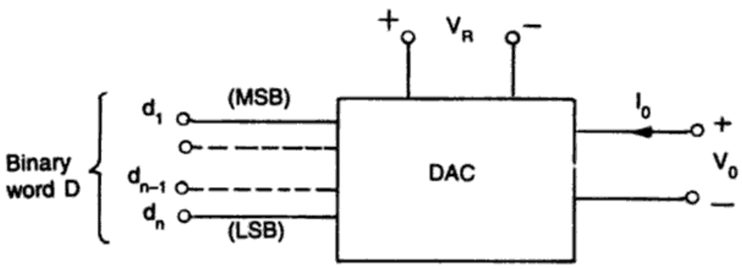


Fig 11: DAC

The input in the digital to analog converter is an n-bit binary word D and is combined with a reference voltage V_r to give an analog output signal. The output of a DAC can be either a voltage or current. For a voltage output DAC, the D/A converter is mathematically described as

$$V_o = K\ V_{FS}\ (d_1 2^{-1} + d_2 2^{-2} + \ldots + d_n 2^{-n}) \quad \text{------ (1)}$$

Where,

V_o = output voltage

V_{FS} = full scale output voltage

K = scaling factor usually adjusted to unity

$d_1\ d_2 \ldots\ldots d_n$ = n-bit binary fractional word with the decimal point located at the left
d_1 = most significant bit (MSB) with a weight of $V_{FS}/2$

d_n = most significant bit (MSB) with a weight of $V_{FS}/2^n$

There are various ways of implementing DAC

- Weighted-Resistor DAC

- 2R ladder DAC
- PWM type DAC

3.1 WEIGHTED RESISTOR DAC

One of the simplest circuits is shown in Figure uses a summing amplifier with a binary weighted resistor network. It has n- electronic switches d_1 d_2....... d_n controlled by binary input word. These switches are single pole double throw (SPDT) type. If the binary input to a particular switch is 1, it connects the resistance to the reference voltage (-V_R). And if the input bit is 0, the switch connects the resistor to the ground. From Figure (a) the output current I_o for an ideal op-amp can be written as

$$I_o = I_1 + I_2 + \ldots + I_n$$

$$= \frac{V_R}{2R} d_1 + \frac{V_R}{2^2 R} d_2 + \ldots + \frac{V_R}{2^n R} d_n$$

$$= \frac{V_R}{R} (d_1 2^{-1} + d_2 2^{-2} + \ldots + d_n 2^{-n})$$

The output voltage

$$V_o = I_o R_f = V_R \frac{R_f}{R} (d_1 2^{-1} + d_2 2^{-2} + \ldots + d_n 2^{-n}) \quad \text{----- (2)}$$

Comparing equation (1) with (2) it can be seen that if R_f = R then K = 1 and V_{FS}= V_R .

The circuit shown in Figure uses a negative reference voltage. The analog output voltage is therefore positive staircase as shown in Figure for a 3-bit weighted resistor DAC. It may be noted that

- ✓ Although the op-amp in Figure is connected in inverting mode, it can also be connected in non-inverting mode.
- ✓ The op-amp is simply working as a current to voltage converter.
- ✓ The polarity of the reference voltage is chosen in accordance with the type of the switch used. For example, for TTL compatible switches, the reference voltage should be = 5 V and the output will be negative.

The accuracy and stability of a DAC depends upon the accuracy of the resistors and the tracking of each other with temperature. There are however a number of problems associated with this type of DAC. One of the disadvantages of binary weighted type DAC is the wide range of resistor values required. It may be observed that for better resolution, the input binary word length has to be increased. Thus, as the number of bit increases, the range of resistance value increases. For 8-bit DAC, the resistors required are 2^0R, 2^1 R, 2^2 R… 2^7 R. the largest resistor is 128 times the smallest one for only 8-bit DAC. For a 12-bit DAC, the largest resistance required is 5.12 MΩ if the smallest is 2.5 kΩ. The fabrication of such a large resistance is IC is not practical. Also the voltage drop across such a large resistor due to the bias current would also affect the accuracy. The choice of smallest resistor value as 2.5 kΩ is reasonable; otherwise loading effect will be there. The difficulty of achieving and maintaining accurate ratios over such a wide range especially in monolithic form restricts the use of weighted resistor DACs to below 8-bits.

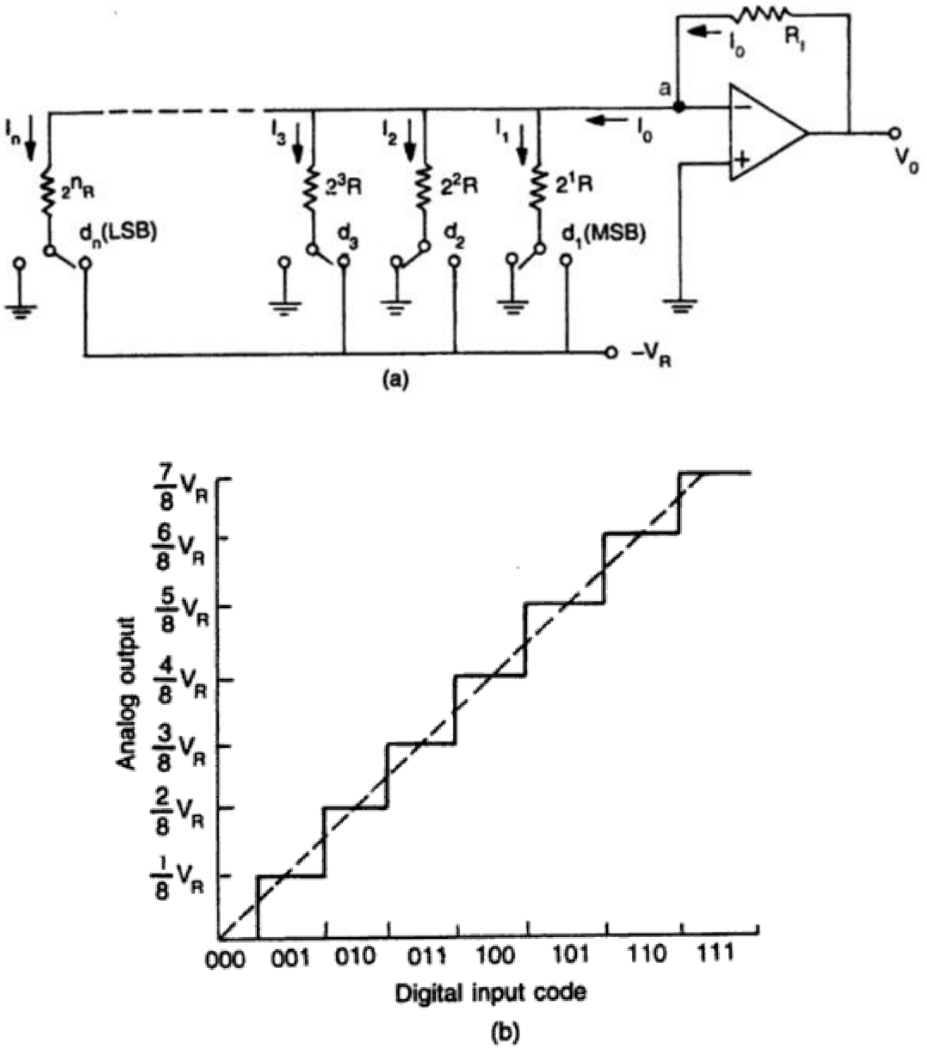


Fig.12 (a)A simple weighted resistor DAC (b) Transfer characteristics of a 3-bit DAC

3.2 R-2R LADDER DAC

Wide range of resistors are required in binary weighted resistor type DAC. This can be avoided by using R-2R ladder type DAC where only two values of resistors are required. It is well suited for integrated circuit realization. The typical value of resistor ranges from 2.5kΩ to 10kΩ.

For simplicity, consider a 3-bit DAC as shown in Figure, where the switch position $d_1\ d_2\ d_3$ corresponds to the binary word 100. The circuit can be simplified to the equivalent form of Figure (b) and finally to Figure (c). then, voltage at node C can be easily calculated by the set procedure of network analysis as

$$\frac{-V_R\left(\frac{2}{3}R\right)}{2R+\frac{2}{3}R} = -\frac{V_R}{4}$$

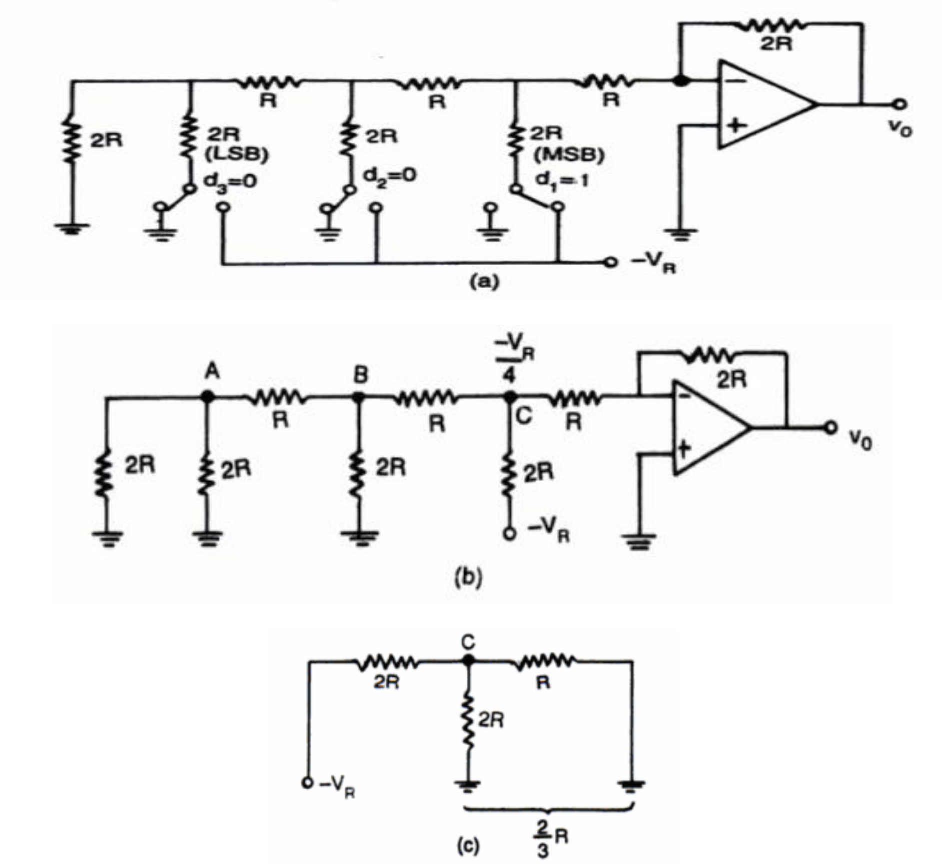


Fig.13 (a) R-2R ladder DAC (b) Equivalent circuit of (a), (c) Equivalent circuit of (b)

The output voltage

$$V_o = \frac{-2R}{R}\left(-\frac{V_R}{4}\right) = \frac{V_R}{2} = \frac{V_{FS}}{2}$$

The switch position corresponding to the binary word 001 in 3 bit DAC is shown in Figure (a). The circuit can be simplified to the equivalent form of Fig(b). The voltages at the nodes (A,B,C) formed by resistor branches are easily calculated in a similar fashion and the output voltage becomes

$$V_o = \left(-\frac{2R}{R}\right)\left(-\frac{V_R}{16}\right) = \frac{V_R}{8} = \frac{V_{FS}}{8}$$

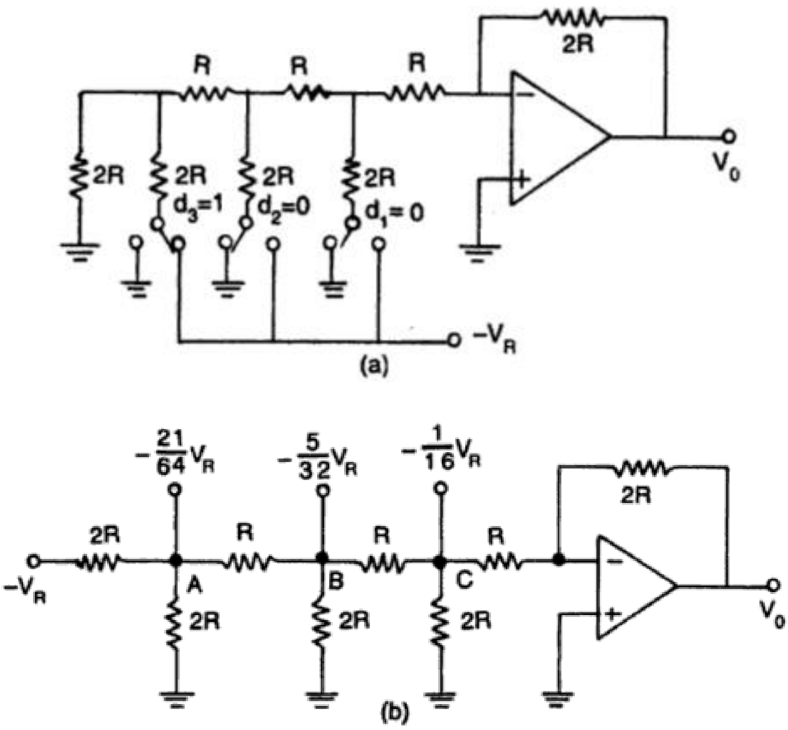


Fig.14 (a) R-2R ladder DAC for switch positons 001 (b) Equivalent circuit

In a similar fashion, the output voltage for R-2R ladder type DAC corresponding to other 3-bit binary words can be calculated.

3.3 PWM TYPE DAC

The PWM signal outputs on a device are variable duty cycle square-waves with 3.3 volt amplitude. These signals can each be decomposed into a D.C. component plus a new square-wave of identical duty-cycle but with a time-average amplitude of zero. Figure below depicts this graphically.

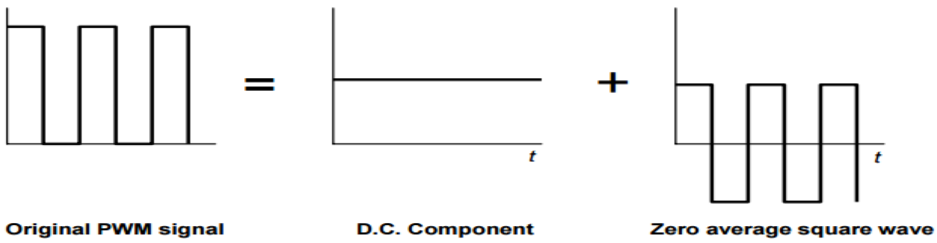


Fig.15 Decomposition of PWM signal

The idea behind realizing digital-to-analog (D/A) output from a PWM signal is to analog low-pass filter the PWM output to remove most of the high frequency components, ideally leaving only the D.C. component. This is depicted in Figure below. The bandwidth of the low-pass filter will essentially determine the bandwidth of the digital-to-analog converter. A frequency analysis of the PWM signal is given in the next section in order to provide a theoretical basis for the filtering strategy.

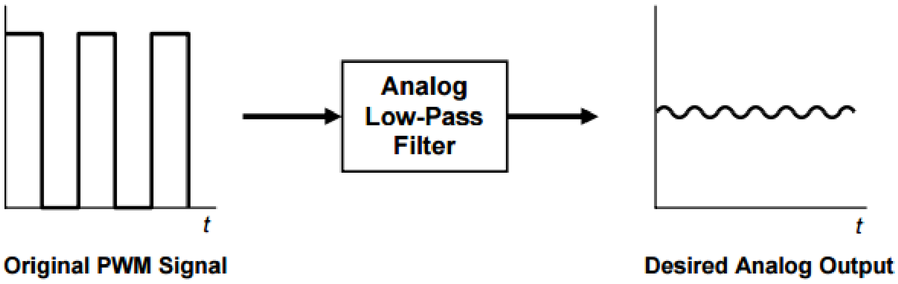


Fig.16 Analog filtering of PWM signal

The PWM/DAC approach is not new, but performance limitations have historically confined its use to low-resolution, low-bandwidth applications. The performance of the method relates directly to the ability of the low-pass filter to remove the high-frequency components of the PWM signal. Use a filter with too low a cut-off frequency, and DAC bandwidth suffers. Use a filter with too high a cut-off frequency or with slow stop-band roll-off, and DAC resolution suffers, but one way to alleviate both of these problems is to increase the frequency of the PWM. However, as PWM frequency increases on conventional microprocessor generated PWM, digital resolution problems begin to manifest.

3.4 ADC/DAC PROBLEMS

1)

The basic step of a 9-bit DAC is 10.3 mV. If 000000000 represents 0 V, what output is produced if the input is 101101111?

Solution

The output voltage for input 101101111 is

$$= 10.3\text{ mV }(1 \times 2^8 + 0 \times 2^7 + 1 \times 2^6 + 1 \times 2^5 + 0 \times 2^4 + 1 \times 2^3 + 1 \times 2^2 + 1 \times 2^1 + 1 \times 2^0)$$

$$= 10.3\text{ mV }(367)$$

$$= 3.78\text{ V}$$

2)

Calculate the values of the LSB, MSB and full scale output for an 8-bit DAC for the 0 to 10 V range.

Solution

$$\text{LSB} = \frac{1}{2^8} = \frac{1}{256}$$

For 10 V range, $\text{LSB} = \frac{10\text{ V}}{256} = 39\text{ mV}$

and $\text{MSB} = \left(\frac{1}{2}\right)\text{ full scale} = 5\text{ V}$

$$\text{Full scale output} = (\text{Full scale voltage} - 1\text{ LSB})$$

$$= 10\text{ V} - 0.039\text{ V} = 9.961\text{ V}$$

3)

What output voltage would be produced by a D/A converter whose output range is 0 to 10 V and whose input binary number is

(i) 10 (for a 2-bit D/A converter)
(ii) 0110 (for a 4-bit DAC)
(iii) 10111100 (for a 8-bit DAC)

Solution

(i) $V_o = 10\text{ V}\left(1 \times \frac{1}{2} + 0 \times \frac{1}{4}\right) = 5\text{ V}$

(ii) $V_o = 10\text{ V}\left(0 \times \frac{1}{2} + 1 \times \frac{1}{2^2} + 1 \times \frac{1}{2^3} + 0 \times \frac{1}{2^4}\right)$

$= 10\left(\frac{1}{4} + \frac{1}{8}\right) = 3.75\text{ V}$

(iii) $V_o = 10\text{ V}\,(1 \times 1/2 + 0 \times 1/2^2 + 1 \times 1/2^3 + 1 \times 1/2^4 + 1 \times 1/2^5 + 1 \times 1/2^6 + 0 \times 1/2^7 + 0 \times 1/2^8)$

$= 10\text{ V}\,(1/2 + 1/8 + 1/16 + 1/32 + 1/64) = 7.34\text{ V}$

4.PROBES AND CONNECTORS

A probe as used in the context of electronic instrumentation, is a device used to connect the input of a measurement instrument such as an oscilloscope or electronic voltmeter to a point in the circuit where the measurement is to be made. A probe may consist of alligator clip leads or it may be a complex circuit including amplifiers and other active components. Along with probes, we will consider certain other hardware that is frequently used to interconnect instruments.

4.1 TEST LEADS

A test lead may be a length of hook up wire with alligator clips at either end or it may have a special probe type end as shown below:

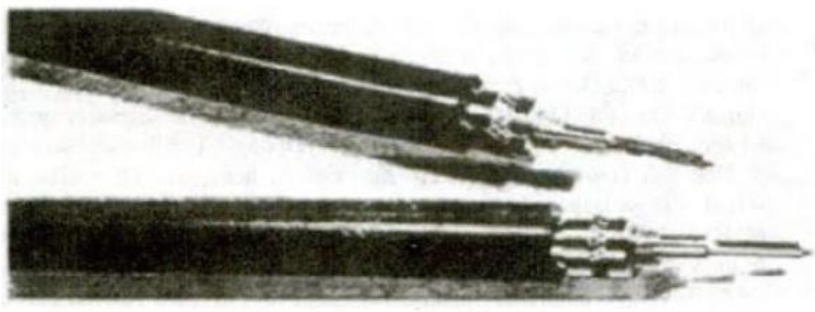


Fig.17 Test Probes.

The probe usually makes it easier to access points in the circuit and is generally safer to use in live circuits than the alligator clips. Besides electrical shock, there is always the possibility of accidentally slipping off the test point and damaging the circuit. Solid state components are notoriously unforgiving of such "goofs". Test leads wire is special type of hook up wire that is well insulated (500-600) V or more but is flexible enough to allow easy use in making measurements. The simple test lead is found most frequently in ac /dc voltmeter/ multimeter applications and in certain other applications where the ac frequency is low. At higher frequency the distributed capacitance and noise pickup are too great for accurate measurement. Following figure shows the mechanism by which interference can occur when open test leads are used.

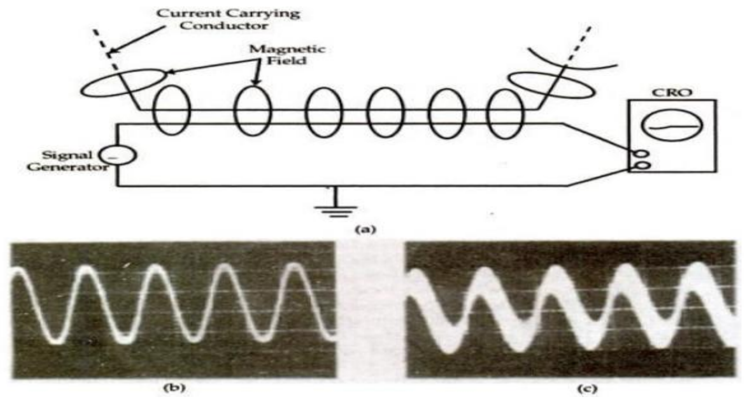


Fig.18 (a)pickup interference (b)reproduction of 100khz signal (c) reproduction with 60hz interference

A nearby conductor carrying an electric current generates a magnetic field and flux from this field cuts the signal wire including spurious signal current. Figure (b), (c) shows the original 100Khz signal and the interference effect happening due to the 60Hz signal. The trace on the oscilloscope used 100mV signal. Besides frequency, two other factors must be considered when using test leads, they are signal amplitude and circuit impedance. If the amplitude is high enough, then the artifacts caused by an interfering signal may be negligibly small. But the source and load impedances in the circuit affect the amplitude of the interfering signal. A possible solution is to use a twisted pair of leads as shown below:

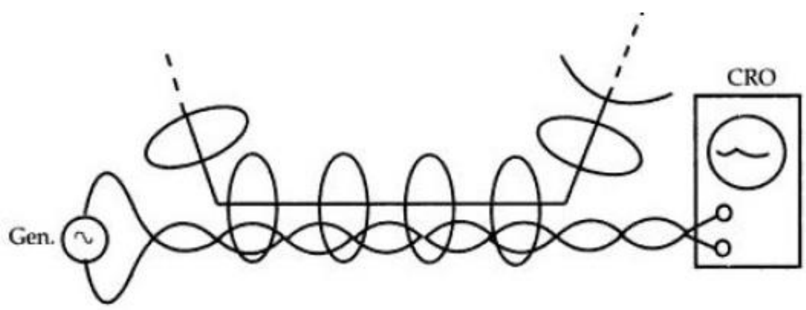


Fig.19 Twisted pair unshielded test leads.

Such test leads can be purchased or they can be made easier inserting one end of each of the two wires into the chuck of an electric drill and then anchoring the other ends. They help in (1) self-shielding and (2) magnetic field affects both conductors equally. The criteria for using simple test leads, then are low frequency of dc signals high signal amplitude and low source and load impedances.

4.2 SHIELDED CABLES

A shielded wire is a conductor that is surrounded by and insulated from another conductor.

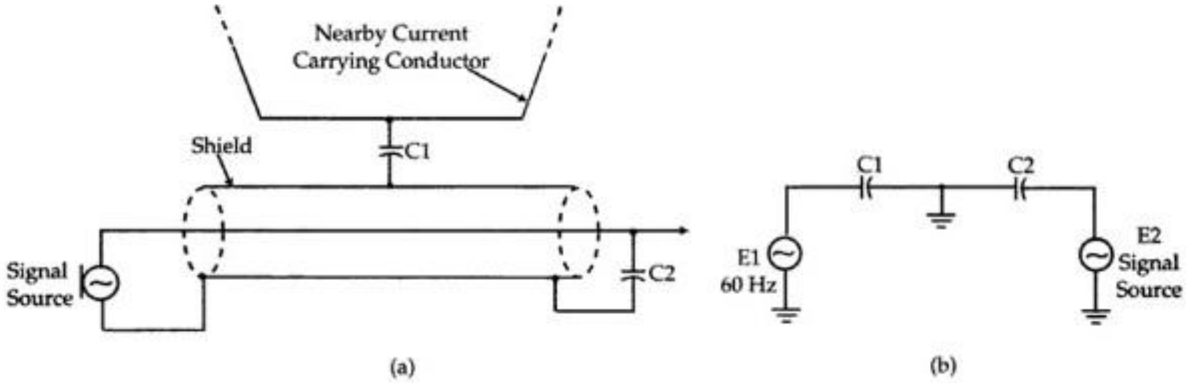


Fig.20 (a)Shielded test lead (b) Equivalent circuit

In most cases the outer conductor is a braided cylinder. Coaxial cable often used as a transmission line in RF circuits, meets the requirement of a shielded wire. The coupling between close proximity conductors is both inductive and capacitive. Conductor C_1 in above figure is the capacitance between the shield and a nearby current carrying conductor and it represents a path for interfering currents. Capacitor C_2 in figure represents the capacitance between the inner conductor and the shield. The two main reasons for shielding can reduce coupling. In some systems, it is due to the fact that C_1 and C_2 are in series, so the total capacitance is less than either capacitance alone. In unbalanced systems, one end of each capacitor is grounded that is at ground potential. The interfering potential usually does not get into signal circuit.

4.3 CONNECTORS

Quite a few connectors are used to interconnect electronic instrumentation and circuits and the probability that two instruments will have different connectors is directly proportional to the urgency of using them together. All electronic facilities seem to keep a rather large number of adapters that are used to allow interfacing instruments that have different inputs or output connectors.

Following figure shows several different connectors. Figure (a) shows banana plug and jack. These connectors are used on power supplies, audio generators and other low frequency instruments including multimeters and low frequency oscilloscopes. The PL-259 "UHF" connector is shown in figure (b). This connector mates with the S)-239 connector that is used as the antenna connector in CB and most other two -way radio transmitters. These connectors were once used extensively on signal generators and oscilloscopes built a decade or more in the past. Most equipment, however tends to use the BNC connector of figure (c). In current practice, the use of the UHF connector is limited to RF power applications that were fitted with the SO-239 mate to the PL-259 have long since obtained a BNC to UHF adapter so that modern cables and probes will fit. A collection of adapters is shown in figure (d) although there are literally dozens in the market.

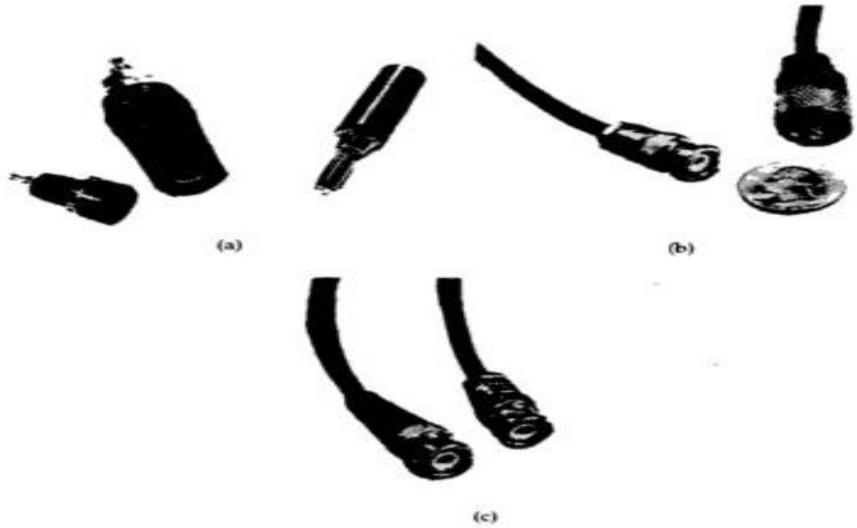


Fig.21 (a) Banana plug and two types of banana jack, (b) BNC (left) & UHF (right) connectors, (c) two types of BNC connectors

Fig.22 Connector Adapters

4.4 LOW CAPACITANCE PROBES

The basic low capacitance probe shown in figure below contains a parallel network consisting of a high value resistor and a small value capacitor.

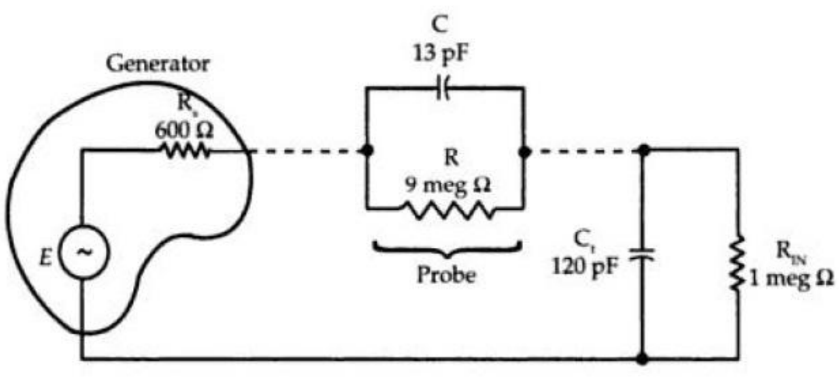


Fig.23 Equivalent instrument input circuit when a low-capacitance probe is used.

This type of probe is called as passive probe because it contains no amplifying devices. The probe shown above presents a/10 the capacitive load on the signal source as did the shielded cable and it provides a constant 10:1 voltage division ratio over a wide range of frequencies. following figure shows commercial available low-capacitance oscilloscope probe.

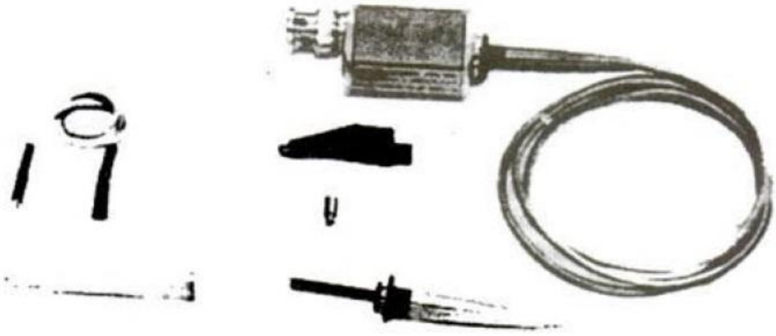


Fig.24 Typical low-capacitance oscilloscope probe

The usual designation for a 10:1 division probe is "10x", a practice that may tend to confuse. When a 10xprobe is used, the displayed voltage is 1/10 the actual voltage, so multiply the indicated voltage by a factor of 10. The RC combinations in the circuit can still cause a phase shift co some manufacturers place a variable trimmer capacitor in the circuit. In the low-cost instruments the capacitors inside the probe itself is made variable, whereas in most professional grade instruments, the variable capacitor is located inside the molded BNC connector that attaches to the instrument. This capacitor will eliminate most shift problems. Figure (b) below shows the normal waveform, while figure (c) shows the access hole for the adjustment capacitor in a Tektronix probe.

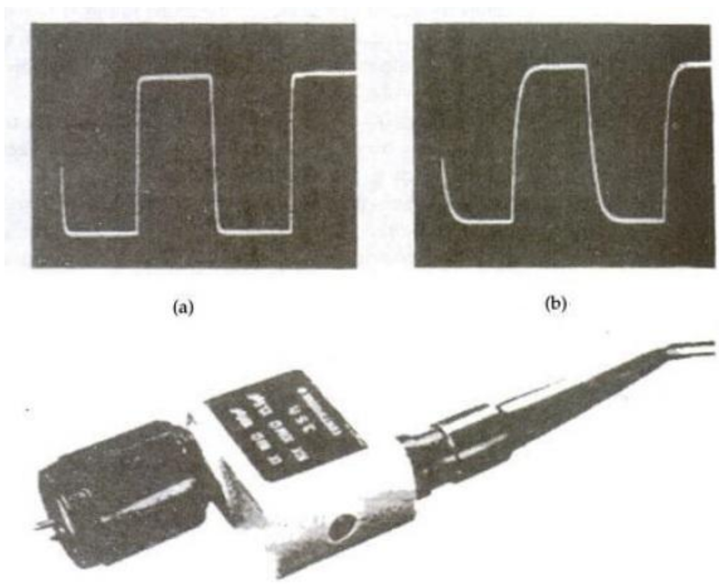


Fig.25 (a) Normal trace, (b) "C" misadjusted, (c) connector end of low capacitance probe showing access hole to compensating capacitor.

4.5 HIGH VOLTAGE PROBES

Very few oscilloscopes or voltmeters have maximum dc voltage range in excess of 1500V. But many rather ordinary circuits operate at potentials far in excess of this value, A color television receiver, for example may use up to approximately 25,000V on the post deflection accelerator anode. A high voltage probe is a special voltage divider as seen in figure below that allows these high potentials to be read on an ordinary voltmeter or oscilloscope.

Fig.26 High voltage divider probe

Most of these probes use the input resistance of the instrument and a 900 Ω resistor inside the probe as the voltage divider, although a few include the entire divider inside the probe. For electronic voltmeters with a 10M Ω input resistance, the divider ratio is 100:1, but for most oscilloscopes the ratio is 1000:1 because of lower input resistance of those instruments. Note in the above figure of high voltage divider probe, it is built of thick, insulating plastic to prevent electrical shock to the user. The finlike structures on the probe effectively lengthen the surface path between the tip of the probe and the users hand. Additionally high voltage probes such as shown above are not recommended for high altitude use. At high altitude use, the barometric pressure is lower, so the ionization potential of air is also lower. Equipment used at high altitudes for example mountain top radar stations or air borne equipment is specially designed for the application, and this type of probe defeats the special precautions taken by the manufacturer of the equipment.

4.6 CURRENT PROBES

The measurement of current by a current meter requires that the circuit be broken so that the meter vane be joined in series with the load. But this is not always either possible or desirable. An active probe such as one of the current to voltage converter circuits will allow the measurement of dc and low frequency ac on a voltage measuring instrument such as an oscilloscope or an electronic voltmeter. But the requirement for circuit still exists.

A passive current probe uses the magnetic field surrounding a current carrying conductor to measure the strength of the current in the conductor. Figure below shows the construction of such a probe using a toroid (ie., doughnut shaped) core that is wrapped with several turns of wire. The conductor carrying an alternating current forms the primary of the transformer while on the core forms the secondary. Alternating current in the conductor will induce current in the wire of the probe, so an output voltage E_0 will be generated and will be proportional to the current flowing.

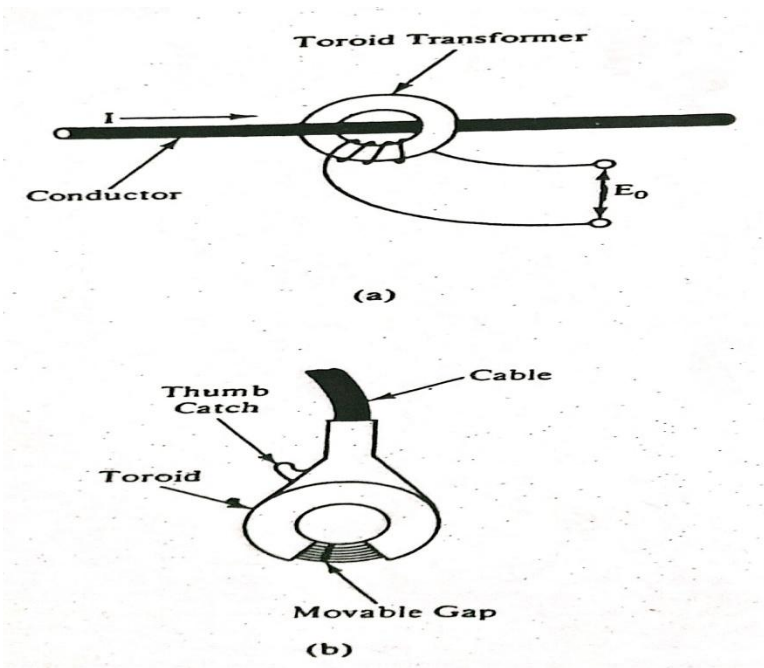


Fig.27 (a) current probe circuit, (b) construction.

Toroid cores present a problem as far as wrapping around a conductor is concerned. Figure (b) shows the constructional details of most typical probes. There is a gap in the toroid. A latch on the side of the core allows the user to spread the gap to admit the conductor. Most of these probes operate over a frequency range of 100Hz to 3mHz but require an output amplifier that produces an output scaled of mV/mA.

4.7 SPECIAL PROBES FOR ICs

Digital and linear integrated circuits (ICs) are not too forgiving for accidental shorts. If a probe should slip off an IC pin, a very frequent occurrence then a destructive short may be created. In addition to this problem, the extremely close spacing of IC DIP makes it difficult to attach normal probes. Most often recommended by equipment makers as a solution to this troublesome problem is the clip-on probe as shown in figure below

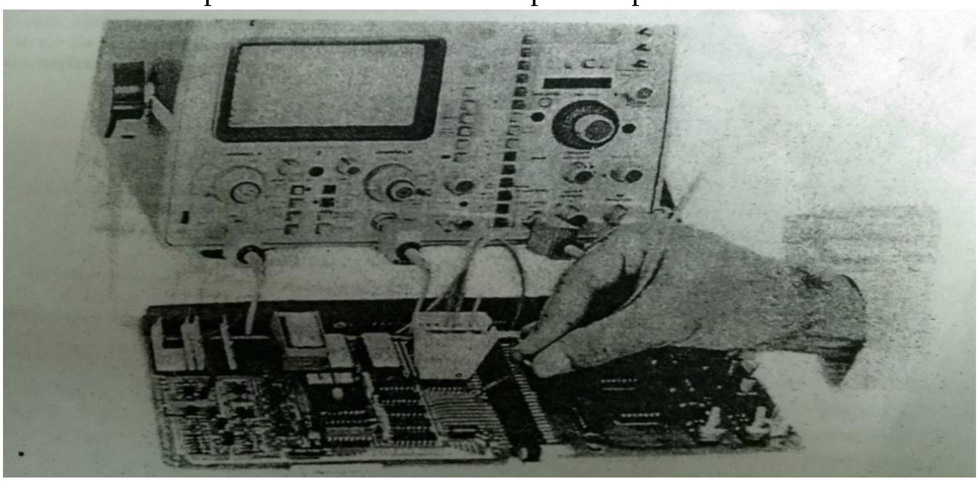


Fig.28 Clip-on probe for integrated circuits.

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SCHOOL OF ELECTRICAL AND ELECTRONICS

DEPARTMENT OF ELECTRONICS AND INSTRUMENTATION

UNIT – III

ELECTRONIC MEASUREMENT AND INSTRUMENTATION – SIC 1305

UNIT-III

ANALYZERS AND DIGITAL INSTRUMENTS

1.WAVE ANALYSER:

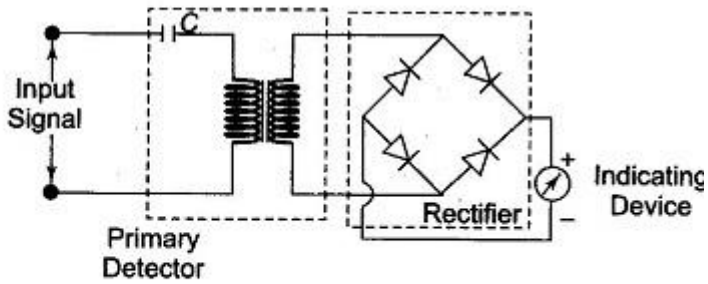


Fig.1 Basic Wave Analyzer.

It consists of a primary detector, which is a simple LC circuit. This LC circuit is adjusted for resonance at the frequency of the particular harmonic component to be measured. The intermediate stage is a full wave rectifier to obtain the average value of the input signal. The indicating device is a simple DC voltmeter that is calibrated to read the peak value of the sinusoidal input voltage. Since the LC circuit is tuned to a single frequency it passes only the frequency to which it is tuned and rejects all other frequencies. A number of tuned filters, connected to the indicating device through a selector switch would be required for a useful Wave Analyzer.

1.1 Frequency Selective Wave Analyzer:

The wave analyzer consists of a very narrow pass-band filter section which can be tuned to a particular frequency within audible frequency range (20Hz-20KHz). The block diagram of wave analyzer is shown below:

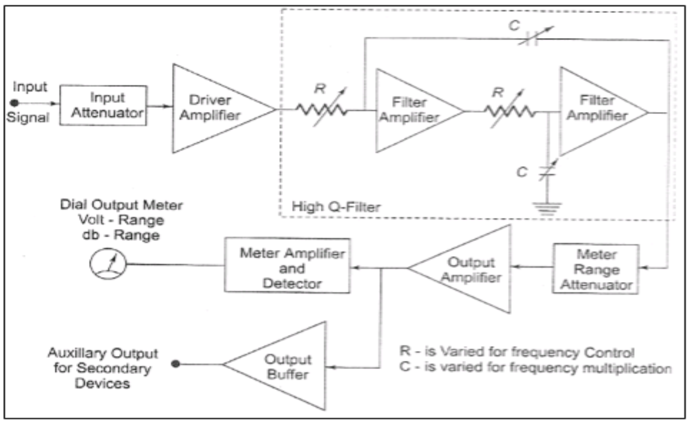


Fig.2 Frequency Selective Wave Analyzer.

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 frequency. 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 analyser is also called frequency selective voltmeter.

The entire AF range is covered in decade steps by switching capacitors in the RC section. Then selected signal output from the final amplifier stage is applied to the meter circuit and to an untuned amplifier. The main function of the buffer amplifier is to drive output devices such as recorders or electronic counters. The meter has several voltage ranges as well as decibel scales marked on it. It is driven by an average reading rectifier type decoder.

The wave analyser must have extremely low input distortion, undetectable by the analyser itself. The bandwidth of the instrument is very narrow, typically about 1% of selective band given by the following response characteristics.

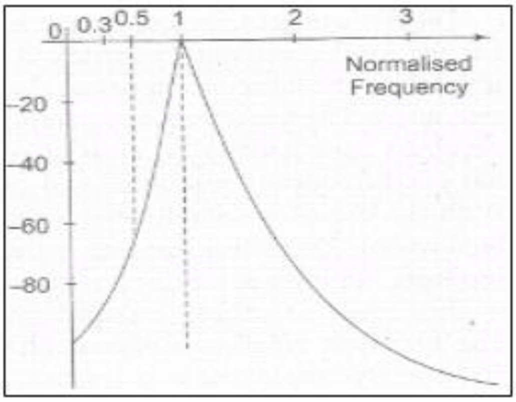


Fig.3 Relative response in Decibels (Dbs)

1.2 Heterodyne Wave Analyzer:

Wave analyzers are useful for measurement in the audio frequency range only. For measurements in the RF range and above (MHz range), an ordinary wave analyzer cannot be used. Hence, special types of wave analyzers working on the principle of heterodyning (mixing) are used. These wave analyzers are known as Heterodyne Wave Analyzer. In this wave analyzer, the input signal to be analyzed is heterodyned with the signal from the internal tunable local oscillator in the mixer stage to produce a higher IF frequency. By tuning the local oscillator frequency, various signal frequency components can be shifted within the pass-band of the IF amplifier. The output of the IF amplifier is rectified and applied to the meter circuit.

An instrument that involves the principle of heterodyning is the Heterodyning tuned voltmeter, shown in figure below:

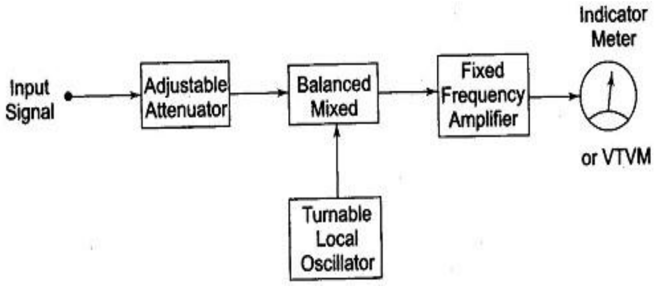


Fig.4 Heterodyne Wave Analyzer

The input signal is heterodyned to the known IF by means of a tunable local oscillator. The amplitude of the unknown component is indicated by the VTVM (vacuum tube voltmeter) or output meter. The VTVM is calibrated by means of signals of known amplitude. The frequency of the component is identified by the local oscillator frequency, i.e. the local oscillator frequency is varied so that all the components can be identified. The local oscillator can also be calibrated using input signals of known frequency. The fixed frequency amplifier is a multistage amplifier which can be designed conveniently because of its frequency characteristics. This analyzer has good frequency resolution and can measure the entire AF frequency range. With the use of a suitable attenuator, a wide range of voltage amplitudes can be covered. Their disadvantage is the occurrence of spurious cross-modulation products, setting a lower limit to the amplitude that can be measured.

Two types of selective amplifiers find use in Heterodyne wave analyzers. The first type employs a crystal filter, typically having a centre frequency of 50 kHz. By employing two crystals in a band-pass arrangement, it is possible to obtain a relatively flat pass-band over a 4 cycle range. Another type uses a resonant circuit in which the effective Q has been made high and is controlled by negative feedback. The resultant signal is passed through a highly selective 3-section quartz crystal filter and its amplitude measured on a Q-meter.

When a knowledge of the individual amplitudes of the component frequency is desired, a heterodyne wave analyzer is used.

A modified heterodyne wave analyzer is shown in figure below:

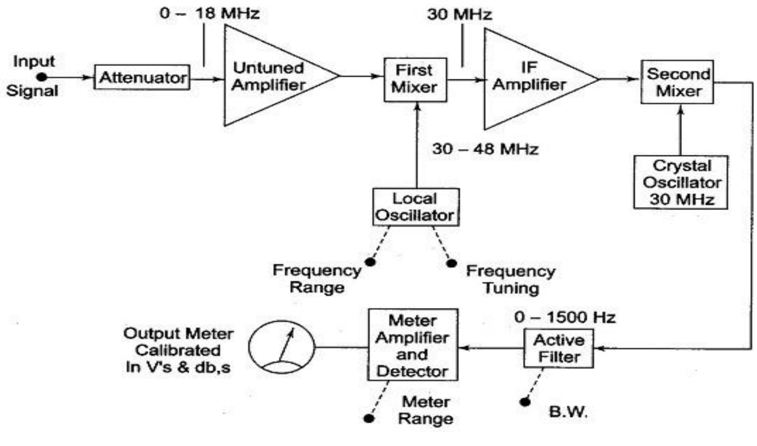


Fig.5 RF Heterodyne Wave Analyzer.

In this analyzer, the attenuator provides the required input signal for heterodyning in the first mixer stage, with the signal from a local oscillator having a frequency of 30 —48 MHz.

The first mixer stage produces an output which is the difference of the local oscillator frequency and the input signal, to produce an IF signal of 30 MHz. This IF frequency is uniformly amplified by the IF amplifier. This amplified IF signal is fed to the second mixer stage, where it is again heterodyned to produce a difference frequency or IF of zero frequency.

The selected component is then passed to the meter amplifier and detector circuit through an active filter having a controlled band-width. The meter detector output can then be read off on a db-calibrated scale, or may be applied to a secondary device such as a recorder.

This wave analyzer is operated in the RF range of 10 kHz — 18 MHz, with 18 overlapping bands selected by the frequency range control of the local oscillator. The bandwidth, which is controlled by the active filter, can be selected at 200 Hz, 1 kHz and 3 kHz.

2. SPECTRUM ANALYSER

The most common way of observing signals is to display them on an oscilloscope, with time as the X-axis (i.e. amplitude of the signal versus time). This is the time domain. It is also useful to display signals in the frequency domain. The instrument providing this frequency domain view is the spectrum analyzer.

A Spectrum Analyzer Block Diagram provides a calibrated graphical display on its CRT, with frequency on the horizontal axis and amplitude (voltage) on the vertical axis. Displayed as vertical lines against these coordinates are sinusoidal components of which the input signal is composed. The height represents the absolute magnitude, and the horizontal location represents the frequency. These instruments provide a display of the frequency spectrum over a given frequency band. Spectrum analyzers use either a parallel filter bank or a swept frequency technique. In a parallel filter bank analyzer, the frequency range is covered by a series of filters whose central frequencies and bandwidth are so selected that they overlap each other, as shown in figure below.

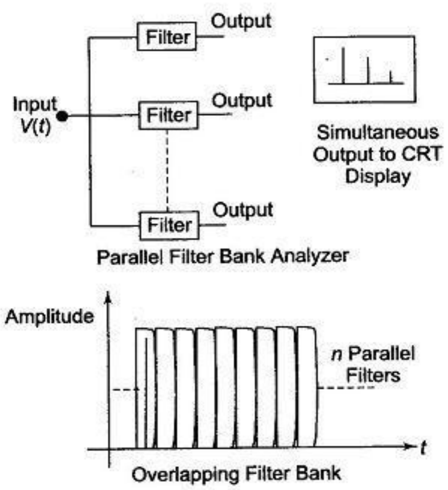


Fig.6 Spectrum Analyser (parallel filter bank analyser)

Typically, an audio analyzer will have 32 of these filters, each covering one third of an octave. For wide band narrow resolution analysis, particularly at RF or microwave signals, the swept technique is preferred.

Basic Spectrum Analyser using Swept Receiver Design:

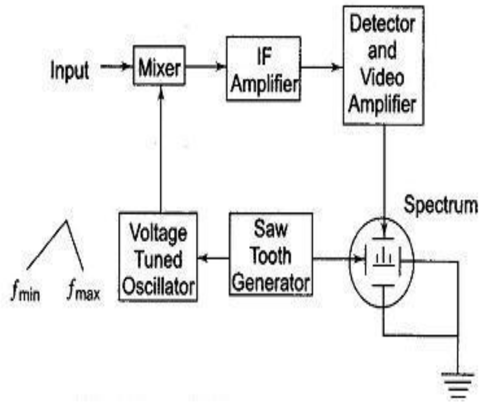


Fig.7 Spectrum Analyser

Referring to above block diagram, the sawtooth generator provides the sawtooth voltage which drives the horizontal axis element of the scope and this sawtooth voltage is frequency controlled element of the voltage tuned oscillator. As the oscillator sweeps from f_{min} to f_{max} of its frequency band at a linear recurring rate, it beats with the frequency component of the input signal and produce an IF, whenever a frequency component is met during its sweep. The frequency component and voltage tuned oscillator frequency beats together to produce a difference frequency, i.e. IF. The IF corresponding to the component is amplified and detected if necessary, and then applied to the vertical plates of the CRO, producing a display of amplitude versus frequency.

The spectrum produced if the input wave is a single tuned A.M. is given in figure below:

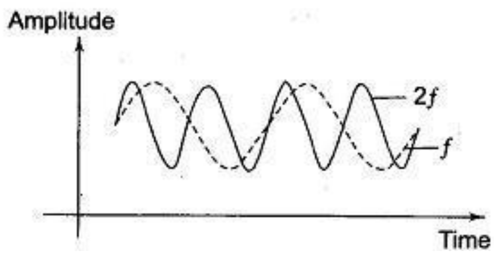


Fig 8a Test wave seen on ordinary CRO

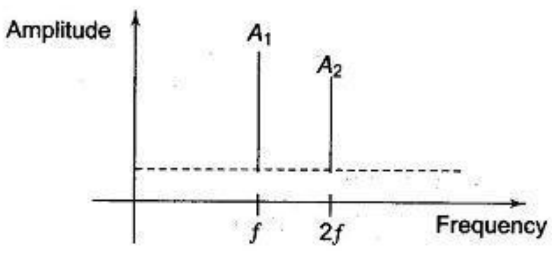


Fig.8b Display on the spectrum CRO

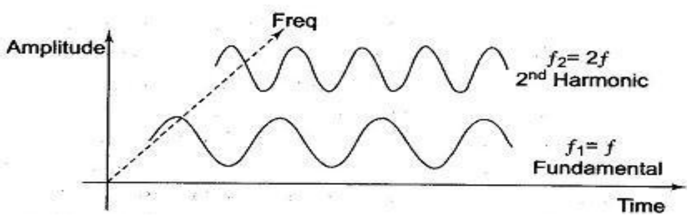


Fig.9 Test waveform as seen on X-axis (time) and Z-axis (frequency)

One of the principal applications of spectrum analyzers has been in the study of the RF spectrum produced in microwave instruments. In a microwave instrument, the horizontal axis can display as a wide a range as 2 — 3 GHz for a broad survey and as narrow as 30 kHz, for a highly magnified view of any small portion of the spectrum. Signals at microwave frequency separated by only a few kHz can be seen individually.

The frequency range covered by this instrument is from 1 MHz to 40 GHz. The basic block diagram is of a spectrum analyzer covering the range 500 kHz to 1 GHz, which is representative of a superheterodyne type

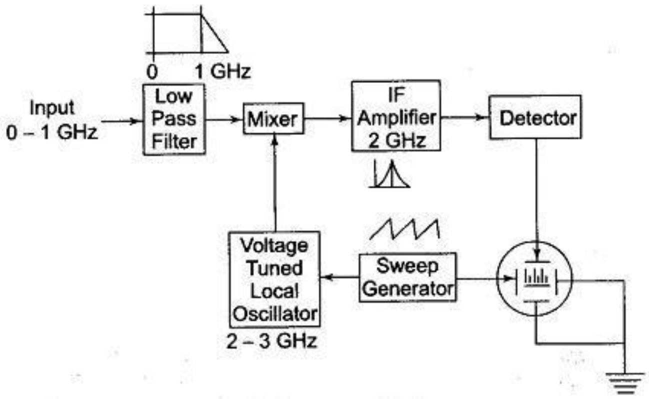


Fig.10 RF Spectrum Analyser

The input signal is fed into a mixer which is driven by a local oscillator. This oscillator is linearly tunable electrically over the range 2 — 3 GHz. The mixer provides two signals at its output that are proportional in amplitude to the input signal but of frequencies which are the sum and difference of the input signal and local oscillator frequency.

The IF amplifier is tuned to a narrow band around 2 GHz, since the local oscillator is tuned over the range of 2 — 3 GHz, only inputs that are separated from the local oscillator frequency by 2 GHz will be converted to IF frequency band, pass through the IF frequency amplifier, get rectified and produce a vertical deflection on the CRT.

From this, it is observed that as the saw tooth signal sweeps, the local oscillator also sweeps linearly from 2 — 3 GHz. The tuning of the spectrum analyzer is a swept receiver, which sweeps linearly from 0 to 1 GHz. The saw tooth scanning signal is also applied to the horizontal plates of the CRT to form the frequency axis. (The Spectrum Analyzer Block Diagram is also sensitive to signals from 4 — 5 GHz referred to as the image frequency of the super heterodyne. A low pass filter with a cutoff frequency above 1 GHz at the input suppresses these spurious signals.) Spectrum analyzers are widely used in radars, oceanography, and bio-medical fields.

3. DISTORTION ANALYSER

Harmonic Distortion Analyser-Fundamental Suppression Type:

A Harmonic Distortion Analyzer measures the total harmonic power present in the test wave rather than the distortion caused by each component. The simplest method is to suppress the fundamental frequency by means of a high pass filter whose cut off frequency is a little above the fundamental frequency. This high pass allows only the harmonics to pass and the total harmonic distortion can then be measured. Other types of Harmonic Distortion Analyzer based on fundamental suppression are as follows.

a)Employing a Resonance Bridge type:

The bridge shown in Figure below. This bridge is balanced for the fundamental frequency, i.e. L and C are tuned to the fundamental frequency. The bridge is unbalanced for the harmonics, i.e. only harmonic power will be available at the output terminal and can be measured. If the fundamental frequency is changed, the bridge must be balanced again. If L and C are fixed components, then this method is suitable only when the test wave has a fixed frequency. Indicators can be thermocouples or square law VTVMs (vacuum tube voltmeters). This indicates the rms value of all harmonics. When a continuous adjustment of the fundamental frequency is desired, a Wien bridge system is employed.

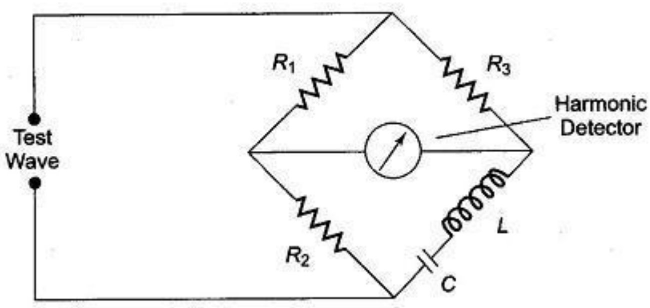


Fig.11 Resonance Bridge

b)Wien's Bridge method:

The bridge is balanced for the fundamental frequency. The fundamental energy is dissipated in the bridge circuit elements. Only the harmonic components reach the output terminals. The harmonic distortion output can then be measured with a meter. For balance at the fundamental frequency, C_1,C_2,C, R_1=R_2=R,R_3=2R_4.

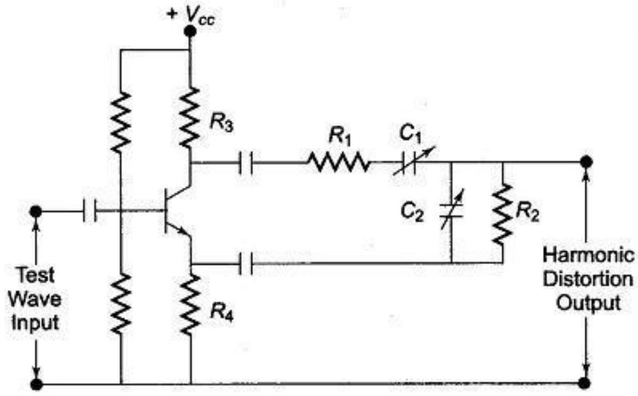


Fig.12 Wien's Bridge method.

c)Bridged T-Network:

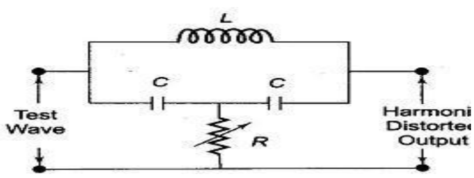


Fig.13 Bridge T-Network method.

Referring to figure below, the L and C's are tuned to the fundamental frequency, and R is adjusted to bypass fundamental frequency. The tank circuit being tuned to the fundamental frequency, the fundamental energy will circulate in the tank and is bypassed by the resistance. Only harmonic components will reach the output terminals and the distorted output can be measured by the meter. The Q of the resonant circuit must be at least 3-5.

One way of using a bridge T-network is given in figure below. The switch S is first connected to point A so that the attenuator is excluded and the bridge T-network is adjusted for full suppression of the fundamental frequency, i.e. minimum output. Minimum output indicates that the bridged T-network is tuned to the fundamental frequency and that the fundamental frequency is fully suppressed.

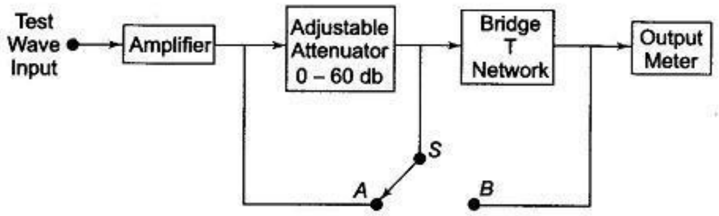


Fig.14 Harmonic Distortion Analyser using Bridged-T-Network.

The switch is next connected to terminal B**,** i.e. the bridged T-network is excluded. Attenuation is adjusted until the same reading is obtained on the meter. The attenuator reading indicates the total rms distortion. Distortion measurement can also be obtained by means of a wave analyzer, knowing the amplitude and the frequency of each component, the Harmonic Distortion Analyzer can be calculated. However, distortion meters based on fundamental suppression are simpler to design and less expensive than wave analyzers. The disadvantage is that they give only the total distortion and not the amplitude of individual distortion components.

4. MEASUREMENT OF FREQUENCY AND TIME

Frequency is the number of events per unit time. In the context of electric circuits the "events" are cycles of alternating current or pulses, and the unit time is the second or a subunit such as milliseconds, microseconds or nanoseconds. The unit of frequency is Hertz (Hz) and 1Hz is equal to 1 cycle per second. In many applications where the Hz is too small Kilohertz is used. The frequency of a pulse train is the number of pulses that occur in one second. Period and duration are both time measurements, but they slightly different. Period is defined as the time interval between identical features or points on successive waveforms. Frequency and period in repetitive wave trains are related by the expression:

$F=1/T$

where F=Frequency in hertz (Hz)

T= period in seconds (s)

4.1 DECIMAL COUNTING ASSEMBLIES

Decimal counting Unit (DCU):

The heart of any electronic counter is a circuit called the decimal counting unit, or DCU. All DCUs, whether discrete or integrated, consist of a decade counter, and a decoder/display driver, and most also include a data latch.

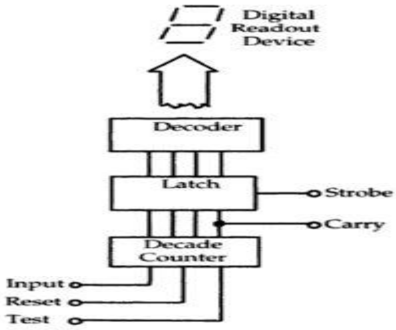


Fig.15 Decade or decimal Counter

An example of a DCU is shown in block form in Figure above.

This circuit has been widely used and is constructed of TTL integrated circuit logic elements. The decade counter uses a chain of flip-flops but it also contains some internal gates that sense the tenth state (i.e., "9") and reset the counter to 0000 if another input pulse is received.

Very few designers now use individual flip-flops to form decade counter circuits, because there are several complete IC decade counters available in both TTL and CMOS lines. In fact, there are several MSI and LSI integrated circuits available that contain all circuitry needed to form a complete electronic counter of several decades.

The four output lines from the decade counter in figure above are coded in the 8421 BCD format, and they change state with every pulse. There are also two other lines: input and reset. The input line accepts the pulses being counted, while the reset line will reset the counter to 0000 when brought HIGH. For normal counting the line is kept LOW, and to clear the counter it is brought momentarily HIGH.

The decoder is used to convert the 4-bit BCD code to the code needed to correctly drive the readout device. The counter outputs can be used to drive the decoder directly, but this results in a difficult-to-read "rolling" display as the number changes. The rolling effect is created because the display will change with the count. For example, if the count is to be "8," then the display will read "O-1-2-3-4-5-6-7-8" in sequence as the count accumulates 10 the "8" state. The rolling effect can be eliminated by using a quad latch circuit. A typical latch is a bank of four cascade type-D flip-flops connected so that their clock terminals are tied together to become a strobe line. If the strobe line is high, then data appearing on the input lines are transferred to the output lines. But if the strobe line is low, then the output lines retain the last data present at the input when the strobe was high.

In normal practice the strobe line is held low while the count is being accumulated, and at the end of the count period the strobe line is brought high momentarily. just long enough to transfer the count data from the input lines to the output lines. These data will then be held on the output lines until another strobe pulse updates the output to reflect the result of the next count.

Decimal Counting Assemblies:

The individual DCU can count only from 0 to 9, but if two DCUs are used in cascade, the assembly can count from 0 to 99, and three DCUs can count from 0 to 999 (etc.). Figure below shows a basic decimal counting assembly (DCA) consisting of three DCU elements. Three DCUs allow counting to 999, and one more DCU must be added to the assembly in cascade for each additional order of magnitude (i.e., for 0 to 9999 use four DCUs; for 0 to 99,999 use five DCUs, etc.).

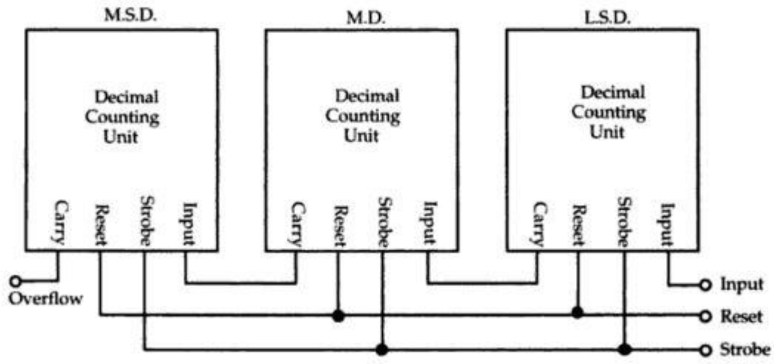


Fig.16 Decimal Counting Assemblies (DCAs)

The strobe and reset lines from all DCUs are tied to common strobe and reset lines that serve the entire DCA. All three DCUs in the assembly will be affected simultaneously by pulses applied to these lines. The carry output of a DCU is the "D" data output line in any given DCU (i.e., the line weighted "8"), and it is used to drive the input of the next DCU in the cascade chain. Recall from your digital electronics studies that the J-K flip-flops used in decade counters change state only on a negative-going transition of the clock terminal. The BCD line that is weighted "8" is HIGH for both "8" and "9" states, but it drops LOW when the decade counter overflows on the "10" count. It therefore toggles the input flip-flop of the next DCU only following the tenth count. This same tenth count also returns the counter to the "0" state. The original signal to be counted, then, is applied to the input of the DCU that is in the least significant digit position in the DCA (LSD in figure above). The carry output of the LSD DCU is applied to the input of the next-digit DCU, and finally the output of that DCU is applied to the input of the most significant stage (i.e., MSD). The output of the MSD stage can then serve as a counter overflow indicator.

A counter constructed of only a basic DCA is called a totalizer; it accumulates input pulses as long as it is turned on and not reset. The DCA will simply totalize counts either until it is turned off or until a reset pulse is applied. If the DCA overflows, the count starts back at zero but continues to increment as pulses are received at the input. When using a totalizer counter, it is sure that the number of events (i.e., input pulses) expected is less than the maximum number that the counter will accept without overflowing.

5.FREQUENCY COUNTER

A frequency counter measures the frequency of an input signal. These are commonly used in laboratories, factories and field environments to provide direct frequency measurements of various devices.

Most frequency counters work by using a counter that accumulates the number of events (oscillations) occurring within a specific period of time (say, one second). After the preset time period, the value in the counter is transferred to a display and the counter is reset to zero. Figure below shows the basic block diagram for a frequency counter. The sections include the DCA, main gate, trigger, input amplifier, main gate flip-flop, time base, and display clock.

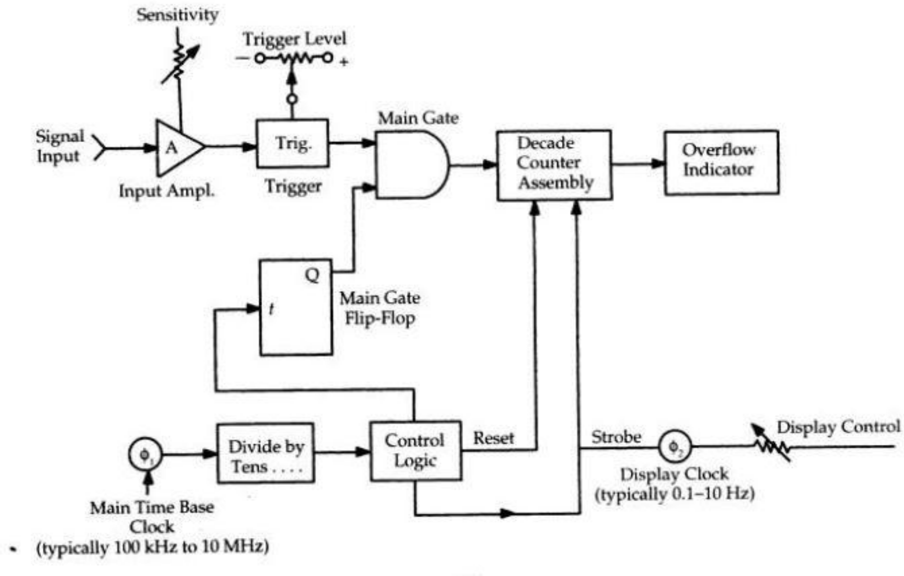


Fig.17 Block diagram of Frequency Counter

The DCA is a totalizer counter as shown in figure above. The overflow stage is a flip-flop that is SET when the MSD carry output goes high. The overflow flip-flop turns on a lamp to make the operator aware of the overflow condition so that the data can be disregarded.

A frequency counter measures events per unit of time (EPUT) (i.e., cycles per second), so the DCA must be turned ON only for a given period of time (e.g., 0.1, 1, or 10 s). The main gate, main gate flip-flop, and time base sections are used to allow input pulses into the DCA for the designated period of time.

The time base section consists of a crystal oscillator that produces pulses at a precise rate such as 100 kHz, 1 mHz, 4 mHz, 10 mHz, and so on. A chain of decade dividers is used to reduce the crystal oscillator frequency to a lower frequency. The time base output frequency will be 10 Hz for 0.1-s, 1 Hz for 1-s, and 0.1 Hz for 10-s measuring periods.

The timing diagram for one complete interval of an EPUT counter is shown in fig below. Pulses t_1, t_2, and t_3 are output from the time base section.

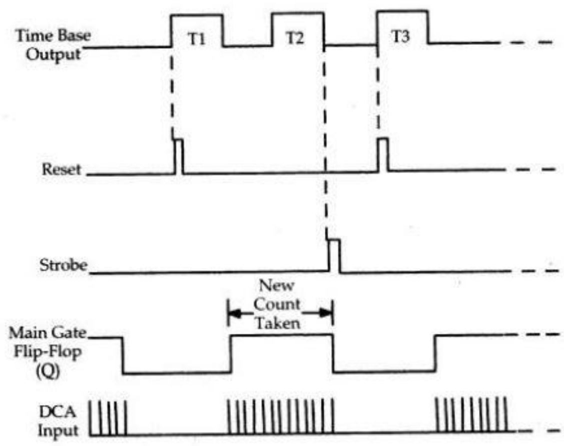


Fig.18 Timing diagram of counter

When pulse t_1 goes high, the control logic section generates a short pulse to reset the DCA to zero. When t_1 goes low again, the output of the J-K main gate flip-flop will go high.

The main (AND) gate has one input tied to the output of the flip-flop, and the other input is tied to the signal being counted. As a result, the main gate passes input pulses to the DCA only when the Q terminal of the flip-flop is high.

The flip-flop remains set until the negative transition of t_2 occurs. At that time the output of the flip-flop drops low, turning off the flow of pulses into the DCA and causing the control logic section to generate a strobe pulse. This pulse tells the DCU latches to transfer data from the counter to the decoders. The display. then, shows only completed count cycles and will hold the previous count until the end of the next interval. The frequency counter of figure counts frequency (.e., events per unit of time) because the DCA is enabled only for a specific unit of time. The frequency of the input signal is the number of counts accumulated on the DCA divided by the time base period in seconds.

frequency (Hz) = counts on DCA/time (s)

5.1 PERIOD COUNTER:

Period is defined as the time elapsing between identical features on successive cycles of a waveform. Period can be calculated from the frequency and is the reciprocal of frequency:

Period = P =1/f(hz)

A period counter can be made by reversing the roles of the input stage and time base; the input amplifier is connected to the main gate flip-flop, and the time base is connected to the main gate.

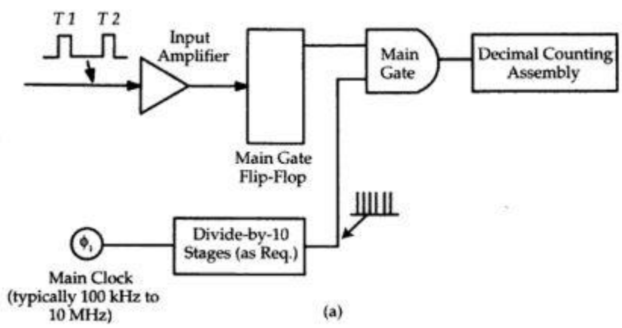


Fig.19 Block diagram of Period counter

The timing diagram for a period counter is shown in figure. The flip-flop is set (i.e., Q goes high) on the negative-going transition of t_l, and this allows the gate to pass pulses from the time base to the DCA (Decimal counting assembly)

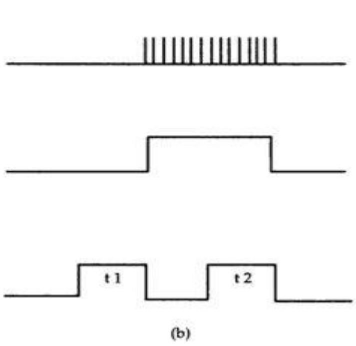


Fig.20 Timing diagram of Period counter

The time base frequency determines the time interval represented by each pulse. For example, a time base frequency of 1000 Hz yields a period counter resolution of 1/1000 s. or 1 ms. Similarly, one 10-kHz time base yields 0.1-ms resolution. The period is given by the count accumulated by the DCA and the time base frequency.

Period, P= DCA Count/f(Hz)

Resolution is the smallest time interval that can be measured on the counter and it is defined as the reciprocal of the time base frequency (1/f).

5.2 COUNTER ERRORS

Several factors tend to reduce the accuracy of an electronic counter, and these can be grouped as **inherent errors and signal-related errors**. The inherent errors are a function of the quality, age, and history of the individual counter. Little can be done about these errors unless their source is a serious need for recalibration of the time base. Signal-related errors, on the other hand, are often correctable by proper manipulation of sensitivity, trigger level, and trigger amplitude controls.

Inherent Errors:

The time base error is expressed in terms of a percentage, or in parts per million. The error from time base inaccuracies is directly reflected in all measurements of frequency or period. For example, suppose a 1-mHz time base is off by 30 Hz; that is, it is actually 1,000,030 Hz instead of 1,000,000 Hz. This is an error of 30 parts per million (30 ppm), which in percent is

$$\frac{1{,}000{,}030 - 1{,}000{,}000}{1{,}000{,}000} \times 100 = 0.003\%$$

The measurement error due to time base inaccuracy is constant regardless of the frequency being measured. That is, there will be a 0.003% error at 1 kHz and the same 0.003% error at the maximum frequency that the device will measure. For example, a 27-MHz signal would be measured with an error of

$$27\text{ mHz} \times \frac{30\text{ Hz}}{\text{mHz}} = 810\text{ Hz}$$

If the time base frequency is 30 Hz high, then the counter reading will be low, and if the time base frequency is low, then the counter reading will be high. Total time base inaccuracy is the sum of several individual errors: **initial error, short-term stability, long-term stability, temperature change, and line voltage change.**

The **initial error** is the calibration error at the time the time base is initially adjusted at the factory or at recalibration in a metrology laboratory. Different methods are used to measure the time base frequency. In many cases the time base oscillator frequency is compared with standard frequency broadcasts of the National Bureau of Standards radio stations WWV, WWVB, or WWVH. Alternatively, it might be compared with the output of a cesium or rubidium beam atomic clock in a metrology calibration laboratory. For high-accuracy time bases, the latter is preferred, although 60 kHz WWVB comparator receivers are very good standards.

The **short-term stability** is the time base oscillator frequency drift per day. **Long-term stability** is the frequency drift per month and is often designated the aging rate. The temperature **and line voltage stability specifications** refer to the frequency change over the 0°C-50°C temperature range and to the ± 10% line voltage change, respectively.

The ±1 count ambiguity is caused by the lack of synchronization between the input signal and time base, as illustrated in figure. During period t_1, ten pulses are gated into the DCA, while during t_2 only nine pulses reach the DCA. On some subsequent count, it may be that nine pulses are gated into the DCA. One fundamental rule for all digital counter instruments is that there is an error of ± 1 count of the least significant digit. In other words, a counter that reads, say, 10,000 Hz is measuring a frequency that lies between 9999 Hz and 10,001 Hz (i.e., 10 kHz + 1 Hz).

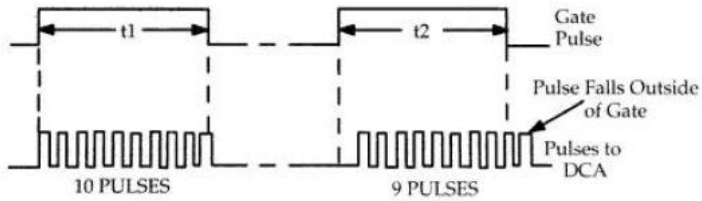


Fig.21 Mechanism for ± 1 count ambiguity occurring in counters.

The ± 1 count ambiguity causes an error that is inversely proportional to the frequency being measured and the gate time.

$$\textbf{Error (\%)} = \pm 100/fT$$

where,

f = frequency being measured in seconds (s)

T = Time when the gate is open.

5.3 SIGNAL RELATED ERRORS

Poor signal quality can introduce errors that add to or subtract from the true count. Most of these errors result from hysteresis crossing errors or from noise on the signal.

Figure below shows how severe ringing on a signal can create extra, spurious counts of the DCA if the trigger level control is adjusted so that the ringing portions of the signal cross the limits, creating two additional "input" pulses, a two-count error.

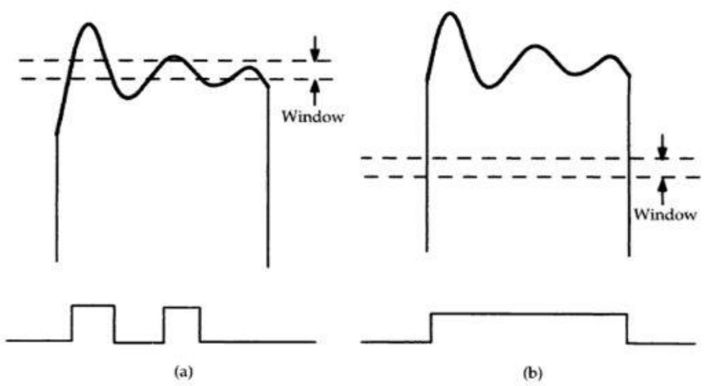


Fig.22 (a) Ringing causing false counts, (b) Proper adjustment of trigger to avoid problem

The remedy is to adjust the trigger level controls so that the ringing portions of the waveform fall outside the window limits. The same problem exists on sine wave input waveforms (Figure below) that have a large amount of harmonic distortion. The cure is the same, however: readjust the trigger level control so that it is operating over a lower portion of the waveform.

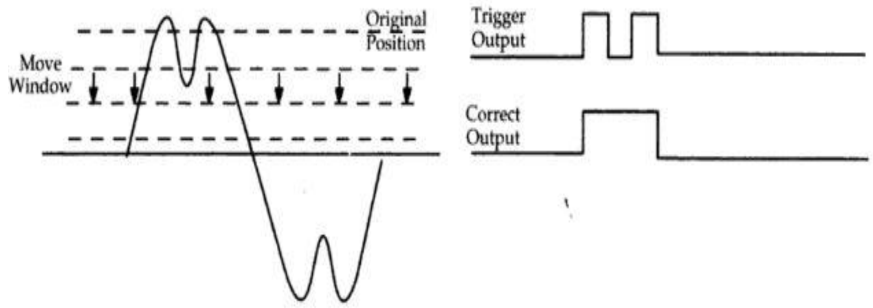


Fig.23 Second harmonic distortion causing false counts.

Similarly, impulse noise riding on the signal can have an amplitude sufficient to cross both limits of the hysteresis window. An example of this phenomenon is shown in figure below, in which a pulse in a symmetrical wave train is carrying impulse noise artifacts.

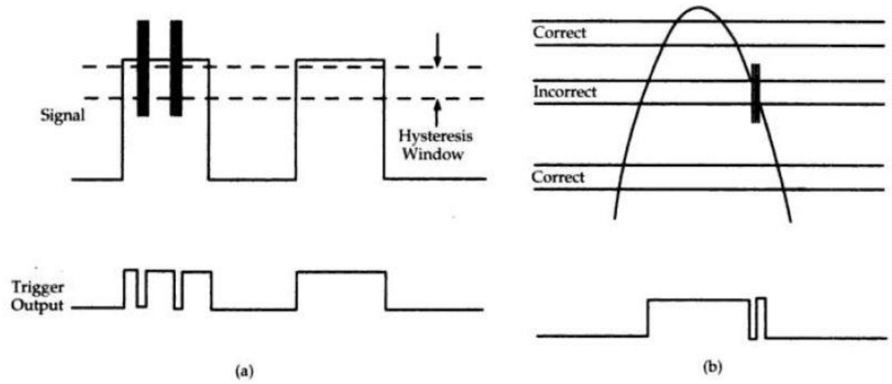


Fig.24 (a) Oscillations on waveforms causing false counts (b) correct & incorrect settings on the trigger of signals with noise or oscillations.

In the case shown, the two noise bursts cross the window limits and thereby force the trigger output to create three pulses instead of just one.

Once again the correction requires re-adjustment of the trigger level control to a point further down the waveform. In the case of a non-square wave, the noise may appear on the leading or trailing edges and still cause the problem. Figure (b) above shows the correct and incorrect positions for the hysteresis window on such a waveform. Note that filtering of the noise is usually not feasible because of the bandwidth requirements of the input amplifier.

Figure below shows a type of error, caused by noise, that is particularly troublesome on period measurements. In this example noise rides on a signal that has a shallow slope, thus creating a band of uncertainty around the signal.

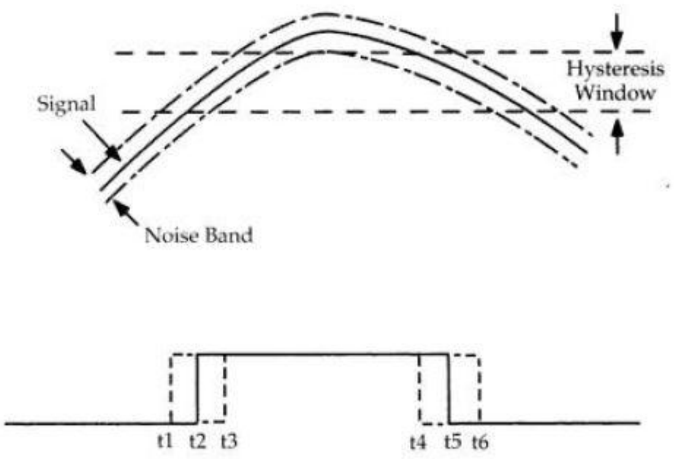


Fig.25 Counter Errors due to hysteresis window and noisy signal.

The trigger circuit should produce a high output when the signal crosses the upper limit, and should drop low again when the signal crosses the lower limit. But noise impulses adding to, or subtracting from the signal amplitude could provide premature, or delayed, trigger transitions. The correct duration of the trigger output pulse in above figure is t_4-t_1. But under worst-case conditions the actual duration may be as much as t_5 -t_0, and that amount represents a considerable error.

The solution for this problem is to cause the signal to slew through the hysteresis band as rapidly as possible. Two methods can be used to implement this solution. One is to narrow the window by adjusting the trigger amplitude control, whereas the other is to increase the waveform's slope by pre-amplification.

On some types of signal waveforms it is sufficient to adjust the trigger level control so that the counter triggers on the steepest portion of the waveform. On sine waves, for example, this point occurs at zero crossings, but on other waveforms it may occur elsewhere on the signal.

6. DIGITAL VOLTMETERS

Digital Voltmeters (DVMs) are measuring instruments that convert analog voltage signals into a digital or numeric readout. This digital readout can be displayed on the front panel and also used as an electrical digital output signal.

Any DVM is capable of measuring analog dc voltages. However, with appropriate signal conditioners preceding the input of the DVM, quantities such as ac voltages, ohms, dc and ac current, temperature, and pressure can be measured. The common element in all these signal conditioners is the dc voltage, which is proportional to the level of the unknown quantity being measured. This dc output is then measured by the DVM.

DVMs have various features such as speed, automation operation and programmability. There are several varieties of DVM which differ in the following ways:

1. Number of digits
2. Number of measurements
3. Accuracy
4. Speed of reading
5. Digital output of several types.

The DVM displays ac and dc voltages as discrete numbers, rather than as a pointer on a continuous scale as in an analog voltmeter. A numerical readout is advantageous because it reduces human error, eliminates parallax error, increases reading speed and often provides output in digital form suitable for further processing and recording. With the development of IC modules, the size, power requirements and cost of DVMs have been reduced, so that DVMs compete with analog voltmeters in portability and size.

Types of Digital Voltmeter:

1)Ramp type digital voltmeter

2)Integrating type digital voltmeter

3)Servo Potentiometer type digital Voltmeter

4) Successive Approximation type digital Voltmeter

They are discussed below:

6.1 RAMP TYPE DIGITAL VOLTMETER:

It is also called as single slope DVM or pulse width type DVM.

Principle: Voltage is converted into time, and the time period is measured with an electronic counter. The reading is displayed as voltage after processing. The time taken by a linear ramp voltage to rise from 0V to the level of the input voltage or to decrease from the level of the input voltage to zero is measured with a counter. This time interval is proportional to the voltage to be measured. The waveforms shown in figure below indicate the method. At the start of the measurement cycle, a ramp voltage is initiated. This voltage can be positive going or negative going.

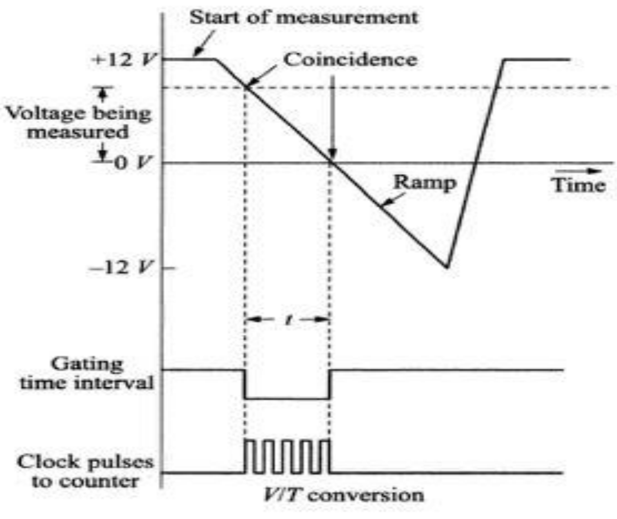


Fig.26 Ramp type voltmeter graph

In the figure, a negative-going ramp is shown. It is continuously compared with the unknown input voltage. At the instant that the ramp voltage equals the unknown voltage, a comparator or coincidence circuit generates a pulse. This pulse opens a gate. The ramp voltage continues to decrease with time until it finally reaches 0 V. A second comparator generates an output pulse that closes the gate. An oscillator generates clock pulses that are allowed to pass through the gate to a number of decade-counting units (DCUs). They totalise the number of pulses passed through the gate. The decimal number displayed by the indicator tubes associated with DCUs is a measure of the magnitude of the input voltage. A block schematic of the ramp-type DVM is shown below.

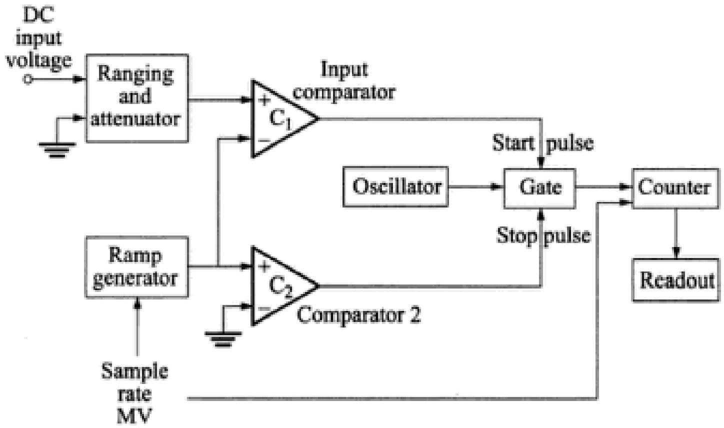


Fig.27 Ramp type Digital voltmeter block diagram

The DC input voltage to be measured is first given to the ranging and attenuator section. If the magnitude of the voltage is large, it is attenuated. If it is small, it is amplified. The sample rate multivibrator determines the slope of the ramp. It also determines the period of measurement or the duration of the measurement cycle, which can be in the range from a few cycles/sec to 1000 cycles/ sec. The sample rate circuit provides an initiating pulse for the ramp generator to start its next ramp voltage. At the same time, a reset pulse is generated, which returns all the DCUs to their zero state. The measuring cycle involves

1. Sampling the input and holding it at the same value 2. Measuring the value of the input.

3. Display. The display remains at the precision value till the next measuring cycle is completed so that there is no flicker in the display and the user finds continuous display. The ramp voltage is compared with the input voltage and when the equality is reached, a start pulse is applied to the logic gate. If it is the AND gate, it is enabled. The pulses from the clock oscillator are counted in the counter. When the ramp voltage becomes zero, the ground comparator senses the same and the AND gate is disabled. The counting will stop.

The number of pulses counted during this time interval is a measure of the input voltage. These counts are totalised and indicated as voltage in the readout.

6.2 INTEGRATING TYPE DIGITAL VOLTMETER:

The principle of operation of an Integrating Type DVM is illustrated in Figure.

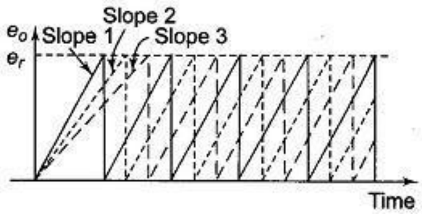


Fig.28 Voltage to frequency conversion

A constant input voltage is integrated and the slope of the output ramp is proportional to the input voltage. When the output reaches a certain value, it is discharged to 0 and another cycle begins. The frequency of the output waveform is proportional to the input voltage. The block diagram is illustrated in figure below:

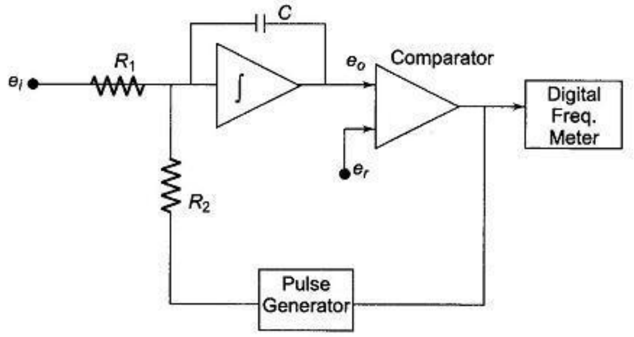


Fig.29 Block Diagram of Integrating type DVM

The input voltage produces a charging current, e_i/R_1 that charges the capacitor 'C' to the reference voltage e_r. When e_r is reached, the comparator changes state, so as to trigger the precision pulse generator. The pulse generator produces a pulse of precision charge content that rapidly discharges the capacitor. The rate of charging and discharging produces a signal frequency that is directly proportional to e_i. The voltage-frequency conversion can be considered to be a dual slope method, as shown in figure below:

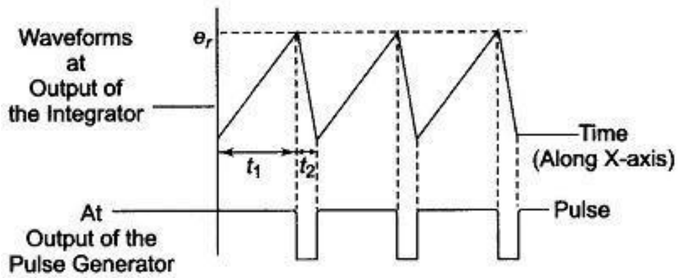


Fig 30 Dual slope method.

$$e_i = \frac{e_r t_2}{t_1}$$

But in this case e_r and t_2 are constants.

Let

$$K_2 = e_r t_2$$

$$e_i = K_2 \left(\frac{1}{t_1}\right) = K_2 (f_0)$$

The output frequency is proportional to the input voltage e_i. This DVM has the disadvantage that it requires excellent characteristics in linearity of the ramp. The ac noise and supply noise are averaged out.

6.3 POTENTIOMETRIC TYPE DIGITAL VOLTMETER

The block diagram for the potentiometric type digital voltmeter is shown in figure below:

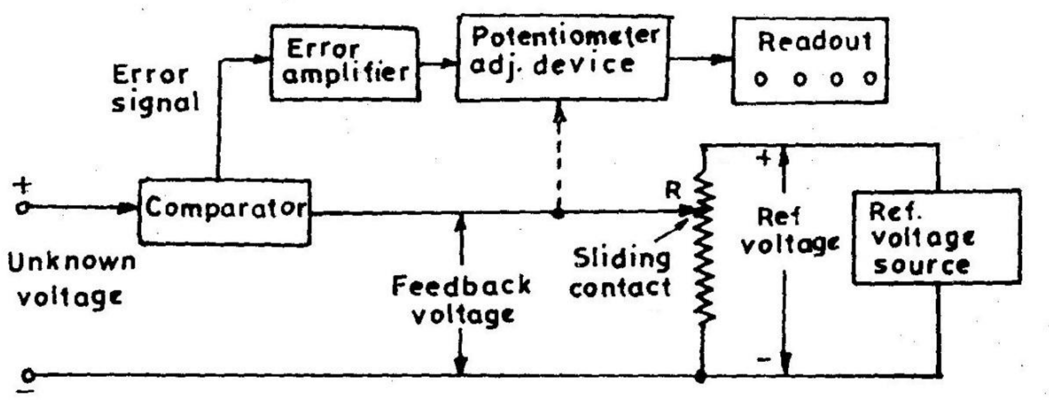


Fig.31 Potentiometric type digital voltmeter

The unknown voltage is filtered and attenuated to a suitable level. This input voltage is applied to a comparator (also known as error detector). This error detector may be chopper (a chopper circuit is used to refer to numerous types of electronic switching devices and circuits used in power control and signal applications. A chopper is a device that converts fixed DC input to a variable DC output voltage directly). The reference voltage is obtained from a fixed voltage source. This voltage is applied to a potentiometric R. The value of the feedback voltage depends upon the position of the sliding contact. The feedback voltage is also applied to the comparator.

The unknown voltage and the feedback are compared in the comparator. The output voltage of the comparator is the difference of the above two voltages. The difference of voltage is called the error signal. The error signal is amplified and is fed to a potentiometer adjustment device which moves the sliding contact of the potentiometer. This magnitude by which the sliding contact moves depends upon the magnitude of the error signal.

The direction of movement of slider depends upon whether the feedback voltage is larger or the input voltage is larger. The sliding contact moves to such a place where the feedback voltage equals the unknown voltage. In that case, there will not be any error voltage and hence there will be no input to the device adjusting the position of the sliding, contact and therefore it (sliding contact) will come to rest.

The position of the potentiometer adjustment device at this point is indicated in numerical form on the digital readout device associated with it. Since the position at which no voltage appears at potentiometer adjustment device is the one where the unknown voltage equals the feedback voltage, the reading of readout device indicates the value of unknown voltage.

The potentiometer adjustment device i.e., the device which moves the sliding contact is a 2 phase servo motor. The reference voltage source must be extremely stable and generally consists of a standard cell or Zener diode sources.

6.4 SUCCESSIVE APPROXIMATION TYPE DIGITAL VOLTMETER:

The successive approximations principle can be easily understood using a simple example: the determination of the weight of an object. By using a balance and placing the object on one side and an approximate weight on the other side, the weight of the object is determined.

If the weight placed is more than the unknown weight, the weight is removed and another weight of smaller value is placed and again the measurement is performed. Now if it is found that the weight placed is less than that of the object, another weight of smaller value is added to the weight already present, and the measurement is performed. If it is found to be greater than the unknown weight the added weight is removed and another weight of smaller value is added. In this manner by adding and removing the appropriate weight, the weight of the unknown object is determined.

The successive approximation DVM works on the same principle. Its basic block diagram is shown in figure below.

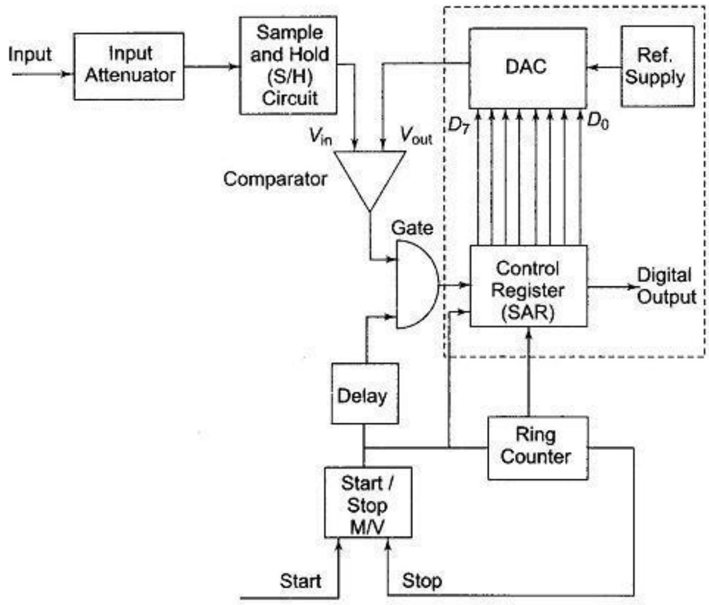


Fig.32 Successive Approximation type Digital Voltmeter.

When the start pulse signal activates the control circuit, the successive approximation register (SAR) is cleared. The output of the SAR is 00000000. V_{out} of the D/A converter is 0. Now, if $V_{in} > V_{out}$ the comparator output is positive. During the first clock pulse, the control circuit sets the D_7 to 1, and V_{out} jumps to the half reference voltage. The SAR output is 10000000. If V_{out} is greater than V_{in} the comparator output is negative and the control circuit resets D_7. However, if V_{in} is greater than V_{out} the comparator output is positive and the control circuits keep D_7 set. Similarly the rest of the bits beginning from D_7 to D_0 are set and tested. Therefore, the measurement is completed in 8 clock pulses.

At the beginning of the measurement cycle, a start pulse is applied to the start-stop multivibrator. This sets a 1 in the MSB of the control register and a 0 in all bits (assuming an 8-bit control) its reading would be 10000000. This initial setting of the register causes the output of the D/A converter to be half the reference voltage, i.e. 1/2 V. This converter output is compared to the unknown input by the comparator. If the input voltage is greater than the converter reference voltage, the comparator output produces an output that causes the control register to retain the 1 setting in its MSB and the converter continues to supply its reference output voltage of 1/2 V_{ref}.

The ring counter then advances one count, shifting a 1 in the second MSB of the control register and its reading becomes 11000000. This causes the D/A converter to increase its reference output by 1 increment to 1/4 V, i.e. 1/2 V + 1/4 V, and again it is compared with the unknown input. If in this case the total reference voltage exceeds the unknown voltage, the comparator produces an output that causes the control register to reset its second MSB to 0. The converter output then returns to its previous value of 1/2 V and awaits another input from the SAR. When the ring counter advances by 1, the third MSB is set to 1 and the converter output rises by the next increment of 1/2 V + 1/8 V. The measurement cycle thus proceeds through a series of successive approximations. Finally, when the ring counter reaches its final count, the measurement cycle stops and the digital output of the control register represents the final approximation of the unknown input voltage.

7.DIGITAL MUTIMETERS

Analog meters require no power supply, they give a better visual indication of changes and suffer less from electric noise and isolation problems. These meters are simple and inexpensive.

Digital meters, on the other hand, offer high accuracy, have a high input impedance and are smaller in size. They gives an unambiguous reading at greater viewing distances. The output available is electrical (for interfacing with external equipment), in addition to a visual readout.

The three major classes of digital meters are panel meters, bench type meters and system meters.

All digital meters employ some kind of analog to digital (A/D) converters (often dual slope integrating type) and have a visible readout display at the converter output.

Panel meters are usually placed at one location (and perhaps even a fixed range), while bench meters and system meters are often multimeters, i.e. they can read ac and dc voltage currents and resistances over several ranges.

The basic circuit shown in figure below is always a dc voltmeter.

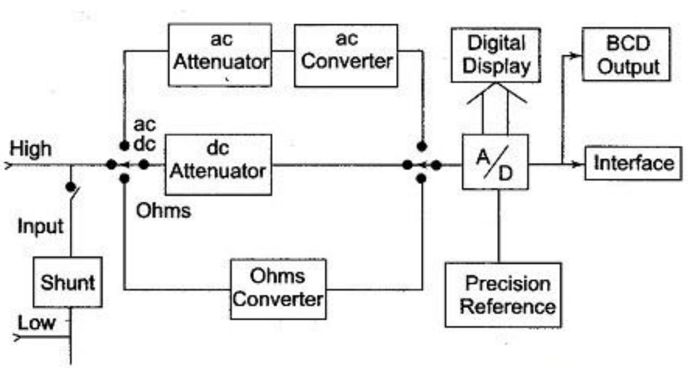


Fig.33 Digital Multimeter.

Current is converted to voltage by passing it through a precision low shunt resistance while alternating current is converted into dc by employing rectifiers and filters. For resistance measurement, the meter includes a precision low current source that is applied across the unknown resistance; again this gives a dc voltage which is digitised and readout as ohms.

A basic Digital Multimeters (DMM) is made up of several A/D converters, circuitry for counting and an attenuation circuit. A basic block diagram of a DMM is shown in figure.

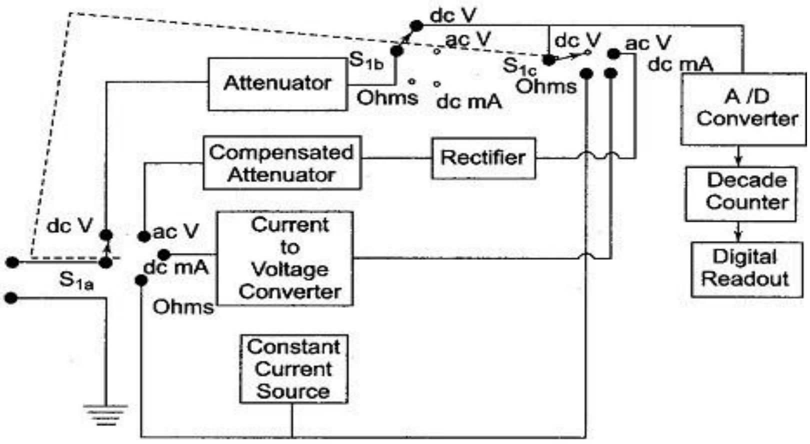


Fig.34 Block diagram of basic Digital Multimeter

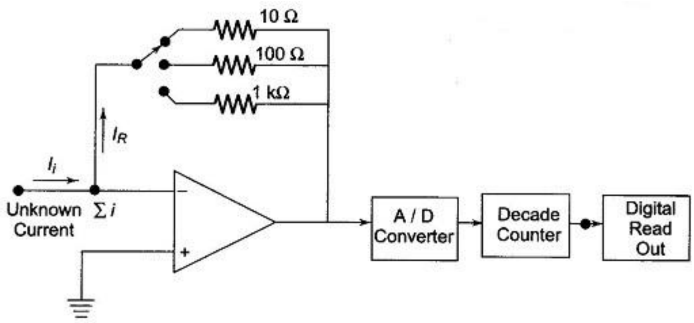


Fig.35 Current to voltage converter

The current to be measured is applied to the summing junction (Σi) at the input of the opamp. Since the current at the input of the amplifier is close to zero because of the very high input impedance of the amplifier, the current I_R is very nearly equal to I_i, the current I_R causes a voltage drop which is proportional to the current, to be developed across the resistors. This voltage drop is the input to the A/D converter, thereby providing a reading that is proportional to the unknown current.

Resistance is measured by passing a known current, from a constant current source, through an unknown resistance. The voltage drop across the resistor is applied to the A/D converter, thereby producing an indication of the value of the unknown resistance.

8. COMPUTER BASED DIGITAL INSTRUMENTS

It would be rare today to find a digital instrument that did not include a computer embedded somewhere inside. The range of possible computers is large, but they share some attributes. Small "controller chip" devices are programmable in a limited way, but they serve the function of computer quite well, despite a limited number of commands and instructions that they will execute.

A single-chip computer might also be used. These LSI devices typically have an internal central processing unit, internal random-access memory (RAM), and internal read-only memory (ROM). They will also have at least one input/output (1/0) port, and some even come with features such as event counters and analog-to-digital (A/D) converters.

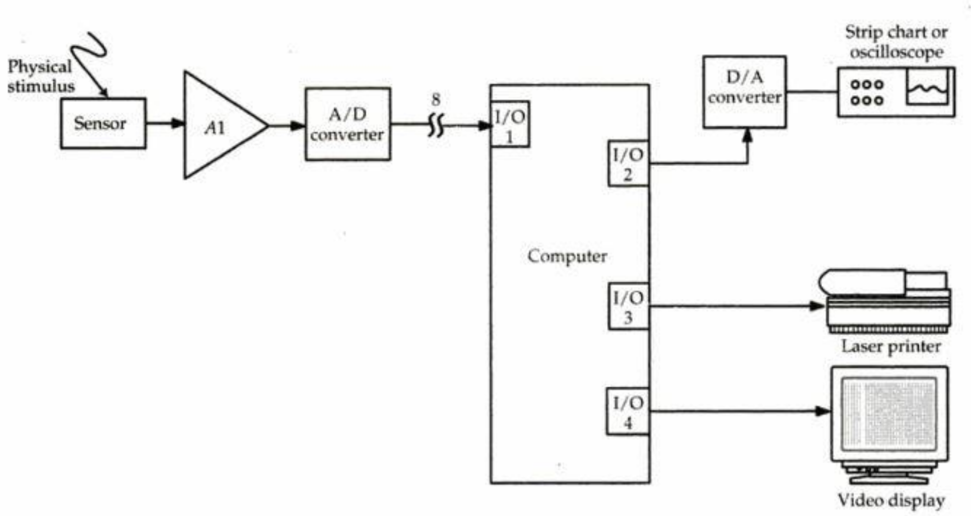


Fig.36 Block diagram of Computer based digital instruments

Still other instruments might rely on a single-board computer, that is, a plug-in printed circuit board that contains a computer chip, memory, and I/O as needed for the application.

A number of instruments, when opened and inspected, reveal a standard, off-the-shelf computer backplane (e.g.. VME Bus or Future bus+) motherboard with standard plug-in printed circuit cards (similar to a desktop personal computer) with functions such as CPU, memory (RAM and ROM), 1/O. and any analog subsystem that might be needed. There may also be mass storage devices such as tape drives, hard disk drives, or floppy disk drives within the instrument.

Figure above shows the block diagram of a simple computer-based instrument. Although simple, it conceptually can represent many, if not most, computer-based instruments on the market. Although the block diagram is simple, the complexity of the instrument is found in the embedded software. And it is the software that makes the instrument so flexible and so powerful.

Briefly described, this computer processes only a single input signal. This signal (stimulus") affects a sensor of some kind, which produces an analog output voltage. (If the sensor uses digital output, then the next two stages are deleted.) The sensor signal may be further amplified in an analog subsystem amplifier (A1), may be filtered, or may receive some other signal processing prior to input to the computer. The output of the analog subsystem is an A/D converter. It accepts an analog input signal and then converts it to a binary word that represents the same quantity. For example, an 8-bit A/D converter will recognize 256 different states. If 00000000 is used to represent 0 V, then there are 255 states remaining to represent a voltage range. If the input range is 10 V, then each one-step change of state will represent 0.039 V. for a maximum input value of 9.96 V (represented by binary word 11111111).

The processing is done in the internal software. The output devices, also connected to various types of I/O ports, depend on the application. There might be a simple alphanumeric display, or there might be a computer video terminal. A laser printer can display not only alphanumeric data but also waveform and graphical data as well. The computer may have to provide analog waveform data to analog displays such as an oscilloscope or paper chart recorder. To accommodate these instruments, a digital to-analog converter (DAC) is needed.

Computer-based instruments are not the be-all and end-all, for there are still plenty of applications for analog instruments and basic digital instruments. But design is now a lot easier, and implementation a lot cheaper, so the power and flexibility of computer based instrumentation are finding a home in progressively lower-priced and simpler products. It is perhaps the sheer flexibility of computer-based approaches to instrument design that is of primary importance. A relatively low cost hardware infrastructure will support a huge array of different instruments. If an open system standards is taken, then flexibility and universality are even more within reach.

8.1 IEEE 488 GPIB INSTRUMENTS

The IEEE-488 bus is also called a general purpose interface bus (GPIB). The Hewlett-Packard interface bus (HPIB) is a proprietary version of the IEEE-488 bus. The main purpose for the IEEE-488/GPIB is automatic test equipment, both generalized and specific.

Test instrumentation that is intended for GPIB service will have a 24-pin "blueline" connector on the rear panel. This connector is one of the Amphenol blueline series not unlike the 36-pin connector used for parallel printer interface on microcom will also be a GPIB ADDRESS DIP switch on the rear panel, usually near the connector. The purpose of the switch is to set the 5-bit binary address where the instrument is located in the system; it determines whether or not the device is a listener only or a talker only and certain other details.

Basics & Structure:

The IEEE-488/GPIB specification provides technical details of the standard bus. The logic levels on the bus are generally similar to TTL: LOW is less than or equal to 0.800 V, while a HIGH is greater than 2.0 V. The logic signals can be connected to the instruments through a multiconductor cable up to 20m (66 ft) in length, provided that an instrument load is placed every 2m. This specification works out to a cable length in meters twice the number of instruments in the system. Most IEEE-488/GPIB systems operate unrestricted to 250 kilobytes per second or faster with certain specified restrictions.

There are two basic configurations for the IEEE/GPIB system (Figure below): linear and star. These configurations are created with the cable connections between the instruments and the computer. The linear configuration is basically a daisy chain method in which the tap-off to the next instrument is taken from the previous one in the series. In the star configuration the instruments are connected from a central point.

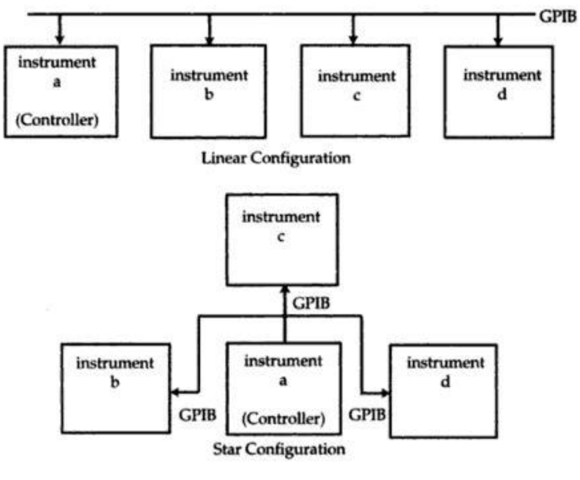


Fig.37 GPIB system Configurations

There are three major buses in the IEEE-488/GPIB system. Each line in each bus has a circuit similar to that shown in figure below.

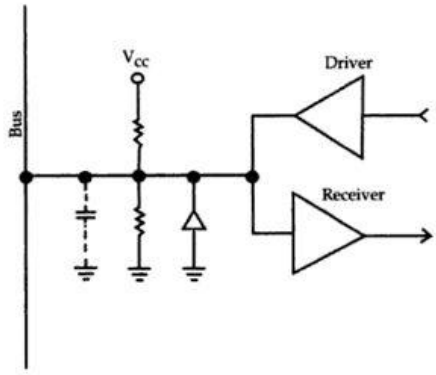


Fig.38 GPIB input-Output circuits.

Besides the shunt protection diode and stray capacitance, there are also pull-up and pull-down resistors that effectively determine the standardized input impedance. Connected to the bus line are receiver and driver circuits. These similar-to-TTL logic elements provide input or output to the instrument. The driver is an output and will be a tristate device; that is, it is inert until commanded to turn on. A tristate output will float at high impedance until turned on. The receiver is basically a noninverting buffer with a high-impedance input. This arrangement of drivers and receivers provides low loading to the bus.

Figure below shows the basic structure of the IEEE-488/GPIB. There are three buses and four different types of devices. The devices are controllers, talker only, listener only, and talker/listener. These devices are defined as follows:

CONTROLLERS. This type of device acts as the brain of the system and communicates device addresses and other interface messages to instruments in the system. Most controllers are programmable digital computers. Both Hewlett-Packard and Tektronix offer computers that serve this function, and certain other companies produce hardware and software that permit other computers to act as IEEE-488/GPIB controllers.

LISTENER: A device capable of listening will receive commands from another instrument, usually the controller, when the correct address is placed on the bus. The listener acts on the message received but does not send back any data to the controller. As shown in basic structure, the signal generator is an example of a listener.

TALKER: The talker responds to the message sent to it by the controller and then sends data back to the controller over the DIO data bus. A frequency counter is an example of a talker.

There is also a combination device that accepts commands from the controller to set up ranges, tasks, and so forth, and then returns data back over the DIO bus to the controller. An example is a digital multimeter (DMM). The controller will send the DMM commands that determine whether the data are ac or dc; whether the data are volts, milliamperes, or ohms, and what specific range the data have—the device is thus acting as a listener. When the measurement is made, the DMM becomes a talker and transmits the data measured back over the DIO bus to the controller.

The IEEE-488/GPIB system has three major buses: general interface management (GIM), data byte transfer (DBT), and data input/output (DIO). These buses operate as follows:

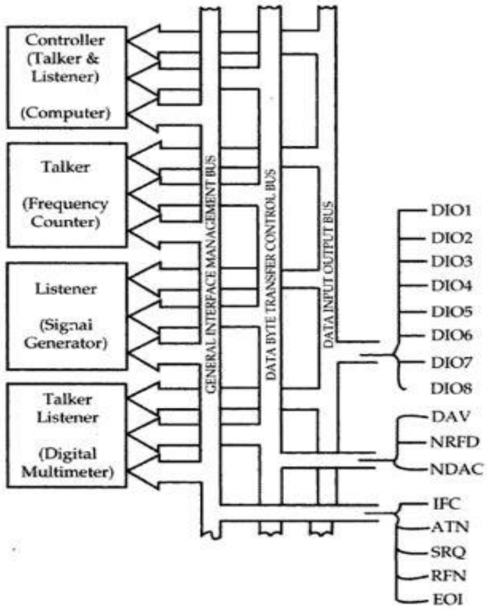


Fig.39 Basic structure of GPIB bus

DIO BUS: The data input/output bus is a bidirectional 8-bit data bus that carries data, interface messages, and device-dependent messages between the controller, talkers, and listeners. This bus sends data asynchronously in byte-serial format. DBT bus. The data byte transfer bus controls the sending of data along the DIO bus. There are three lines in the DBT bus: data valid (DAV), not ready for data (NRFD), and not data accepted (NDAC). These signal lines are defined as follows:

DAV: The data valid signal indicates the availability and validity of the data on the line. If the measurement is not finished, for example, the DAV signal will be false.

NRFD: The not ready for data signal lets the controller know whether or not the specific device addressed is in a condition to receive data.

NDAC: The not data accepted signal line is used to indicate to the controller whether or not the device accepted the data sent to it over DIO bus.

GIM BUS: The general interface management bus coordinates the system and ensures an orderly flow of data over the DIO bus. It has the following signals: interface clear (IFC), attention (ATN), service request (SRQ), remote enable (REN), and end or identify (EOI). These signals are defined as follows:

IFC: The interface clear signal is used by the controller to place all devices in a predefined quiescent or standby condition.

ATN**:** The attention signal is used by the controller/computer to let the system know how data on the DIO bus lines are to be interpreted and which device is to respond to the data.

SRQ: The service request signal is used by a device on the system to ask the controller for attention. This signal is essentially an interrupt request.

REN: The remote enable signal is used by the controller to select between two alternate sources of device programming data.

EOI**:** The end or identify signal is used by talkers for two purposes. It will follow the end of a multiple byte sequence of data in order to indicate that the data are now finished. It is also used in conjunction with the ATN signal for polling the system.

SCHOOL OF ELECTRICAL AND ELECTRONICS

DEPARTMENT OF ELECTRONICS AND INSTRUMENTATION

UNIT – IV

ELECTRONIC MEASUREMENT AND INSTRUMENTATION – SIC 1305

UNIT-4

1. DATA ACQUISITION AND OSCILLOSCOPES

1.1 DATA ACQUISITION SYSTEM

In order to optimize the characteristics of the system in terms of performance of the system, data handling capacity and cost, different relevant sub-systems are combined together. The system used for data processing, data conversion, data transmission, data storage is called data acquisition system. The typical data acquisition system consists of sensors with necessary signal conditioning, data conversion, data processing, data handling and transmission, storage and display systems. The modern electronic instrumentation is not becoming more sophisticated because of tremendous development in micro-electronic devices, op-amps, multiplexers, data converters, microprocessors and microcontrollers. The data measurement systems and process controls are very flexible and more programmable now a days, because of development in automation field. The data acquisition system relates to collection of the input data in the digital form rapidly, economically and accurately as necessary.

Objectives of Data Acquisition system:

i) The data acquisition system must acquire the necessary data at correct speed and at the correct time.

ii) It must use all the data efficiently to inform the operator about the state of the plant.

iii) It must monitor the operation of complete plant so that optimum online safe operations are maintained.

iv) It must provide effective human communication system which helps in identifying the problem areas. This minimizes unit availability and maximizes the unit output at lower cost.

v) It must be able to collect, summarise and store data properly for diagnosis and record purpose of any operation.

vi) It must be able to compute unit performance indices using online real time communication.

vii) It must be flexible. Also the expansion facility for the future requirement must be provided by it.

viii) It must be reliable and should not have a down time greater than 0.1%.

The data acquisition systems are basically used to measure and record the signals obtained in two ways. Firstly the signal may be originating from direct measurement of an electrical quantity such as ac or dc voltage, frequency, component value such as resistance, capacitance etc. Such signals are always found in electronic component testing, environmental studies etc. Secondly the signal may originate from the transducers such as pressure transducers, thermocouples.

The data processing involves a variety of operations ranging from simple comparison to complicated arithmetical manipulations. This can be used to collect various data or information, perform some operations if required, convert this information to the suitable form using converters, perform more number of calculations to remove unwanted noise signal, gather results to be displayed and so on. The transmission of data can take place over very long distances or very short distances. The results which are gathered may be displayed directly on the digital panel or may be on CRT. The data stored may be permanent or temporary.

To collect data rapidly, shift digitiser or some high resolution devices may be used. For converting analog signal to digital, additional transducers, amplifiers and multiplexers are used.

The use of sample and hold circuit increases the speed with which accurate conversion of information is possible.The data acquisition system is mainly classified as analog data acquisition system and digital data acquisition system. The analog data acquisition systems mainly deal with the measurement information which is in the analog form. An analog signal is the continuous signal such as voltage versus time or displacement due to the pressure. While the digital data acquisition system may consists of number of discrete and discontinuous pulse representing high and low pulses which is in digital form. The relationship of these pulses with time gives the nature or magnitude of the quantity.

1.2 Analog Data Acquisition System:

The basic components used in the analog data acquisition system are as follows,

1. Transducer: The transducer is used to convert the physical quantity into an electrical signal. The transducers such as strain gauge, thermocouples, piezoelectric devices, photosensitive are most widely used. The transducer generates a voltage proportional to the physical quantity being measured. This voltage is applied as a input to the data acquisition system. Apart from this some special sensors produce frequency which can be counted by an electronic counter. This frequency forms the integral part of the quantity being measured. Otherwise the signal may be modulated then voltage level is reduced with the help of discriminator.

2. Signal Conditioner: This device includes the supporting circuitry for the transducers. It allows the output voltage of transducer to amplify upto desired level. It also converts the output voltage to the desired form so that it is accepted by the next stage. It produces the conditions in transducers so that they work properly. It also provides excitation power and balancing circuits.

3. Multiplexers: It allows a single channel to share it with more than one input quantity. It accepts multiple analog inputs. With the help of multiplexer we can transmit more than one quantity using same channel. The multiplexers are mostly used when many quantities are to be transmitted. Also when the distance between the transmitting end and receiving end is more, the multiplexers are used. Multiplexers reduce the cost of installation, maintenance and periodic replacement of channels if those are used for separate input signals.

4. Calibrating Equipment: Before each test, the calibration is carried out. This is called pre-calibration. Similarly after each test calibration is carried out and it is called post-calibration. It usually consists millivolt calibration of all input circuits and shunt calibration of all bridge type transducers.

5. Integrating Equipment: This block is used for integration or the summation of a quantity. The digital techniques are normally used for integration purposes.

6. Visual Display Devices: These are necessary to monitor the input signal continuously. These devices include panel meters, numerical displays, single or multichannel CROs, storage CRO etc.

7. Analog Recorders: These are required to record the output signal. The analog recorders include strip chart recorder, magnetic tape recorder etc.

8. Analog Computers: These are used as data reduction device. The output voltage of the analog computer may be converted to digital form for further computations. Even though the accuracy of analog computations is comparatively less than the digital one the analog computers are used because of its less cost.

1.3 Digital Data Acquisition system:

The simple digital data acquisition system is shown below:

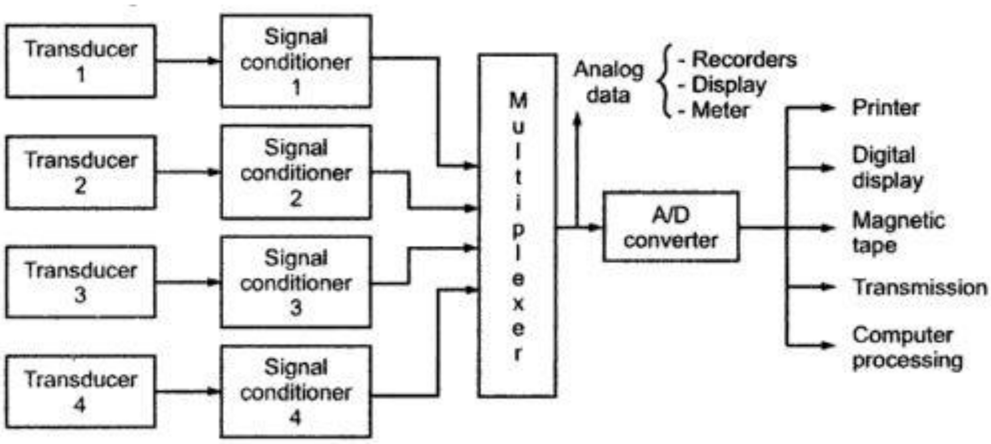


Fig.1 Generalized data acquisition system

The digital data acquisition system includes all the blocks shown in the figure. It may use some additional function blocks. The essential functions of a digital data acquisition system are as follows,

i) It handles the analog signals,

ii) It performs measurement,

iii) It converts analog signal into digital data and handles it,

iv) It performs internal programming and control. The various components of the digital data acquisition system are as follows

1. Transducers: They convert the physical quantity into a proportional electrical signal which is given as a input to the digital data acquisition system.

2. Signal Conditioners: They include supporting circuits for amplifying, modifying or selecting certain positions of these signals.

3. Multiplexers: The multiplexer accepts multiple analog inputs and connects them sequentially to one measuring instrument.

4. Signal Converters: The signal converters are used to translate analog signal to a form which is suitable for the next stage that is analog to digital converter. This block is optional one.

5. Analog to Digital Converters (A/D converter) : The analog to digital converter converts the analog voltage to its equivalent digital form. The output of the analog to digital converter may be fed to the digital display devices for display or to the digital recorders for recording. The same signal may be fed to the digital computer for data reduction or further processing.

6. Auxiliary Equipments: The devices which are used for system programming functions and digital data processing are included in the auxiliary equipments. The typical functions of the auxillary equipment includes linearization and limit comparison of the signals. These functions are performed by the individual instruments or the digital computer.

7. Digital Recorders: They record the information in digital form. The digital information is stored on punched cards, magnetic tape recorders, type written pages, floppies or combination of these systems. The digital printer used provides a high quality, hard copy for records minimizing the operator's work.

The data acquisition systems are used, now a days in increasing wide fields. These are becoming very much popular because of simplicity, accuracy and the most important reliability of the systems. These are widely used in industrial areas, scientific areas, including aerospace, biomedical and telemetry industries.

When the lower accuracy is tolerable or when wide frequency bandwidth is needed, the analog data acquisition systems are used. The digital data acquisition systems are used when the physical quantity being measured has very narrow bandwidth. When the high accuracy with low per channel cost is required, the ultimate solution is to use the digital data acquisition system.

1.4 Single channel Data Acquisition system:

The basic single channel data acquisition system consists of a signal conditioner circuit, analog to digital converter, buffer. The analog to digital converter performs the conversions repetitively at a free running internally determined rate. The basic single channel data acquisition system is as shown in the figure below

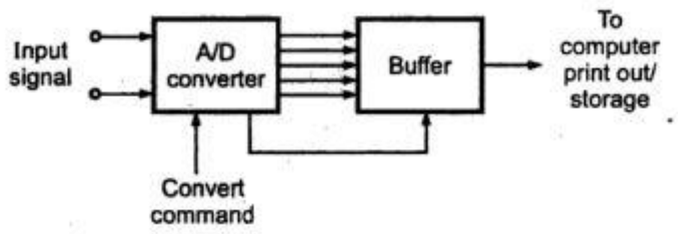


Fig.2 Single channel DAS

The digital outputs from the buffer are further fed to either digital computer or storage or print out device. The most popular example of the single channel data acquisition system is the digital panel meter (DPM).

The digital outputs are obtained from the analog to digital converter. The analog to digital converters used for the data acquisition system are designed such that they can accept external commands to convert and hold operations. A/D converters based on dual slope techniques are mostly used for the conversion of low frequency data, generally from thermocouples. The successive approximation technique is most widely used because it gives high resolution and high speed at moderate cost.

Many times it is observed that the signal level is very low compared to the input requirement. In such cases, the amplification of the input signal is done to bring its level to match the input requirement. This is called pre-amplification. If the input signals are to be isolated from the system physically, the conductive paths are broken by using mostly opto-coupled isolation amplifier. The pre-amplifiers may be coupled with active filters before the processing of data. These filters minimise the effect of noise carrier and interfering high frequency components. Sometimes special purpose filter such as tracking filter may be used to preserve the phase dependent data.

1.5 Multi-channel Data Acquisition system:

The different subsystems of DAS can be time shared by two or more input sources. The number of techniques are employed for time shared measurements. Basically there are three types of multiplexed systems. They are as follows,

1) Multichannel analog multiplexed system:

The basic block diagram is shown below

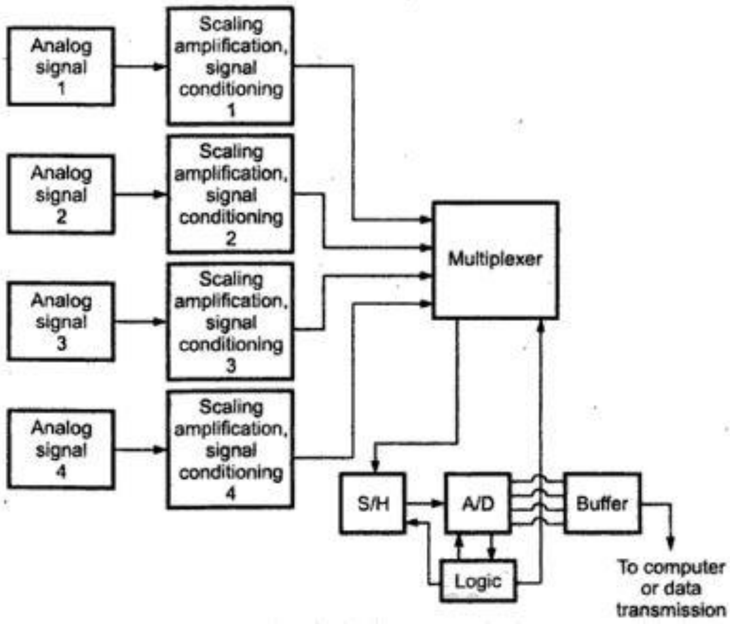


Fig.3 Multichannel DAS (A/D preceded by a multiplexer)

In this system, there is a single analog to digital converted preceded by a multiplexer. The individual analog signals are applied directly. They are amplified and signal conditioned and connected to the multiplexer. Then analog to digital converter converts these analog signals to digital signals.

For the utilization of time, the multiplexer gets or selects new channel to be converted while the previous data stored in the sample and hold circuit is converted into digital form. When this conversion completes, the status line from the converter makes the sample and hold circuit to go back to sample mode. Then it again collects the signal of the next channel. When the data is collected, on proper command, the sample and hold circuit mode changes to the hold mode. The conversion starts again while the multiplexer selects next channel.

This process is comparatively slower but has obvious advantage of low cost due to sharing of more than one channel.

2) Multiplexing outputs of sample and hold circuits:

When more number of channels are to be monitored at the same time, the multiplexing of the outputs of sample and hold circuit is done. In this case, each sample and hold circuit is attached to each channel. They are synchronously updated by timing circuit. The sample and hold circuit are first multiplexed and then they are connected to the analog to digital converter. This gives the sequential read out of the signals.

The advantage of this multiplexing technique is that this is moderately faster than earlier.

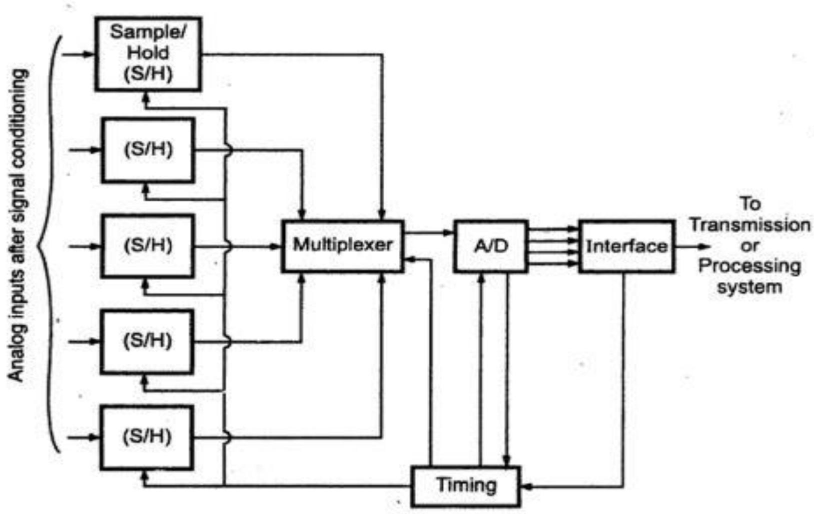


Fig.3 Simultaneous sampled system multiplexer.

3) Multiplexing after A/D conversion:

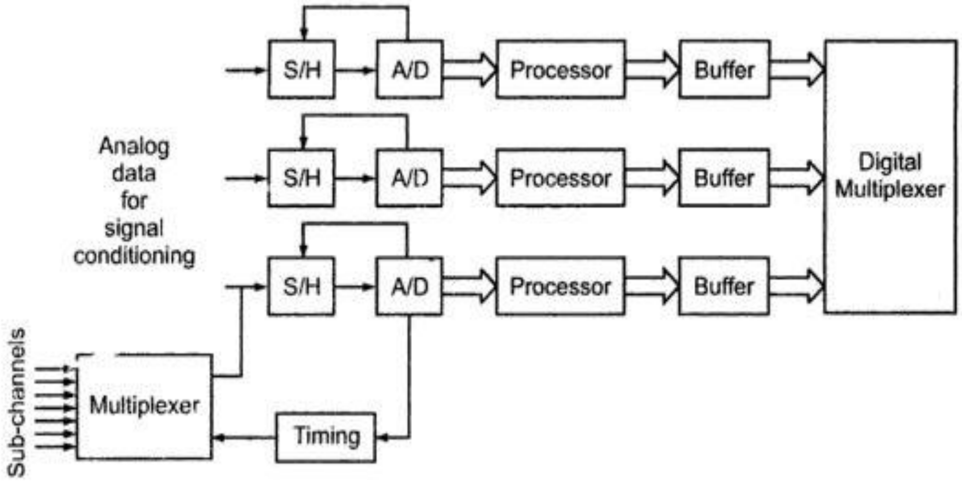


Fig.4 Muti-channel DAS using digital multipexing

In this technique, each sample and hold circuit and the analog to digital converter is assigned to the individual channel. The conversion speed is as per the requirement. This technique is exhibiting parallel conversion. In industrial systems, the number of strain gauges, thermocouples, LVDTs are distributed over large area. As the analog signal is digitised at the centre, the data transmission provides great immunity against line frequency and other interferences. The data in digital form performs logical operations. Based on relative speed at which changes occur in the data, the scanning rate can be increased or decreased. These are also called fast data acquisition systems.

4) PC based Data acquisition system:

Nowadays easily available, low cost and wide use of Personal Computer (PC) led the various manufacturers to develop interfaces between PC and the outputs of transducers. The outputs are further processed and analyzed by the PC. The advanced techniques allows user to use a PC ADD-ON card which is available in various configurations. By putting this card in PC, the analog signals can be directly interfaced to PC. This gives very powerful communication and analysis of the multiple measurement data.

The main features of PC based DAS are as follows,

i) The PC based system is used to display the parameters of the system continuously. This helps the operator in monitoring all the parameters instantaneously and conveniently.

ii) The system parameters are displayed with some display attributes such as blinking, underline, inverse video, extra bright so that the attention of the operator is called. Sometimes colour graphic display is used to indicate normal operation, close to the upper limit and out of the limit.

iii) Sometimes some man-machine interfaces called MIMIC displays are used. These are useful in displaying the data measured at any location on the plant near the icon of that location on the screen. This helps operator to take quick corrective action.

iv) Several parameters are plotted individually or simultaneously on the screen to show their characteristics. This helps in pointing out the variations in these parameter measured at two different time. Hence the PC based data acquisition systems gives meaningful and reliable results even though the large number of inputs are measured.

The basic block diagram of PC based data acquisition system is as shown in the figure below.

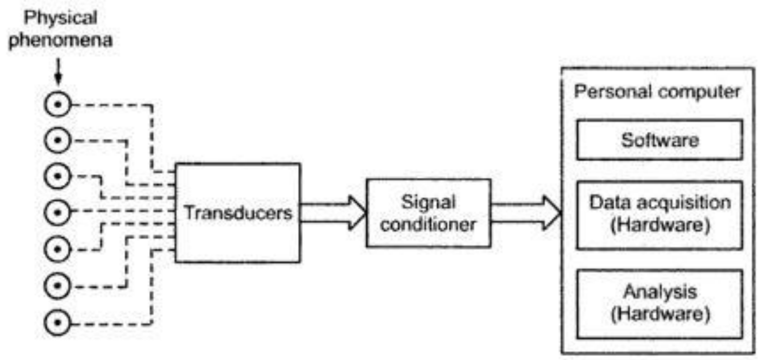


Fig.5 PC based data Acquisition system

In the PC based data acquisition system, the personal computer consists of additional hardware such as data acquisition hardware and data analysis hardware. These are associated by the software. The software contains the complete programs written for accessing the data, performing calculations with them and display the result on video display unit (VDU). The data analysis hardware is used to speed up the computations and analysis mainly in case of digital signal processing (DSP) applications. The data acquisition hardware generally acts as a multipurpose, multifunction card. The input signal between the range 10 mV to 10 V is amplified by digitally programmable gain amplifier to a standard level. This amplified input signal is converted to digital signal with the help of high speed analog to digital converters interfaced to the PC bus. This signal is recorded, displayed or may be processed further for analysis by the personal computer.

2. DATA LOGGERS

A data logger (also datalogger or data recorder) is an electronic device that records data over time or in relation to location either with a built in instrument or sensor or via external instruments and sensors. Increasingly, but not entirely, they are based on a digital processor (or computer). They generally are small, battery powered, portable, and equipped with a microprocessor, internal memory for data storage, and sensors. Some data loggers interface with a personal computer, and use software to activate the data logger and view and analyze the collected data, while others have a local interface device (keypad, LCD) and can be used as a stand-alone device.

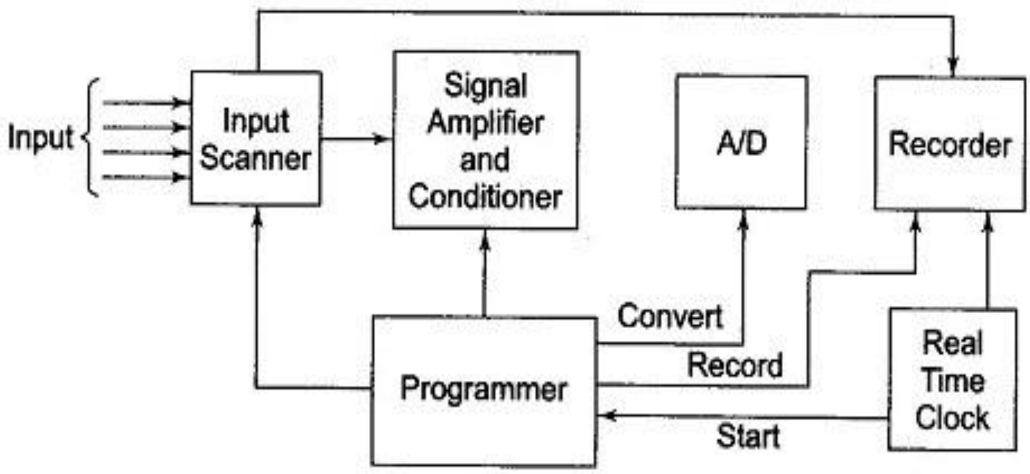


Fig.6 Block diagram of Data logger.

The basic blocks of the data logger system are as follows.

1. Input scanner
2. Signal conditioner
3. Analog to digital converter
4. Recording system
5. Programmer.

The data logger can measure any electrical output from almost all the transducers. The input signals those are fed to the input scanner are high level pressure transducer signal, low level thermocouple signal, pneumatic signals from pneumatic transducers, on-off signals from relays or switches, a.c. signals, digital quantities.

The low d.c. level signals are first amplified, then conditioned and then fed to analog to digital converter. The high level signals are directly fed to the analog to digital converter. The pneumatic and a.c. signals are first converted to d.c. voltage level, conditioned and then converted to digital form. In this way all types of input signals are converted to suitable form that can be handled by the data logger. The purpose of conditioner is to provide similarity between the signals from various transducers which might not have linear characteristics. The filter circuits are also used to remove noise signal and ripple voltage suppression.

The input scanner is an automatic switch which can select each signal in turn. The scanner is selecting only one input signal at a time, the data logger needs only one signal amplifier and conditioner also one analog to digital converter and signal digital recorder. The signal amplifier amplifies the low level signals and maintain all the input signals to the 5 V level. The signal if varies non linearly with the parameter measured, the linearization of signal is done by the signal conditioner. The analog signals are then converted to the digital form which are suitable to drive the digital recorders. The programmer controls the sequence operation of the various items of the logger. It commands the scanner when to select new channel. It receives information from the scanner, converter and recorder. The real time clock is used to automate the system. The clock commands the programmer to sequence the measurements at the intervals selected by the user.

3. PROBLEMS IN SIGNAL ACQUISITION

Recording strong signals (i.e., those over about 50 mV) is usually a straightforward job, and few problems will be encountered. But as the full-scale value of the signal amplitude drops, then certain acute problems begin to emerge. Signals in the millivolt and high microvolt range will exhibit some of these problems, while signals in the low microvolt and nanovolt range usually exhibit the problems.

Noise signals mixed with the desired signal will be recorded and displayed as a valid signal unless steps are taken to eliminate the noise or at least reduce its value to a point of negligible effect. Noise is any electrical signal or tracing anomaly that is not part of the desired signal. Several different types of noise arc recognized: white noise, impulse noise, and interference noise.

White noise supposedly contains all frequencies, phases, and amplitudes, so it gets its name from the analogy to white light, which contains all visible colors. Such noise is also called gaussian noise, although it is neither truly "white" nor "gaussian" unless there are no bandwidth limits present. True white noise has a bandwidth of de to daylight and beyond. In most instrumentation systems, however, there are bandwidth limitations to consider, so the noise is actually pseudogaussian (also called "pink noise." i.e., bandwidth-limited white noise). The bandwidth limitations are often put in place to limit the effects of noise on the system. Because it integrates to zero given sufficient time, true gaussian noise can be eliminated by low-pass filtering or bandpass filtering. Bandwidth-limited noise, however, does not usually integrate to zero but rather to a very low value.

An example of pseudogaussian bandwidth-limited noise is the "hiss" heard between stations on an FM broadcast band receiver (such noise is present between stations on the AM band as well but is obscured by other forms of noise that are also present). Most such noise in instrumentation systems is due to thermal sources, and it has an rms value of

$$E = \sqrt{4KT\,BR}$$

where E = the rms value of the noise potential in volts (V)

K = Boltzmann's constant (1.38 x 10^{-23} J/K)

T = the temperature in kelvin (K) (use 300 K as standard room temperature)

B = the bandwidth in hertz (Hz)

R = the resistance in ohms (Ω) Note in above equation that a simple resistor, with no power applied, will generate a noise signal due to thermal agitation of the atoms within the resistor.

In above example a signal of 4.1 kV is created by nothing more than thermal agitation of molecules in a circuit resistance. Although this signal may appear to have a low amplitude, keep in mind that many signals recorded on oscilloscopes and graphics recorders are also in the microvolt range. For example, the human electroencephalograph (EEG) is a recording of brain wave potentials and must be able to deal with signals with peak amplitudes in the (1-80)µV range. In that type of application a 4.1µV noise signal represents a sizable artifact.

The answer to this problem is to keep the circuit impedances in the early stages that is those stages that most of the gain stages follow-very low so that the 'R' term in equation above is low. Additionally, the bandwidth of recording amplifiers should be adjusted to that required to faithfully reproduce the input signal waveform. This tactic will reduce the bandwidth term (B) in above equation.

Several other types of noise are peculiar to solid-state amplifiers: shot noise, Johnson noise, and flicker noise. In low-cost amplifiers these noise sources can add up to significant amplitude. Although low-pass filtering offers relief, it is better to specify a low-noise preamplifier when one is dealing with low-level signals. Impulse noise is due to local electrical disturbances such as arcs, lightning bolts, electrical motors, and so forth. Shielding of signal lines will help somewhat, as will low-pass filtering. but the best solution is to eliminate the noise at its source. Filtering, incidentally, is a two-edged sword, and it must be done prudently. Filtering tends to broaden pulse signals, and that can create even more problems than it cures.

Other electrical devices nearby can induce signals into the instrumentation system, the chief among these sources being the 60-Hz ac power lines. It is wise to use only differential amplifier inputs, because of their high common mode rejection ratio (CMRR). Signals from the desired source can be connected across the two inputs, and so become a differential signal, while the 60-Hz interference field affects both inputs equally and thus is common (and is thereby suppressed by the amplifier CMRR).

4. INTERFERENCE AND SCREENING, GROUNDING AND GUARDING TECHNIQUES

It is sometimes possible to erroneously manufacture a valid differential or single-ended signal from a common mode interference signal. There are two principal ways this is done, and both involve the improper use of shielded wire inputs. One source of this problem, called a ground loop, is shown in figure (a) below

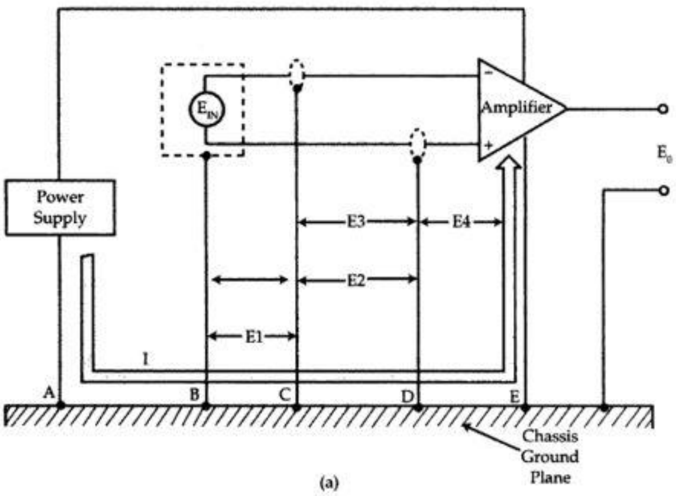


Fig.7 (a) Improper Grounding

This problem arises from the use of too many grounds. In this example the shielded source, the shielded input lines, and the de power supply are all grounded to different points on the ground plane. Power supply direct currents I flow from the dc power supply at point A to the amplifier common point E. Since the ground has resistance (as all conductors do), the voltage drops E_1 through E_4 are formed when the direct current flows. These voltages are seen by the amplifier as valid signals and can become especially troublesome if 'I' is a varying current.

The solution to this problem is to use single-point grounding, also called star grounding, as shown in figure (b) below.

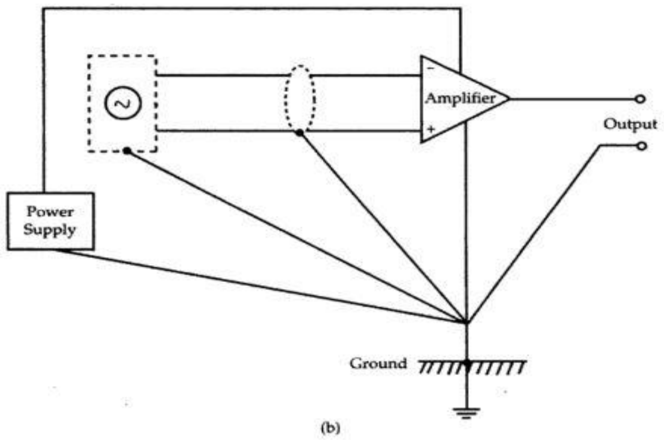
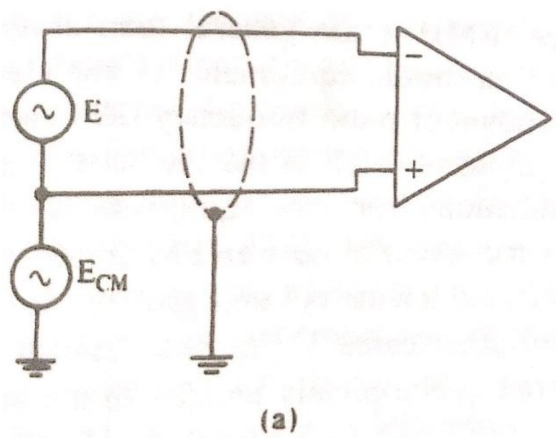


Fig.8 (b) Proper single point grounding

Some amplifiers used in sensitive signal acquisition systems keep the power and signal grounds separate except at a single, specified common point. In fact, a few models go further by creating several ground buses, especially where dc power plus analog, digital, video, and RF signals must be mixed together on a printed wiring board. In some instances the shield on the input lines must not be grounded at both ends. In those cases it is usually better to ground the shield only at the amplifier end of the cable.

Figure below shows some of the causes of and cures or differential artifact signals manufactured from common mode signal sources. The circuit in fig (a) uses standard single shielding, and the equivalent circuit of the same is shown in fig (b) which reveals a potential problem. The shield has a capacitance to ground with input wires C_1 & C_2 and a cable resistance are always present represented in fig (b) as R_1 & R_2.



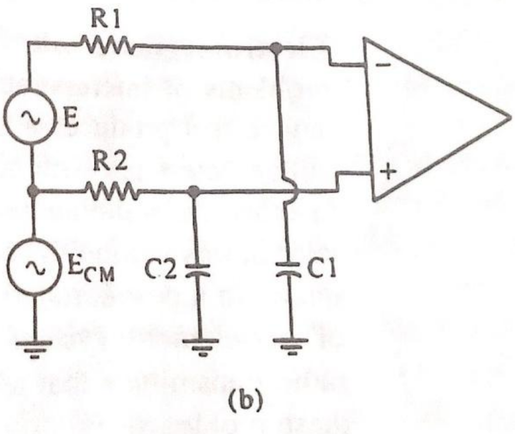


Fig.9 (a) Differential and common mode signals **Fig.10 (b) Equivalent circuit**

The system works well if $R_1=R_2$ and $C_1=C_2$, but even small imbalances in the RC networks can allow E_{CM} to develop a differential signal.

A guard shield circuit shown below can be used to overcome this problem. The guard shield is driven by signals from the two input lines through high value resistors R_A & R_B and in many cases through a common mode amplifier. This tactic has the effect of placing both sides of the cable capacitances at the same potential so that $E_{C1}=E_{C2}=0$. The outer shield is not strictly needed but it can serve to reduce coupling from the outside common mode radiation source to the input signal lines , thereby reducing the value of E_{CM} .

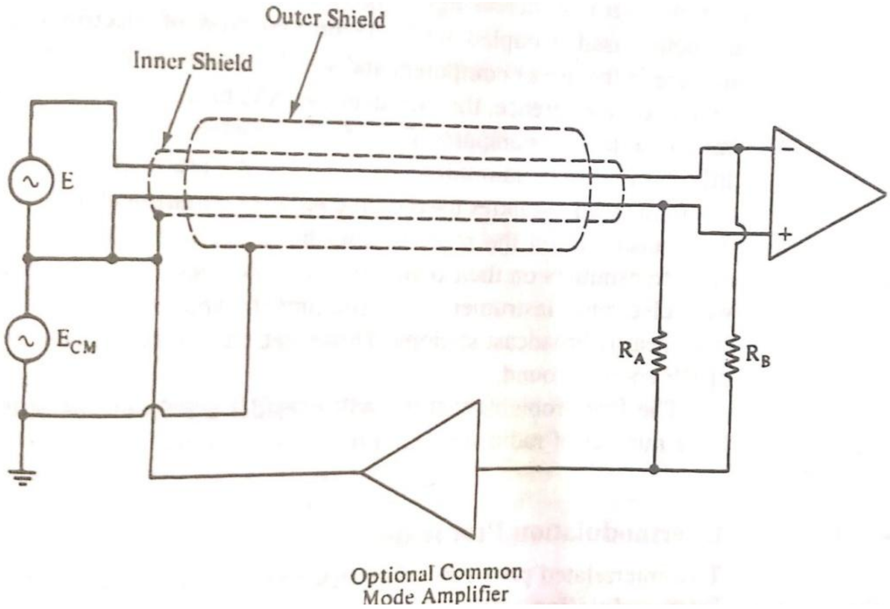


Fig.11 Guard shield connections

5. INTRODUCTION- CATHODE RAY TUBE (CRT)

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 a function of time, is called cathode ray oscilloscope, commonly known as C.R.O. The C.R.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.

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 C.R.O. 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 the input voltage signal applied to the y-axis of C.R.O. 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 C.R.O. 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 C.R.O.

The Cathode Ray Tube:

The cathode ray tube (CRT) is the heart of the C.R.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 figure below

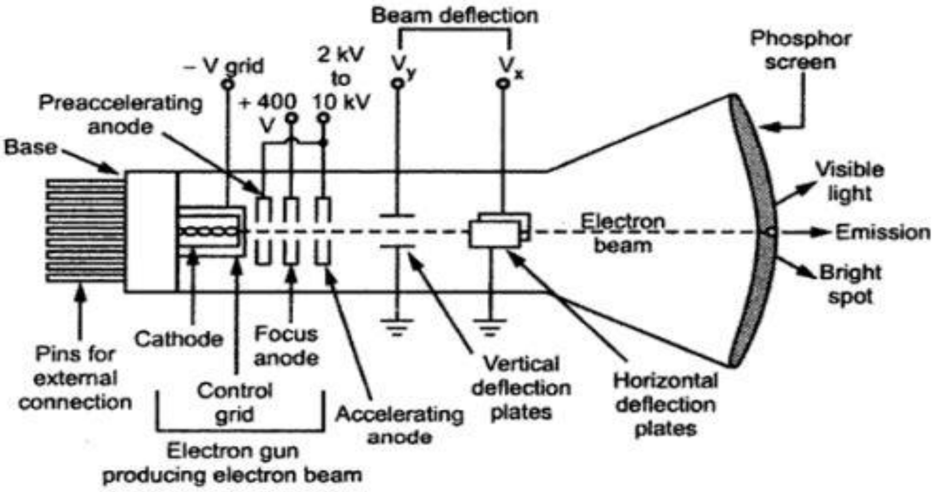


Fig.12 Block diagram of cathode ray tube (CRT)

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 thermally heated cathode, emitting the electrons. The control grid is given negative (-) potential with respect to cathode. 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 bombardment. 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. Since the electron beam consists of many electrons, the beam tends to diverge. This is because the similar (negative) charges on the electron repel each other. To compensate for such repulsion forces, an adjustable electrostatic field is created between two cylindrical anodes, called the focusing anodes. The high positive potential is also given to the pre-accelerating anodes and accelerating anodes, which results into the required acceleration of the electrons.

Both focusing and accelerating anodes are cylindrical in shape having small openings located in the centre of each electrode, co-axial with the tube axis. The pre-accelerating and accelerating anodes are connected to a common positive high voltage which varies between 2 kV to 10 kV. The focusing anode is connected to a lower positive voltage of about 400 V to 500 V.

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 figure above, the electron beam passes through these plates. A positive voltage applied to the Y input terminal (V_y) 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.

Similarly, a positive voltage applied to X-input terminal (V_x) will cause the electron beam to deflect horizontally towards the right; while a negative voltage applied to the X-input terminal will cause the electron beam to deflect. horizontally towards the left of the screen. The amount of vertical or horizontal deflection is directly proportional to the correspondingly applied voltage.

When the voltages are applied simultaneously to vertical and horizontal deflecting plates, the electron beam is deflected due to the resultant of these two voltages. The face of the screen can be considered as an x-y plane. The (x,y) position of the beam spot is thus directly influenced by the horizontal and the vertical voltages applied to the deflection plates V_x, and V_y, respectively.

The horizontal deflection (x) produced will be proportional to the horizontal deflecting voltage, V_x, applied to X-input.

$$x \propto V$$

$$x = K_X V_X$$

where K_x is constant of proportionality. The deflection produced is usually measured in cm or as number of divisions, on the scale, in the horizontal direction.

Then $K_x = x/V_x$ where K_x is expressed as cm/volt or division/volt, is called **horizontal sensitivity** of the oscilloscope.

Similarly, the vertical deflection (y) produced will be proportional to the vertical deflecting voltage, V_y, applied to the y-input.

$$y \propto V$$

$$y = K_y V_y$$

Then $K_y = y/V_y$ and K_y, the **vertical sensitivity**, will be expressed as cm/volt, or division/volt.

The values of vertical and horizontal sensitivities are selectable and adjustable through multipositional switches on the front panel that controls the gain of the corresponding internal amplifier stage. The bright spot of the electron beam can thus trace (or plot) the x-y relationship between the two voltages, V_x and V_y.

The schematic arrangement of the vertical and the horizontal plates, controlling the position of the spot on the screen is shown in the figure below:

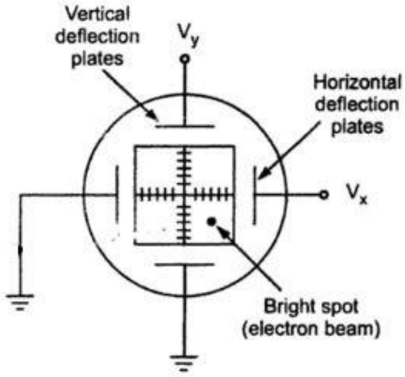


Fig.13 Arrangements of Plates in CRT

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 as short as a few microsecond, or as long as tens of seconds or even minutes. Medium persistence traces are mostly used for general purpose applications. 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. Short persistence is needed for extremely high speed phenomena.

The screen is coated with a fluorescent material called phosphor which emits light when bombarded by electrons. There are various phosphors available which differ in colour, persistence, and efficiency.

One of the common phosphor is Willemite, which is zinc, orthosilicate, ZnO+SiO2, with traces of manganese. This produces the familiar greenish trace. Other useful screen materials include compounds of zinc, cadmium, magnesium and silicon. The kinetic energy of the electron beam is converted into both light and heat energy when it hits the screen. The heat so produced gives rise to "phosphor burn" which is damaging and sometimes destructive. This degrades the light output of phosphor and sometimes may cause complete phosphor destruction. Thus the phosphor must have high burn resistance to avoid accidental damage.

Many phosphor materials having different excitation times and colours as well as different phosphorescence times are available. The type P_1, P_2, P_{11} or P_{31} are the short persistence phosphors and are used for the general purpose oscilloscopes.

Phosphor	Colour		Persistence	Relative luminance	Relative writing speed	Applications
	Under excitation	After glow				
P1	yellow- green	yellow- green	medium	45	35	General purpose
P2	blue-green	green	medium	60	70	General purpose
P4	white	white	medium to short	50	75	Black and white T.V.
P7	blue-white	yellow- green	medium-short	45	95	Radar
P11	blue-violet	blue	medium-short	25	100	Photographic recording
P15	blue-green	blue-green	visible -short	15	25	Flying spot scanners for T.V.
P19	orange	orange	long	25	3	Radar
P31	green	green	medium-short	100	75	General purpose
P33	orange	orange	very long	20	7	Radar
P39	green	green	medium- long	50	40	Computer graphics

Glass Tube: All the components of a CRT are enclosed in an evacuated glass tube called envelope. This allows the emitted electrons to move about freely from one end of the tube to the other end.

Base: The base is provided to the CRT through which the connections are made to the various parts.

6. CATHODE RAY OSCILLOCOPE (CRO)-BLOCK DIAGRAM

The block diagram of CRO is shown below:

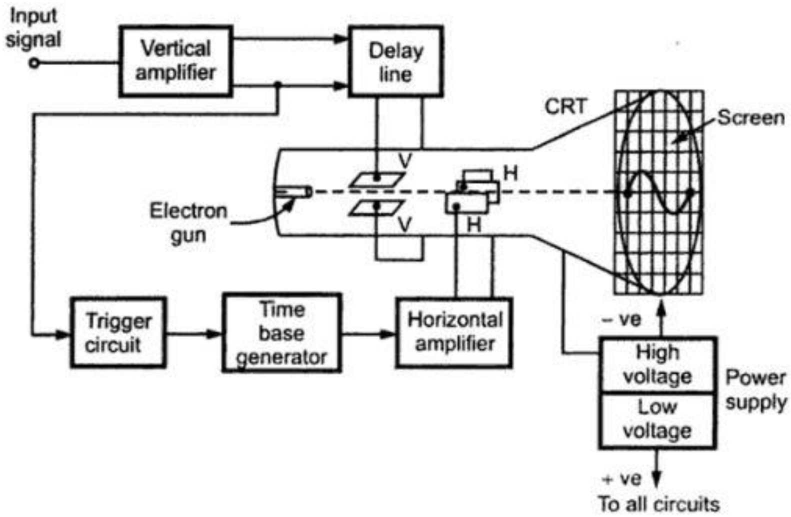


Fig.14 Block diagram of Cathode ray Oscilloscope (CRO)

Cathode Ray Tube (CRT): This is the cathode ray tube which is the heart of C.R.O. It is used to emit the electrons 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 to 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.

The block diagram of a vertical amplifier is shown in the figure below.

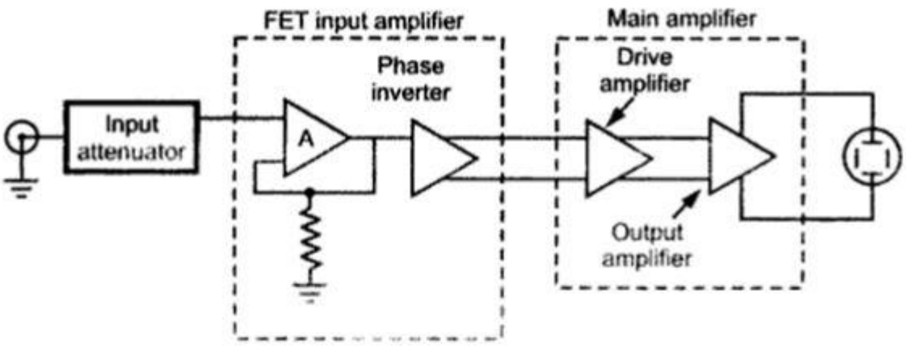


Fig.14 Vertical amplifier

It consists of several stages with overall fixed sensitivity. The amplifier can be designed for stability and required bandwidth very easily due to the fixed gain. The input stage consists of an attenuator followed by FET source follower. It has very 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 anti-phase 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 and 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 vertical 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 delayed by some time using a delay line circuit as shown in the figure below:

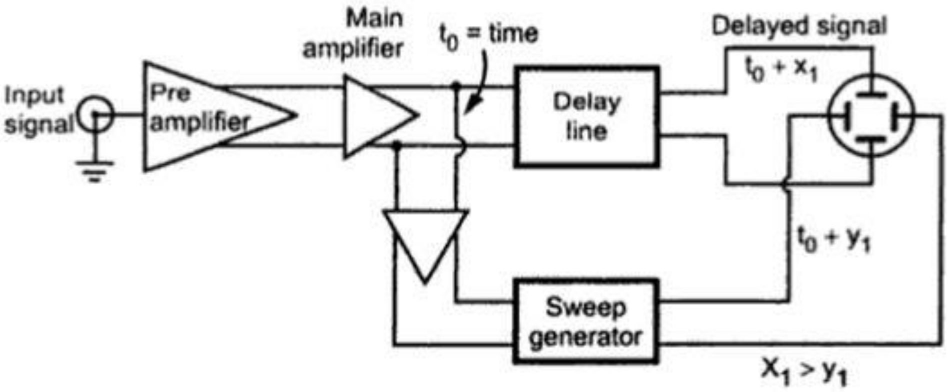


Fig.15 Delay line circuit

If the trigger pulse is picked off at a time $t = t_o$ after the signal has passed through the main amplifier then signal is delayed by x_1 nanoseconds while sweep takes y_1 nanoseconds to reach. The design of delay line is such that the delay time x_1 is higher than the time y_1. Generally x_1 is 200 nanoseconds while the y_1 is 80 nanoseconds, thus the sweep starts well in time and no part of the signal is lost.

There are two types of delay lines used in C.R.O. which are :

i) Lumped parameter delay line

ii) Distributed parameter delay line

They are discussed below:

I) Lumped parameter delay line:

Lumped parameter delay line consists of number of cascaded symmetrical LC networks called T sections Each section is capable of delaying the signal by 3 to 6 nano seconds. Such a T filter section is shown in the figure below.

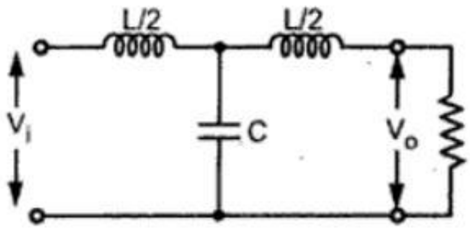


Fig.16 T-section filter.

The T-section filter acts as low pass filter having cut-off frequency given as

$$f_c = \frac{1}{\pi\sqrt{LC}}$$

If V_i consists of frequencies much less than the cut-off frequency, output signal V_0. will be a faithful reproduction of V_i but delayed by the time,

$$t_s = \frac{1}{\pi f_c} = \sqrt{LC}$$

where

t_s - Delay for a single T network

where, $t_d = nt_s$

here t_d = Total delay

n = Number of T sections A practical delay line circuit in C.R.O. is driven by push-pull amplifier and is shown in the figure

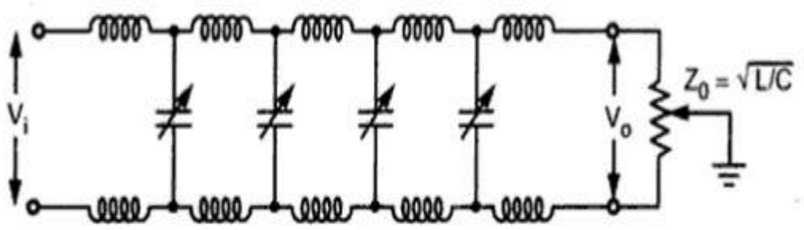


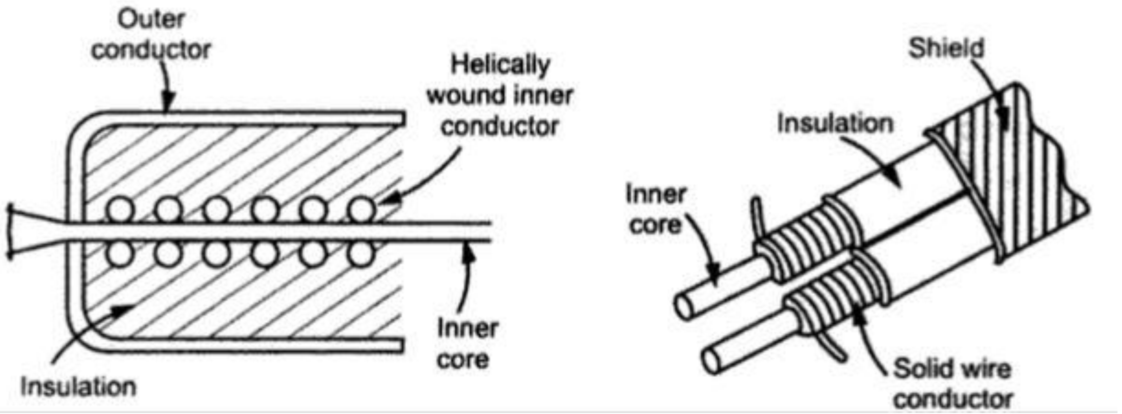
Fig.17 Practical delay line

II) Distributed parameter delay line:

It is basically a transmission line constructed with a wound helical coil on a mandrel and extruded insulation between it. It is specially manufactured co-axial cable with high inductance per unit length. The construction of such line is shown in the figure below.

The inductance can be increased by winding the helical inner conductor on ferromagnetic core. This increases the characteristics impedance Z_o and delay time. Typical parameters for helical, distributed parameter delay line are $Z_o = 1000\ \Omega$ and t_d = 180 nsec/m. The co-axial delay line is advantageous as:

i) It does not require careful adjustment as lumped parameter. ii) It requires less space.



a) Helical distributed delay line **b)Constructional details**

Fig.18 Distributed 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.

Horizontal Amplifier: The sawtooth voltage produced by the time base generator may not be of sufficient strength. Hence before giving it to the horizontal deflection plates, it is amplified using the horizontal amplifier.

Power Supply: The power supply block provides the voltages required by CRT to generate and accelerate an electron beam and voltages required by other circuits of the oscilloscope like horizontal amplifier, vertical amplifier etc.

The negative High Voltage (HV) supply has following advantages:

i) The accelerating anodes and the deflection plates are close to ground potential. This ground potential protects the operator from shocks.

ii) The deflection voltages are measured with respect to ground hence blocking or coupling capacitors are not necessary.

iii) Insulation required between controls and chasis is less.

There are two sections of a power supply block. The High Voltage (HV) section and Low Voltage (LV) section. The high voltages of the order of 1000V to 1500 V are required by CRT. Such high negative voltages are used for CRT. The low voltage is required for the heater of the electron gun, which emits the electrons. This is a positive voltage of the order of few hundred volts. This voltage is also used for other circuits of C.R.O. This is the discussion of basic block diagram of a simple C.R.O.

6.2 CRT Circuits

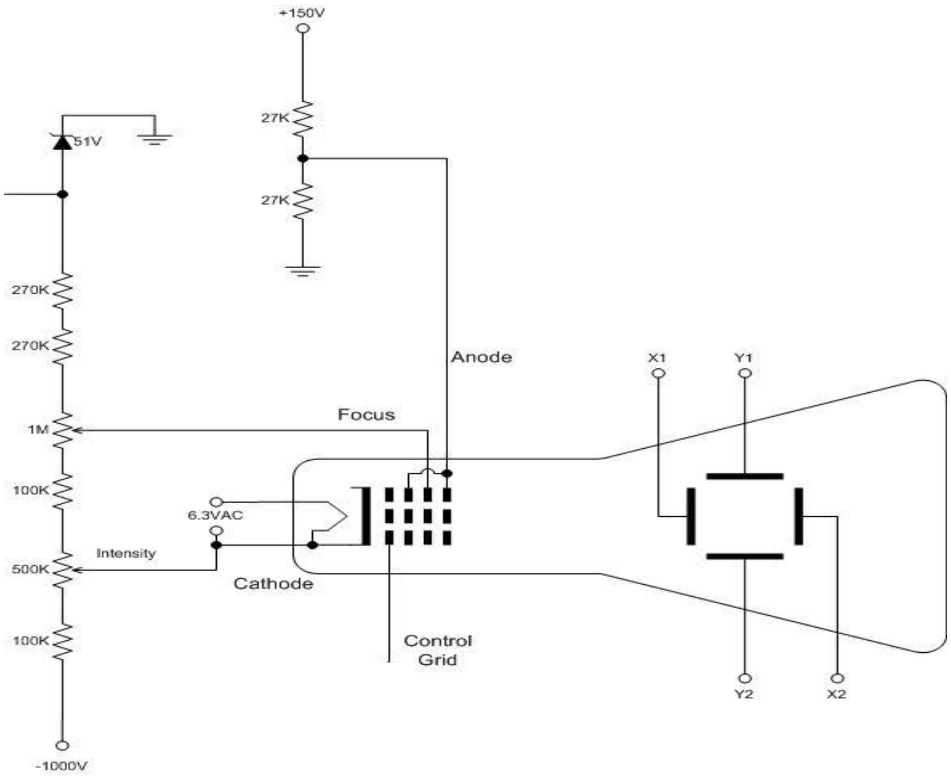


Fig.19 CRT Circuits

The CRT needs an anode to cathode supply voltage of 1000V or more to give a bright and sharp display. The deflection plates which move the dot around the screen must be at about the same voltage as the anode. So if the cathode is at or near ground potential as is common in vacuum tube circuits, then the deflection plates are 1KV above ground. If the deflection plates are to be driven by a DC coupled transistor amplifier, and the bases are to be near ground so that they may in turn be driven by low voltage circuits then the output transistors must have a V_{CEO} rating of over 1KV. Transistors with that kind of collector breakdown voltage that are also fast enough to handle megahertz signals are rare indeed.

If, on the contrary, the anode voltage is at ground, then we would need an amplifier with an output stage that could swing 60 or so volts above and below ground. While this is possible, it would be more complex than necessary. A better solution is to put the CRT anode 70 to 80 volts above (more positive than) ground and the cathode about 950 volts below ground. In this way, the drivers for the deflection plates could work from a 150 Volt supply and swing ±60 Volts from a quiescent value of about 75 Volts.

In the circuit above, the two 27KΩ resistors set the anode potential to 75V. The second and fourth grids are internally connected to the anode and along with the third or focus grid form a lens that focuses the electron beam from the cathode to a small sharp dot on the screen. The control grid is held near -1000V by circuitry not shown here. The voltage divider on the left of the drawing provides proper bias voltage for the focus grid and the cathode. The top pot adjusts focus and the lower one sets the brightness of the dot on the screen.

Vertical deflection system

The main function of this amplifier is to amplify the weak signal so that the amplified signal can produce the desired signal. To examine the input signals are penetrated to the vertical deflection plates through the input attenuator and number of amplifier stages.

The schematic arrangement of the vertical plates, controlling the position of the spot on the screen is shown in the figure below:

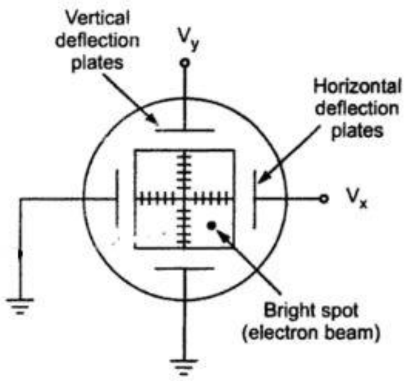


Fig.20 Arrangements of Vertical plates in CRT

The input signals are generally not strong to provide the measurable deflection on the screen. Hence the vertical amplifier stage is used to 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.

The function of vertical deflection system is to provide an amplified signal of the proper level to drive the vertical deflection plates without introducing any appreciable distortion into the system. The input sensitivity of many CROs is of the order of a few milli-volts per division and the voltage required for deflecting the electron beam varies from approximately 100 V (peak to peak) to 500 V depending on the accelerating voltage and the construction of the tube. Thus the vertical amplifier is required to provide this desired gain from milli-volt input to several hundred volt (peak to peak) output. Also the vertical amplifier should not distort the input waveform and should have good response for entire band of frequencies to be measured. The deflection plates of CRO act as plates of a capacitor and when the input signal frequency exceeds over 1 MHz, the current required for charging and discharging of the capacitor formed by the deflection plates increases. So the vertical amplifier should be capable of supplying current enough to charge and discharge the deflection plate capacitor.

As we know that electrical signal is delayed by a certain amount of time when transmitted through an electronic circuitry. In CRO, output signal voltage of the vertical amplifier is fed to the vertical plates of CRT and some of its portion is used for triggering the time base generator circuit, whose output is supplied to the horizontal deflection plates through horizontal amplifier. The whole process, which includes generating and shaping of a trigger pulse and starting of a time-base generator and then its amplification, takes time of the order of 100 ns or so. So the input signal of the vertical deflection plates of a CRT is to be delayed by at least the same or little more amount of time to allow the operator to see the leading edge of the signal waveform under study on the screen.

The block diagram of a vertical amplifier is shown in the figure below.

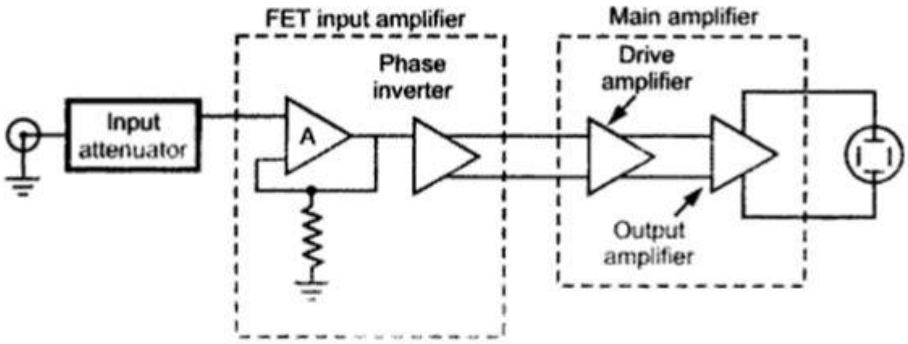


Fig.21 Vertical amplifier

It consists of several stages with overall fixed sensitivity. The amplifier can be designed for stability and required bandwidth very easily due to the fixed gain. The input stage consists of an attenuator followed by FET source follower. It has very 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 anti-phase 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 and harmonic, greater power output per tube and reduced number of defocusing and nonlinear effects.

Triggered sweep

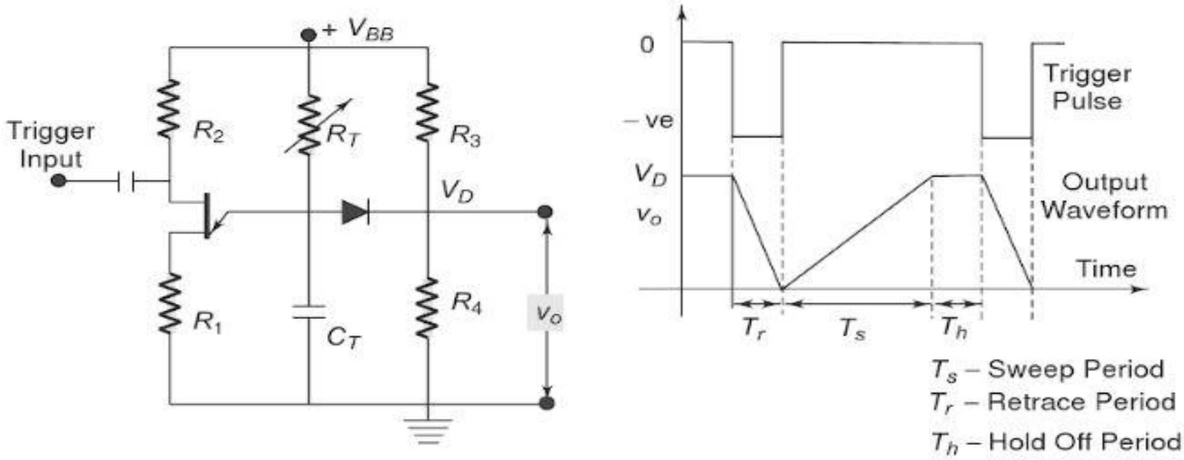


Fig.22 Triggered sweep and its output waveform

The continuous sweep is of limited use in displaying periodic signals of constant frequency and amplitude. When attempting to display voice or music signals, the pattern falls in and out of sync as the frequency and amplitude of the music varies resulting in an unstable display.

A triggered sweep can display such signals, and those of short duration, e.g. narrow pulses. In triggered mode, the input signal is used to generate substantial pulses that trigger the sweep. Thus ensuring that the sweep is always in step with the signal that drives it.

As shown in figure above, resistance R_3 and R_4 form a voltage divider such that the voltage V_D at the cathode of the diode is below the peak voltage V_P for UJT conduction. When the circuit is switched on, the UJT is in the non-conducting stage, and C_T charges exponentially through R_T towards V_{BB} until the diode becomes forward biased and conducts, the capacitor voltage never reaches the peak voltage required for UJT conduction but is clamped at V_D. If now a -ve pulse of sufficient amplitude is applied to the base and the peak voltage V_P, is momentarily lowered, the UJT fires. As a result, capacitor C_T, discharges rapidly through the UJT until the maintaining voltage of the UJT is reached; at this point the UJT switches off and capacitor C_T charges towards V_{BB}, until it is clamped again at V_D. Above figure shows the output waveform.

6.3 CRO SPECIFICATIONS

Oscilloscopes are a very common form of test equipment - possibly the most important type of test equipment. As a result it is often necessary to be able to choose one either from the test equipment store or as a rental or when buying an oscilloscope. When selecting an oscilloscope, there are many different specifications and parameters to consider, each one related to the performance. When selecting an oscilloscope, what are the most important specifications and parameters and which ones will affect the performance of the scope in the particular application.

1) Types of oscilloscope: One of the major specifications associated with buying an oscilloscope is the actual type of oscilloscope that is required. Some types of scope will be able to perform other measurements better than others; some use current technology whereas others are older; and there may be cost implications as well. Analogue, analogue storage, digital, digital storage, digital sampling, USB scopes and many more types are available.

2) Oscilloscope bandwidth specification: One important oscilloscope specification is related to the frequency or speed of the waveforms that can be measured. This is determined by the bandwidth of the oscilloscope and it is found that the capability of the oscilloscope to accurately display the waveform falls off with increasing frequency. The oscilloscope specification for bandwidth will typically be quoted in the format: Bandwidth = -3dB at 1500 MHz.

3) Vertical DC gain accuracy: It is important when measuring the amplitude of signals, to know the accuracy of the measurement that is being made. As oscilloscopes are not intended to be used instead of digital multimeters, it is not anticipated that the voltage elements of the oscilloscope specification will be as accurate.

4)Vertical channel Resolution: Digital oscilloscopes need to convert the incoming analogue signal into a digital signal. The vertical channel resolution determines the "granularity" of the signal. Most digital oscilloscopes have 8-bit resolution.

5) Rise Time Specification: Another important oscilloscope specification which needs to be accommodated is the rise time of the oscilloscope. This is a particularly important specification for any digital circuits where the edges on square waves and pulses are often of great importance. The oscilloscope must have a sufficiently fast rise time to capture the rapid transitions accurately, otherwise important information may not be displayed and the results could be misleading. The rise time of the oscilloscope is defined as the time it takes for the image to rise from 10% to 90% of the final value.

6) Oscilloscope sample rate: the sample rate oscilloscope specification is becoming a more widespread and important specification. The sample rate is specified in samples per second (S/s). The faster the oscilloscope samples the waveform, the greater the resolution of the detail on the waveform and with greater sample rates the less the likelihood that any critical information will be lost.

Oscilloscope sample rate = 2.5 x Highest frequency

7) Memory Depth: This is the memory for storing signals. The greater the memory depth the more signal it is possible to capture at the highest sample rate.

CRO Controls

The various front panel controls of a simple C.R.O. are described in this section. These are divided into four groups,

1. Basic controls
2. Vertical section
3. Horizontal section
4. Z-axis Intensity control

1.Basic Controls :

<u>1. ON-OFF :</u> The on-off switch turns on or off the C.R.O.

<u>2. Intensity :</u> This controls the intensity or brightness of the light produced by beam spot. It actually controls the number of electrons per second that are bombarding the screen which determines the brightness of the spot.

<u>3. Focus :</u> This controls the sharpness of the spot. A sharper spot is always preferred. Focusing or the spot is obtained by varying the voltage applied to the focusing anodes of the cathode ray tube.

4. Astigmatism : This is another focus control. With the help of focus control and astigmatism control, a very sharp spot can be obtained both in the centre and also at the edges of the screen. With the astigmatism control the voltage to accelerating anodes is varied.

5. Scale Illumination : Most C.R.O. s have some sort of plastic screen in front of the cathode ray tube. This screen has a grid engraved on it, giving it an appearance similar to that of graph paper. This is called graticule. This scale facilitates the measurement on the oscilloscope. The scale illumination control, illuminates the scale and hence the lines on the scale can be seen very easily.

2.Vertical Section: Most oscilloscopes have two vertical inputs. These are usually called inputs 1 and 2 or A and B. Two input signals can be applied to these two inputs and thereby both the signals can be observed on the screen simultaneously. This is very useful for comparing two signals. The following controls serve for each vertical input.

a) volts/division:

This control sets the vertical scale; that is, it determines how much the spot will be deflected by an input signals applied to vertical input terminals. The usual units are either volts per centimeter or volts per division, where division refers to the grid marks on the screen. The actual input voltage can be found by measuring the deflection and multiplying it by the scale factor. Thus, if the scale control was set to 5 V/cm and deflection is 1.3 cm, the input would be 6.5 V. The control is shown in the figure below

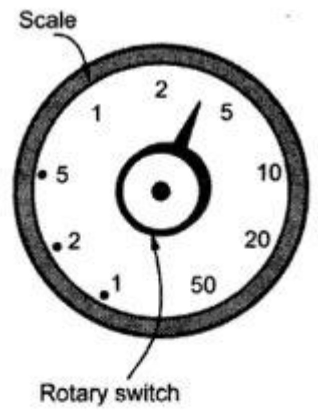


Fig.23 Volts/division selection

Suppose the alternating voltage signal of amplitude 10 V is to be displayed. Then if volts/division are selected as 10 then it will be displayed as shown in the figure (a) while if volts/division = 5 is selected, it will be displayed as shown in the figure (b)

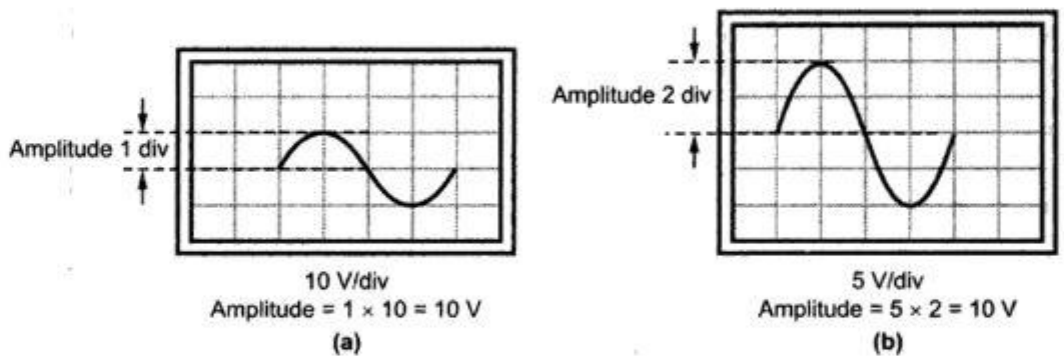
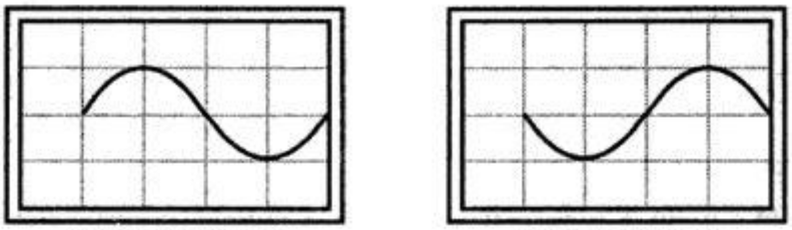


Fig.24 Effects of volts/division

b) Invert:

This control inverts the input signal ; that is, it multiplies it by -1. Then positive input voltages become negative and cause downward deflections. The effect of invert is shown in the figure below



(a) Original signal (b) With invert ON

Fig.25 The effect of Invert

c) Position :

With the help of this control, the pattern obtained on the screen can be shifted, as a whole, vertically upwards or downwards. This is achieved by adding a d.c. offset voltage to the input signal.

d) X 10 (Multiplied by 10):

This control makes the gain of the vertical amplifier 10 times as great as normal, it changes the scale factor by factor of 10. Thus if the X 10 switch is turned 'ON' and the scope is set on 0.05 V/cm, if the actual scale factor is 0.005 V/cm or 5 mV/cm.

e) Vertical Coupling :

This switch controls the coupling to the vertical amplifier. The usual choices are A.C., D.C., or ground. The meaning of these various positions are as follows:

i. A.C. : The vertical amplifier is a.c. coupled to the input. Thus the d.c. component of the input is blocked, and only the a.c. components of the input signal deflect the beam vertically. This allows to observe small a.c. signals or large d.c. background.

ii. D.C. : The vertical amplifier has d.c. coupling throughout, so that the deflection corresponds to both the a.c. and the d.c. components of the input.

iii. GROUND : The input to the amplifier is grounded. There will be no vertical deflection. If no voltage is applied to horizontal plates, the spot will be at the position corresponding to ground. It is useful for measuring voltage with respect to ground.

f) Vertical Mode Control: The control serves for the vertical section of the scope as a whole. Assume that two input signals are simultaneously applied to the two vertical inputs of the scope. Then this switch determines what is displayed on the screen. Thus usual choices are :

1 only, 2 only, 1+2;1-2, Alternate, and Chop. The meaning of each of these is described briefly below:

i) 1 only : Only the signal at input 1 is displayed.

ii) 2 only : Only the signal at input 2 is displayed.

iii) 1 + 2 : Sum : The sum of the inputs 1 and 2 is displayed.

iv) 1-2 : Sum : Difference : The difference between input 1 and input 2 is displayed.

v) Alternate : Input 1 is displayed first, then input 2 is displayed, then input 1again and so on. By using the vertical position control, the two traces can be separated vertically, and thus, relations between the two signals can be studied.

vi) Chop : In this mode first input 1 is displayed for a fraction of a microsecond, then input 2 for a fraction of microsecond, then input 1 again, and so on. In this way, plots of both inputs can be drawn at the same time. The chop mode is useful with low frequency signals, while the alternate mode is useful for high frequency signals.

3. Horizontal section :

a) Time Base Control: Very often the oscilloscope is used to observe the waveform of time varying signals. Most of the horizontal section of the scope is devoted to generating a time base for such signals. The time base control is calibrated in terms of time per centimeter or time per division. A typical unit might be 0.10 msec/cm, meaning that horizontal deflection of the spot will be 1 cm in 0.1 msec. The usual range on a scope is from about 0.1 sec/cm to 20 to 50 nsec/cm. This is shown in the figure.

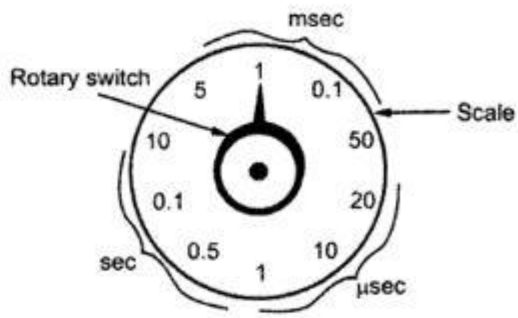


Fig.26 Times/division selection

If signal has time period of 20 msec then with two different time base control selections, it can be displayed as shown in the figure below (a) and (b).

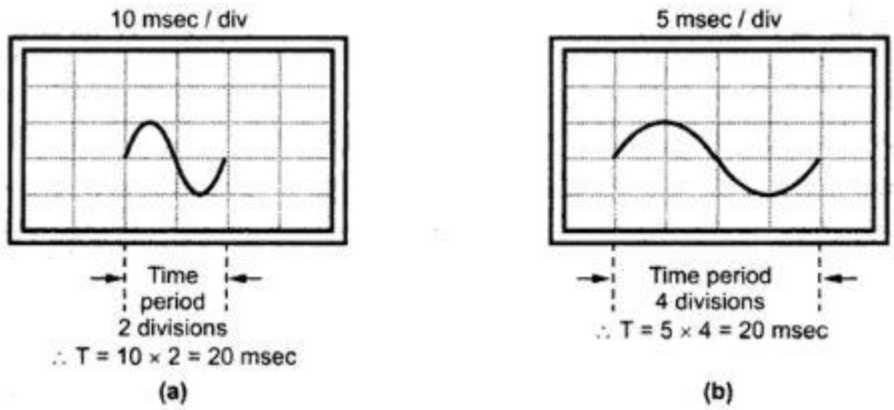


Fig.27 Effect of Time base control.

b) Position : This knob can be used to shift the display, as a whole to left or right.

c) Synchronization: It has been mentioned earlier that to obtain the stationary pattern on the screen, the synchronization is must. It is used to operate the time base generator such that the frequency of saw tooth voltage is an integral multiple of input signal frequency. There are various signals which can be applied to the trigger circuit. The signals can be selected using a synchronous selector switch. The types of signals which can be selected are:

i) Internal : The trigger is obtained from signal being measured through the vertical amplifier

ii) Line : The input to the trigger circuit is from a.c mains (say 230 V, 50 Hz) supply. This is useful when observing the signals which are synchronized with power line, such as ripple in a power line.

iii) External : The input to the trigger circuit is from the external trigger circuit.

d) Sweep Selector : When the sweep selector switch S, is in linear position, the horizontal amplifier receives an input from the saw tooth sweep generator which is triggered by the synchronous amplifier. The external signal also can be applied to the horizontal deflecting plates, by putting a selector switch S, to the external position.

4. Z-axis Intensity control:

It is used for brightening the display. Periodic positive pulses are applied to the grid and alternatively negative pulses are applied to cathode, to brighten the beam during its sweep period. This control is obtained by inserting a signal between the ground and the control grid or ground and the cathode.

6.4 MEASUREMENTS ON CRO

The various characteristics of an input signal and the properties of the signal can be measured using C.R.O. The various parameters which can be measured using C.R.O. are voltage, current, period, frequency, phase, amplitude, peak to peak value, duty cycle etc. Let us discuss the amplitude, frequency and phase measurements using C.R.O.

(1) Voltage measurement:

The C.R.O. includes the amplitude measurement facilities such as constant gain amplifiers and the calibrated shift controls. The waveform can be adjusted on the screen by using shift controls so that the measurement of divisions corresponding to the amplitude becomes easy. Generally to reduce the error, peak to peak value of the signal is measured and then its amplitude and r.m.s. value is calculated.

To measure the amplitude use the following steps:

1. Note down the selection in volts/ division from the front panel, selected for measurement

2. Adjust shift control to adjust signal on screen so that it becomes easy to count number of divisions corresponding to peak to peak value of the signal.

3. Note down peak to peak value in terms of the number of divisions on screen.

4. Use the following relation to obtain peak to peak value in volts.

$$V_{p\text{-}p} = (\text{Number of divisions or units noted}) \times \left(\frac{\text{volts}}{\text{divisions}}\right)$$

5.The amplitude can then be calculated as :

$$V_m = \text{Amplitude} = \frac{V_{p\text{-}p}}{2}$$

while the rms value of sinusoidal wave can be given as:

$$V_{RMS} = \frac{V_m}{\sqrt{2}} = \frac{V_{p\text{-}p}}{2\sqrt{2}} \quad \text{only for sinusoidal signals}$$

(2) Current measurement:

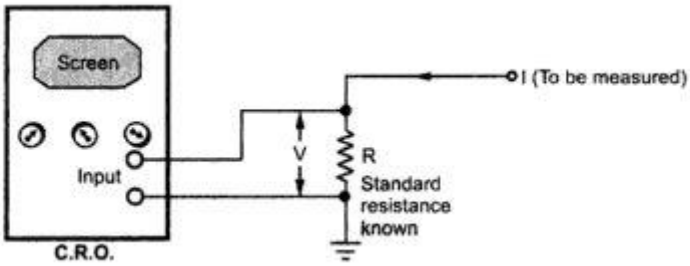


Fig.28 Arrangement for current measurement

The CRO is basically voltage indicating device. Hence to measure the current, the current is passed through a known standard resistance. The voltage across resistance is displayed on CRO and is measured. This measured voltage divided by the known resistance gives the value of the unknown current. The arrangement is shown in the figure above. Current is obtained using formula,

$$I = \frac{V_{measured} \text{ on C.R.O.}}{R}$$

(3) Period & Frequency measurement:

In such measurement, the waveform is displayed on the screen such that one complete cycle is visible on the screen. Thus accuracy increases if a single cycle occupies as much as the horizontal distance on the screen.

Note the time/ division selected on the front panel. Then the period of the waveform can be obtained as,

$$T = (\text{Number of divisions occupied by 1 cycle}) \times \left(\frac{\text{time}}{\text{division}}\right) = \text{time period}$$

The frequency is the reciprocal of the period. That is given by:

$$f = \frac{1}{T}$$

This is the method of frequency measurement without Lissajous pattern.

6.5 MEASUREMENT OF PHASE AND FREQUENCY USING LISSAJOUS PATTERNS:

It is interesting to consider the characteristics of patterns that appear on the screen of a CRT when sinusoidal voltages are simultaneously applied to horizontal and vertical plates. These patterns are called 'Lissajous Patterns'. When two sinusoidal voltages of equal frequency which are in phase with each other are applied to the horizontal and vertical deflection plates, the pattern appearing on the screen is a straight line as its clear from figure below

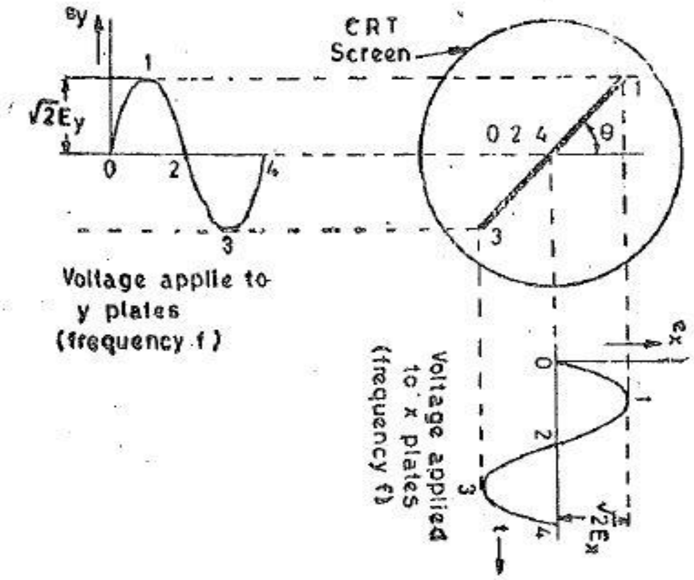


Fig.29 Lissajous pattern with equal frequency voltages and zero phase shift

Thus when two equal voltages of equal frequency but with 90^0 phase displacement are applied to a CRO, the trace on the screen is a circle. This is shown in figure below

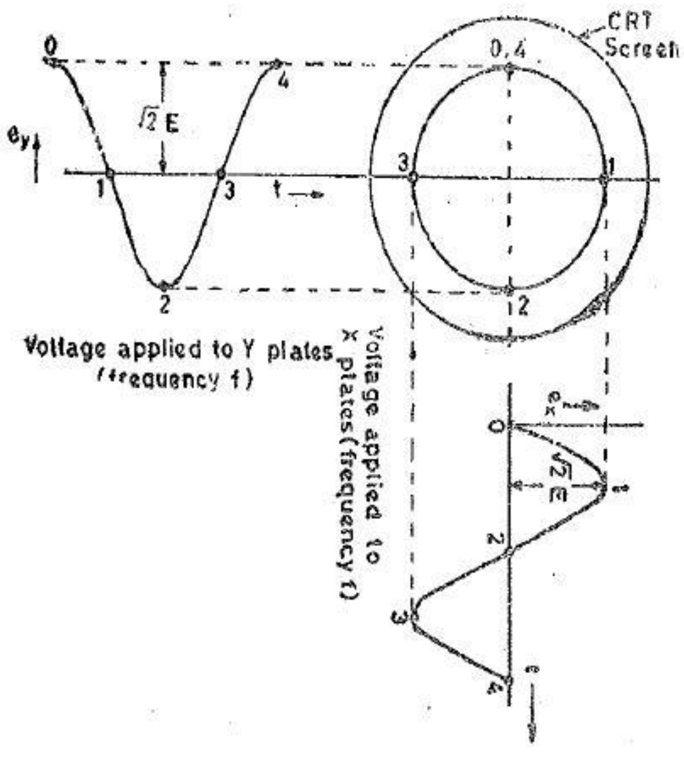


Fig.30 Lissajous pattern with equal voltages of equal frequency and phase shift of 90^0

When two equal voltages of equal frequency but with a phase shift ϕ (not equal to 0° or 90°) are applied to a CRO we obtain an ellipse as shown in figure below. An ellipse is also obtained when unequal voltages of same frequency are applied to the CRO.

A number of conclusions can be drawn from the above discussions. When two sinusoidal voltages of same frequency are applied :

(i) A straight line results when the two voltages are equal and are either in phase with each other or 180° out of phase with each other. The angle formed with the horizontal is 45° when the magnitudes of voltages are equal. An increase in the vertical detection voltage causes the line to have an angle greater than 45^0 with the horizontal. On the other hand a greater horizontal voltages makes the angle less than 45° with the horizontal. :

(ii) Two sinusoidal waveforms of the same frequency produce a Lissajous pattern, which may be a straight line, a circle or an ellipse depending upon the phase and magnitude of the voltages.

A circle can be formed only when the magnitude of the two signals are equal and the phase difference between them is either 90° or 270^0. However, if the two voltages are not equal and/or out of phase an ellipse is formed. If the Y voltage is larger, an ellipse with vertical major axis is formed while if the X plate voltage has a greater magnitude, the major axis of the ellipse lies along horizontal axis.

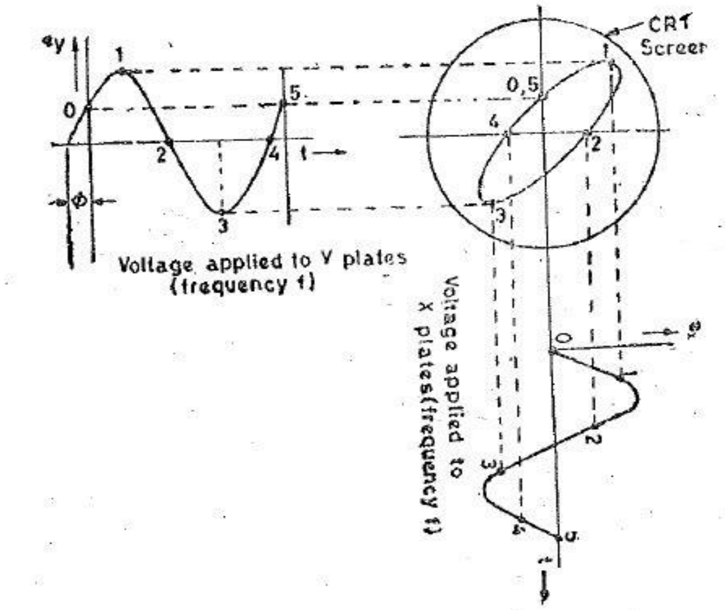


Fig.31 Lissajous pattern with two equal voltages of same frequency and phase shift of ϕ

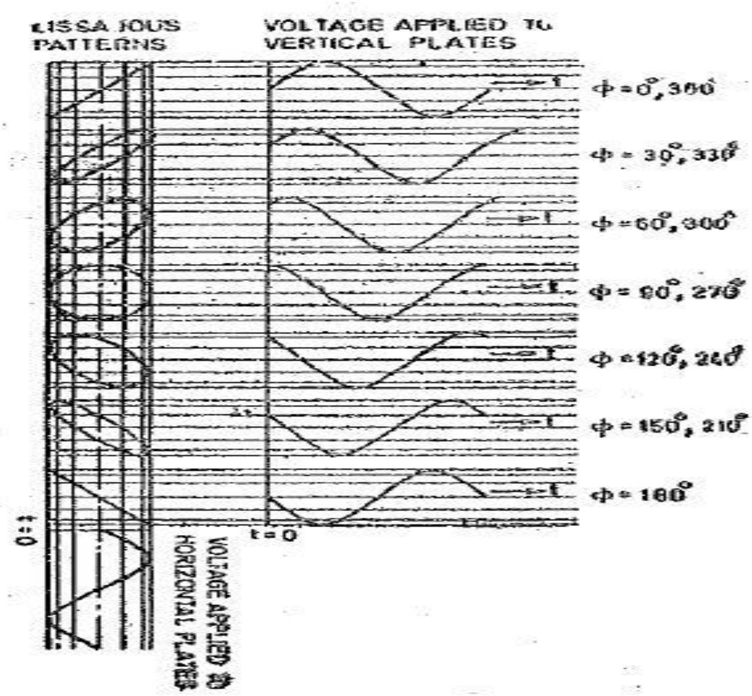


Fig: Lissajous patterns with different phase shifts

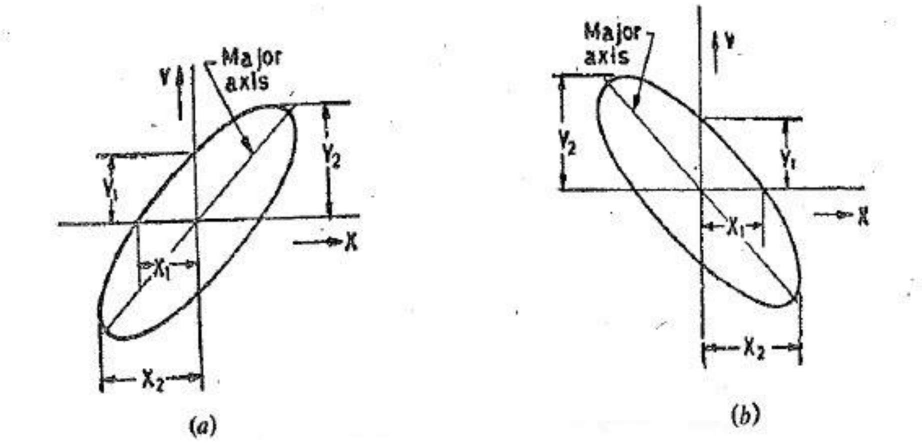


Fig.32 Determination of angle of Phase shift

The sine of phase angle between the voltages is given by

$$\sin\phi = \frac{Y_1}{Y_2} = \frac{X_1}{X_2}$$

For convenience, the gains of the vertical and horizontal amplifiers are adjusted so that the ellipse fits exactly into a square marked by the lines on the graticule. If the major axis of the ellipse lies in the first and third quadrants (i.e., its slope is positive) as in figure (a), the phase angle is either between 0° to 90° or between 270° to 360°. When the major axis of ellipse lies in second and fourth quadrants i.e., when its slope is negative as in figure (b), the phase angle is either between 90° and 180° or between 180° and 270°.

Frequency Measurements: Lissajous patterns may be used for accurate measurement of frequency. The signal, whose frequency is to be measured, is applied to the Y plates. An accurately calibrated standard variable frequency source is used to supply voltage to the X plates, with the internal sweep generator switched off. The standard frequency is adjusted until the pattern appears as a circle or an ellipse, indicating that both signals are of the same frequency. Where it is not possible to adjust the standard signal frequency to the exact frequency of the unknown signal, the standard is adjusted to a multiple or a submultiple of the frequency of the unknown source so that the pattern appears stationary.

Let us consider an example Suppose sine waves are applied to X and Y plates as shown in figure below

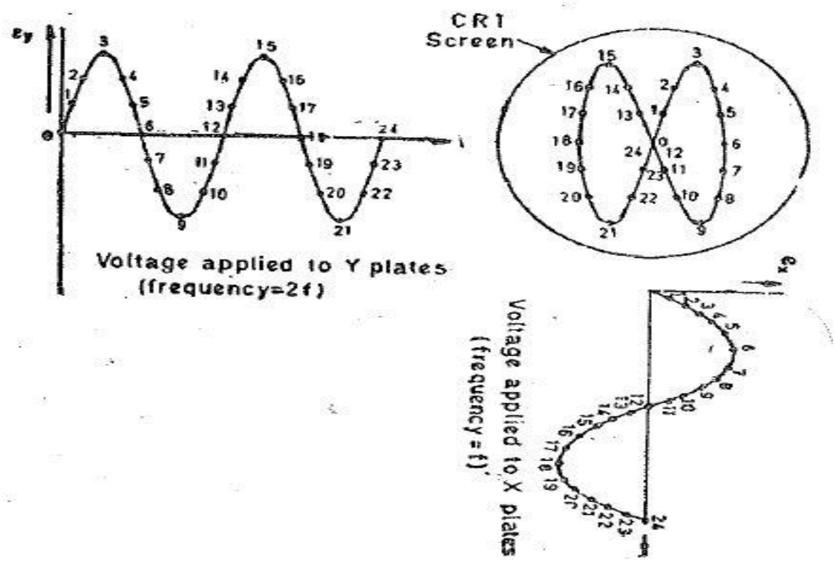


Fig.33 Lissajous patterns with frequency ratio 2:1

Let the frequency of wave applied to Y plates is twice that of the voltage applied to X plates. This means that the CRT spot travels two complete cycles in the vertical direction against one in the horizontal direction. The two waves start at the same instant. Lissajous pattern may be constructed in the usual way and a 8 shaped pattern with two loops is obtained. If the two waves do not start at the same instant we get different patterns for the same frequency ratio. The Lissajous patterns for other frequency ratios can be similarly drawn. Some of these patterns are shown in figure below.

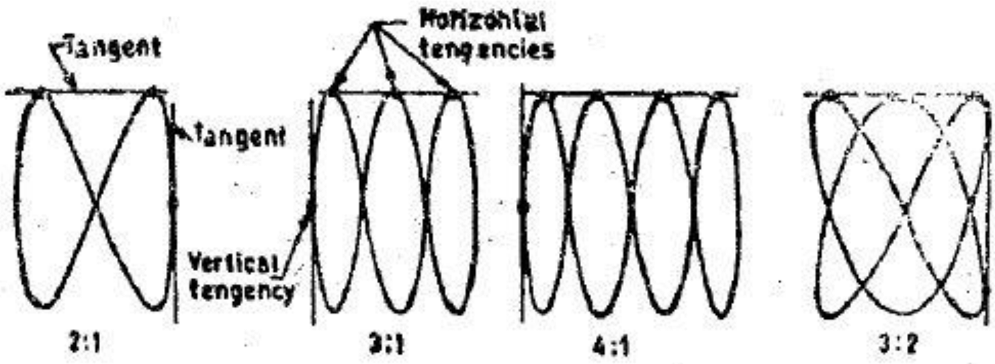


Fig.34 Lissajous patterns with different frequency ratios

It can be shown that for all the above cases, the ratios of the two frequencies is :

$$\frac{f_v}{f_x} = \frac{\text{number of times tangent touches top or bottom}}{\text{number of times tangent touches either side}}$$

$$= \frac{\text{number of horizontal tangencies}}{\text{number of vertical tangencies}}$$

where, f_V = frequency of signal applied to Y-plates

f_X = frequency of signal applied to X-plates

7. STORAGE CRO, DIGITAL STORAGE OSCILLOSCOPE

The storage type CRO is rapidly becoming one of the most useful tools in the presentation of very slowly swept signals and finds many applications in the mechanical and biomedical fields. In the conventional CRT the persistence of the phosphor ranges from micro seconds to perhaps seconds. In applications where the persistence of the screen is smaller than the rate at which the signal sweeps across the screen, the start of the display will have disappeared before the end of the display is written.

With the variable-persistence or storage CRO, the slowly swept trace can be kept on display continuously by adjusting the persistence of the CRT screen to match the sweep time. Persistence times much greater than a few seconds or even hours, are available, making it possible to store events on a CRT screen. The storage CRO uses a special CRT, called the storage tube. This special CRT contains all the elements of a conventional CRT, such as the electron gun, the deflection plates, and a phosphor screen, but in addition holds a number of special electrodes A schematic representation of one type storage tube is given in figure below.

The storage mesh or storage target, mounted just behind the phosphor screen, is a conductive mesh covered with a highly resistive coating of magnesium fluoride. The write gun is a high-energy electron gun, similar to the conventional gun, giving a narrow focussed beam which can be deflected and used to write the information to be stored. The write gun etches a positively charged pattern on the storage target by knocking off secondary-emission electrons. Because of the excellent insulating properties of the magnesium fluoride coating, this positively charged pattern remains exactly in the same position on the storage target where it was first deposited. The electron beam, which is deflected in the conventional manner both in the horizontal and the vertical directions, therefore traces out the waveform pattern on the storage target.

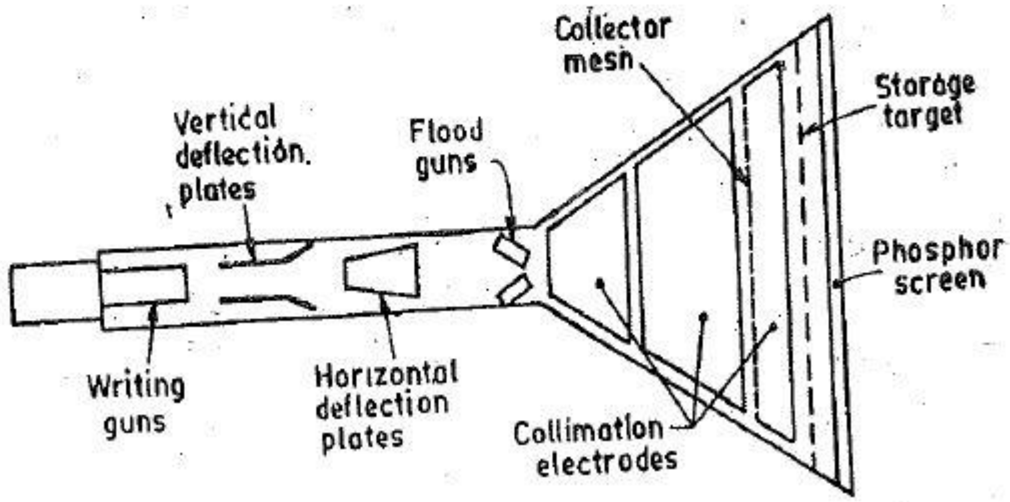


Fig.35 Schematic diagram of Storage type CRT

The stored pattern may be made available for viewing at a later time by the use of two special electron guns, called flood guns. The flood guns are placed inside the CRT in a position between the deflection plates and the storage target and they emit low-velocity electrons over a large area towards the entire screen. When the flood guns are switched on (the viewing mode), low energy electrons are sprayed towards the screen. The electron trajectories are adjusted by the collimation electrodes which constitute a low voltage electrostatic lens system, so that the flood-electrons cover the entire screen area. Most of the flood electrons are collected by the collector mesh and therefore never reach the phosphor screen. In the area near the stored positive charge on the storage target, the positive field pulls some of the flood-electrons through the storage mesh and these electrons continue to hit the phosphor. The CRT display therefore will be an exact copy of the pattern which was initially stored on the target and the display will be visible as long as the flood guns continue emission of low-energy electrons. To erase the pattern which is etched on the storage mesh, a negative voltage is applied to the storage target, neutralizing the stored positive charge.

To obtain variable persistence, the erase voltage is applied in the form of pulses instead of a steady d.c. voltage. By varying the width of these pulses the rate of erasing is varied. The variable-persistence control on the front panel of the scope is then the width control of the erase-pulse generator.

7.1 SAMPLING OSCILLOSCOPE

An ordinary oscilloscope has a Bandwidth of 10 MHz. The high frequency (HF) performance can be improved by means of sampling the input waveform and reconstructing its shape from the sample, i.e. the signal to be observed is sampled and after a few cycles the sampling point is advanced and another sample is taken. The shape of the waveform is reconstructed by joining the sample levels together. The sampling frequency may be as low as 1/10th of the input signal frequency (if the input signal frequency is 100 MHz, the bandwidth of the CRO vertical amplifier can be as low as 10 MHz). As many as 1000 samples are used to reconstruct the original waveform. Figure below shows a block diagram of a sampling oscilloscope.

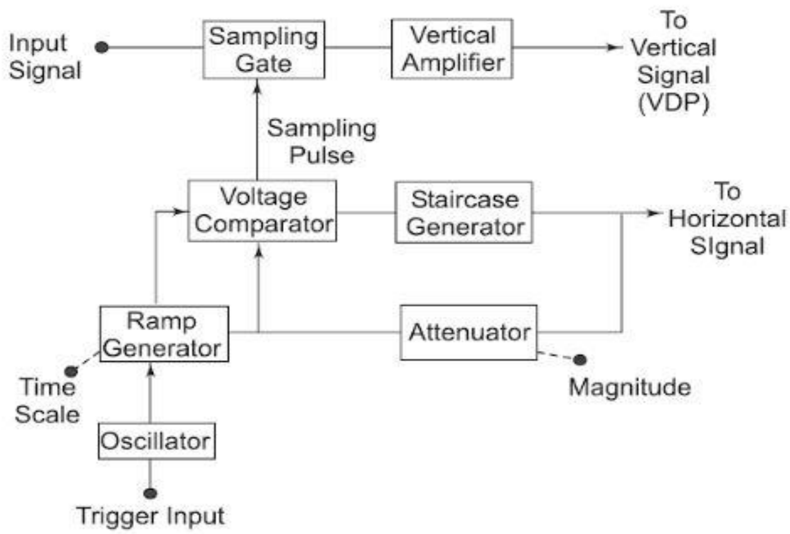


Fig.36 Block diagram of Digital Sampling Oscilloscope

The input waveform is applied to the sampling gate. The input waveform is sampled whenever a sampling pulse opens the sampling gate. The sampling must be synchronized with the input signal frequency. The signal is delayed in the vertical amplifier, allowing the horizontal sweep to be initiated by the input signal. The corresponding waveforms are also shown in below figure

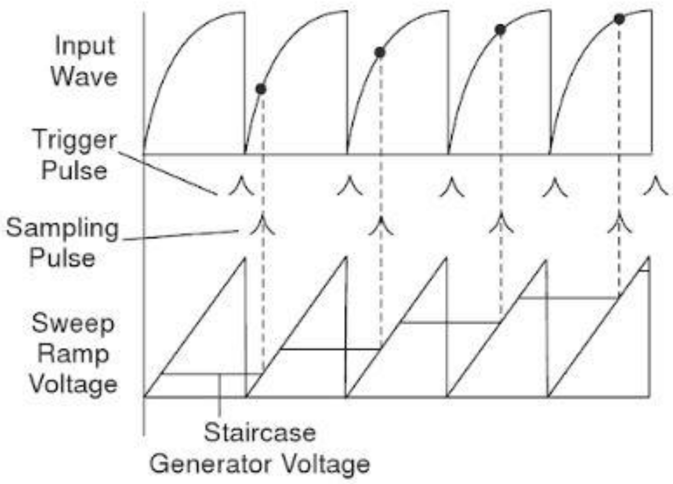


Fig.37 Various waveforms at each block of a sampling oscilloscope.

At the beginning of each sampling cycle, the trigger pulse activates an oscillator and a linear ramp voltage is generated. This ramp voltage is applied to a voltage comparator which compares the ramp voltage to a staircase generator, When the two voltages are equal in amplitude, the staircase advances one step and a sampling pulse is generated, which opens the sampling gate for a sample of input voltage.

The resolution of the final image depends upon the size of the steps of the staircase generator. The smaller the size of the steps the larger the number of samples and higher the resolution of the image.

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SCHOOL OF ELECTRICAL AND ELECTRONICS

DEPARTMENT OF ELECTRONICS AND INSTRUMENTATION

UNIT – V

ELECTRONIC MEASUREMENT AND INSTRUMENTATION – SIC 1305

UNIT V

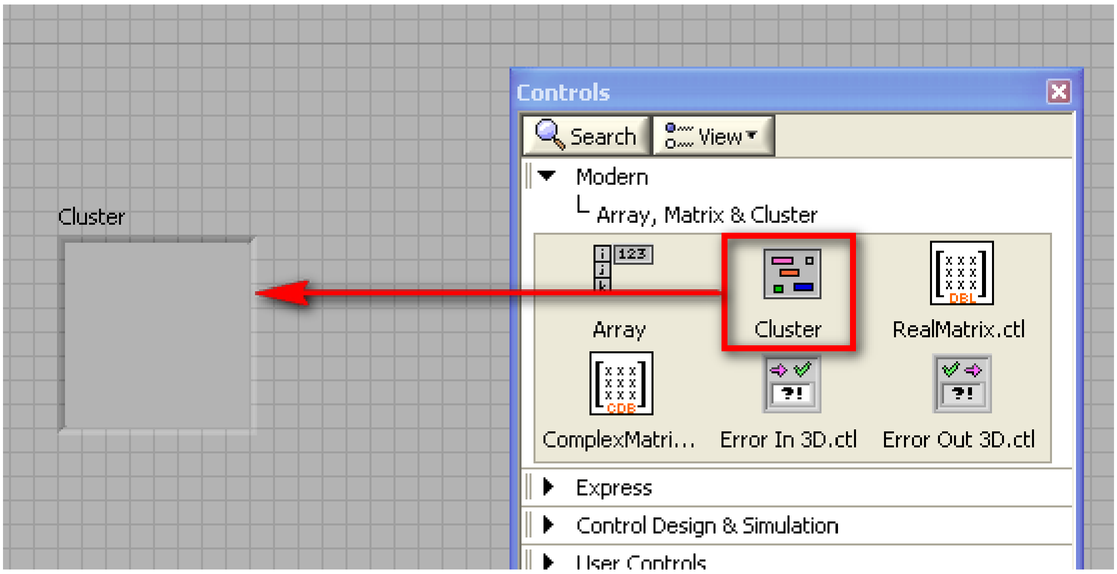
VIRTUAL INSTRUMENTATION

1. CLUSTERS IN LABVIEW

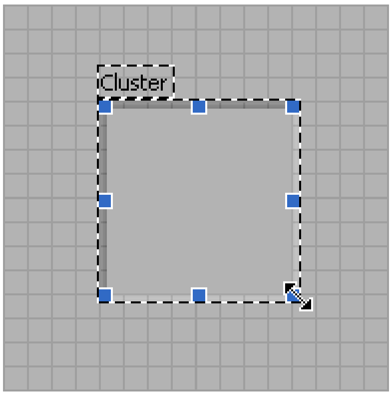
Clusters are group of data elements of mixed types. An example of a cluster is the LABVIEW error cluster, which combines a Boolean value, a numeric value and a string. A cluster is similar to a record or a struct in text-based programming languages.

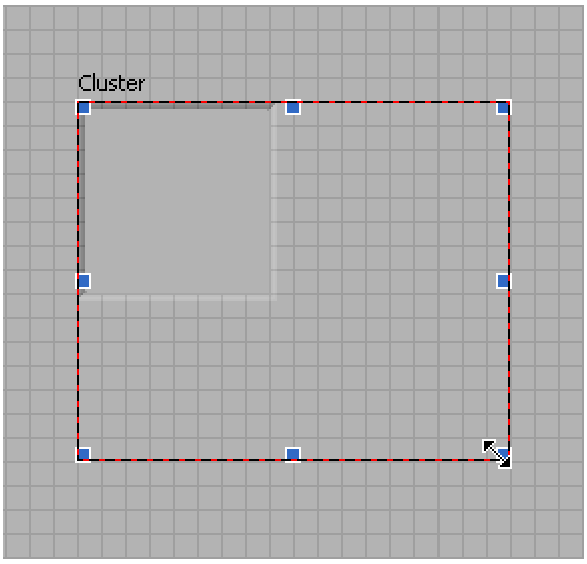
Creating Clusters

1. Create a new VI.
2. Right-click on the front panel to display the **Controls** palette.
3. On the **Controls** palette, navigate to **Modern»Array, Matrix, & Cluster** and drag the **Cluster** shell onto the front panel.

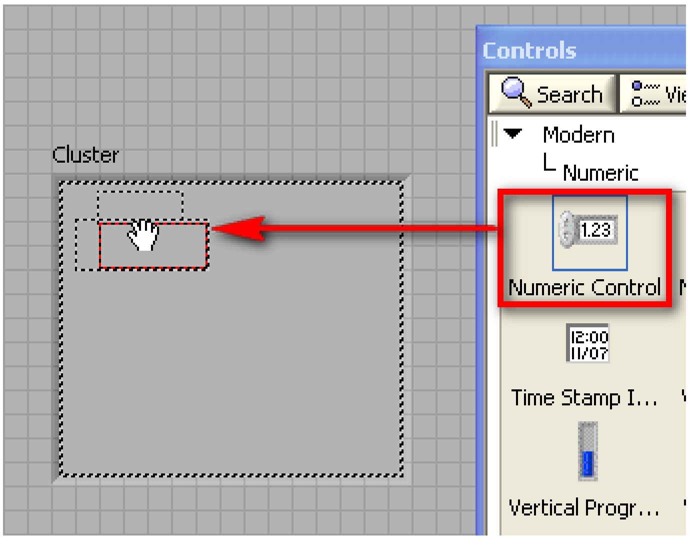


4. Resize the **Cluster** shell so that it is big enough to contain multiple elements.

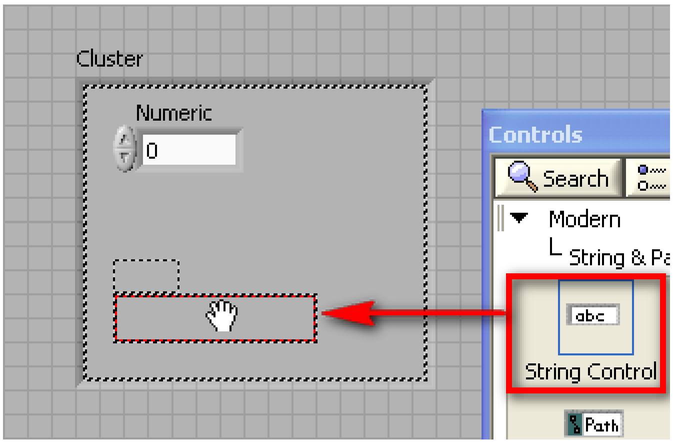




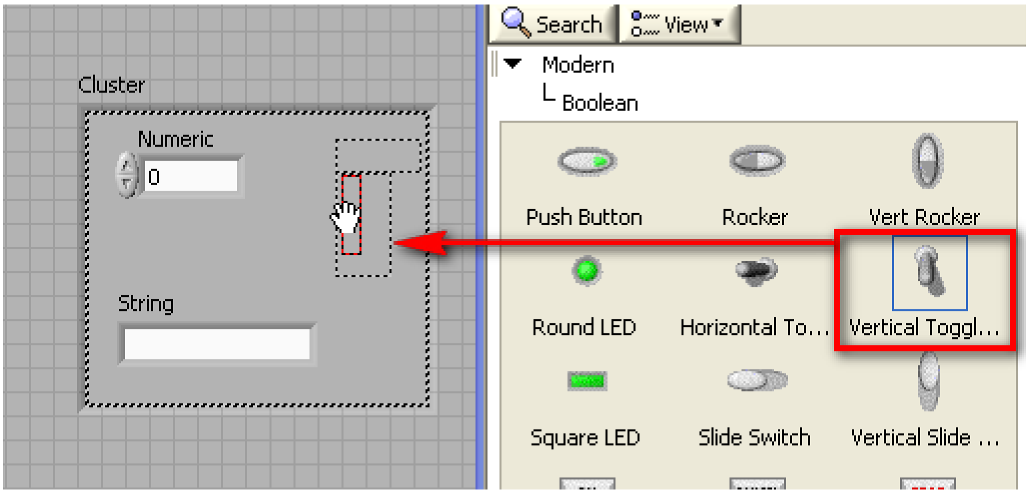
5. On the **Controls** palette, navigate to **Modern»Numeric** and drag and drop a numeric control inside the Cluster shell.



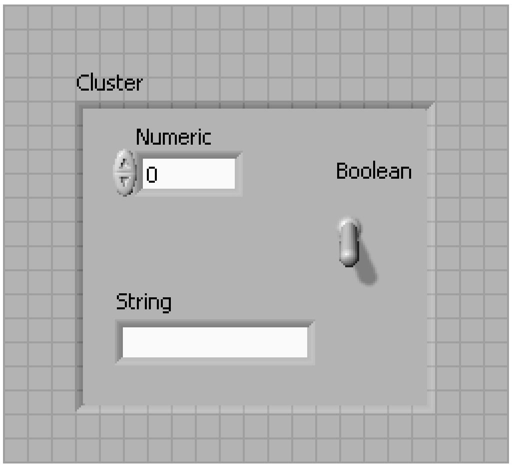
6. On the **Controls** palette, navigate to **Modern»String & Path** and drag and drop a **String Control** inside the Cluster shell.



7. On the **Controls** palette, navigate to **Modern»Boolean** and drag and drop a **Vertical Toggle Switch** inside the Cluster shell.



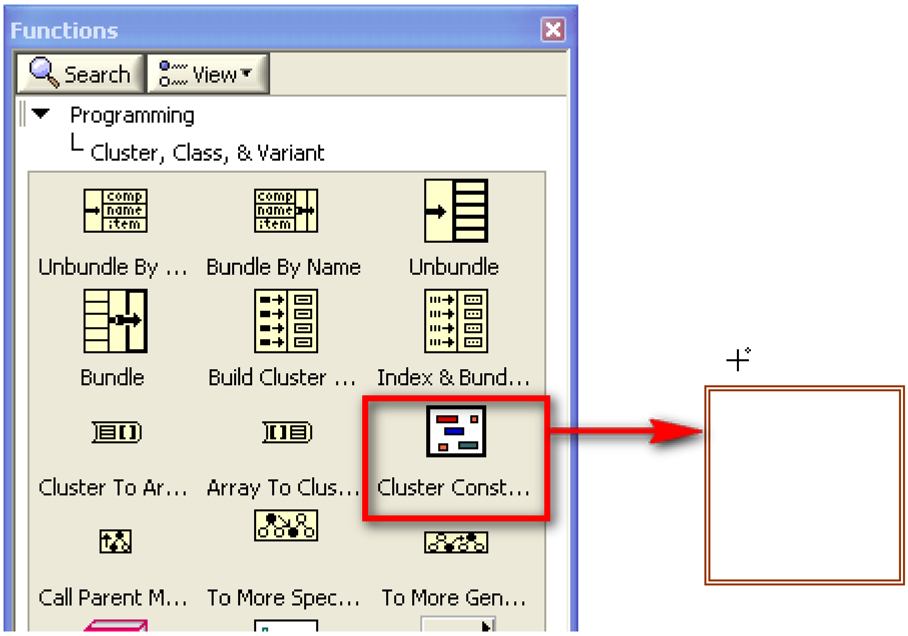
8. Our cluster should now look similar to the one shown below.



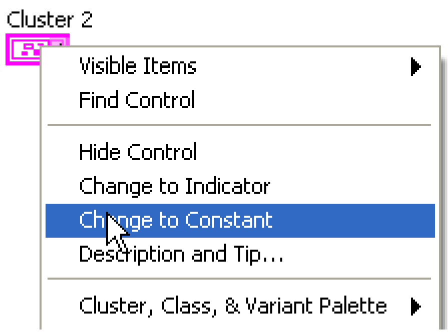
We can now wire the numeric, string, and Boolean controls throughout the block diagram with one wire rather than three separate wires.

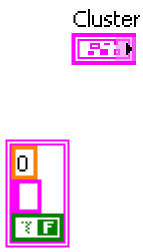
Creating Cluster Constants

Similar to array constants, we can use cluster constants to store constant data or as a basis for comparison with another cluster. Create cluster constants the same way you created array constants in the steps discussed earlier.



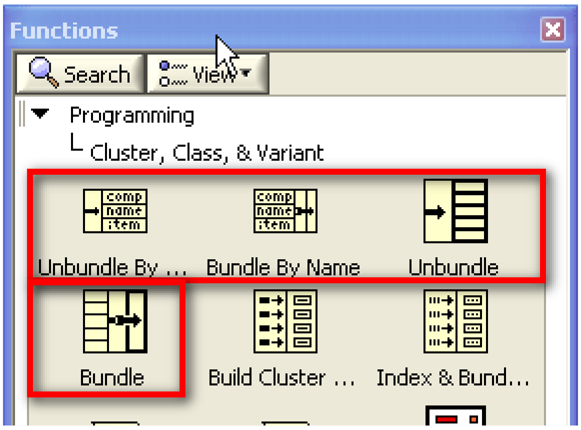
If we already have a cluster control or indicator and want to make a cluster constant that contains the same data types, make a copy of the cluster control or indicator on the block diagram and then right-click on the copy and select **Change to Constant** from the shortcut menu.



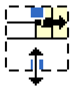


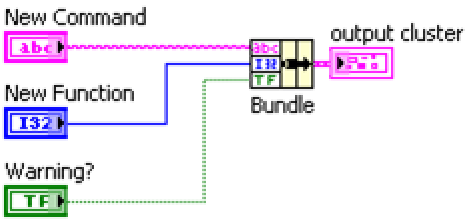
Cluster Functions

This section examines four main cluster functions often used to manipulate clusters. These are the **Bundle**, **Unbundle**, **Bundle By Name**, and **Unbundle By Name** functions.

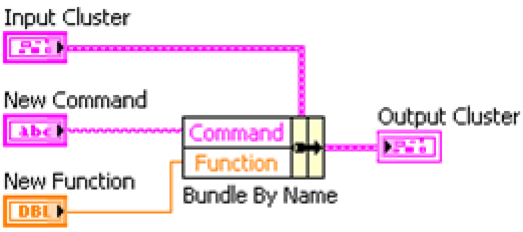


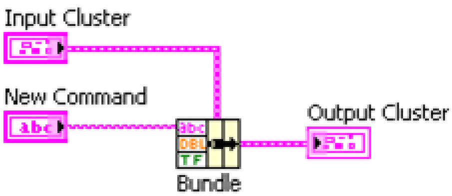
Use the **Bundle** function to assemble a cluster from individual elements. To wire elements into the **Bundle** function, use our mouse to resize the function or right-click on the function and select **Add Input** from the shortcut menu.



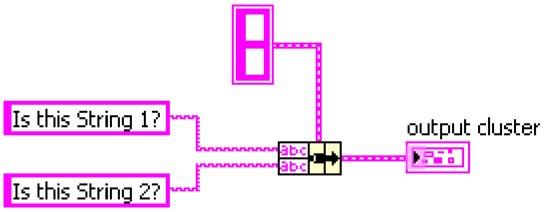


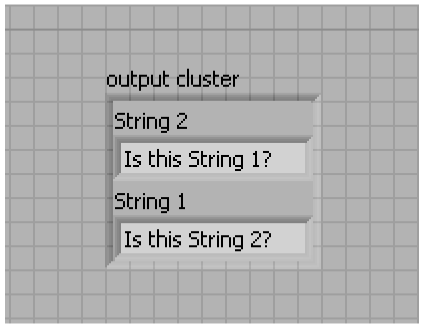
Use the **Bundle By Name** or the **Bundle** function to modify an existing cluster. We can resize the **Bundle By Name** function in the same manner as the **Bundle** function.



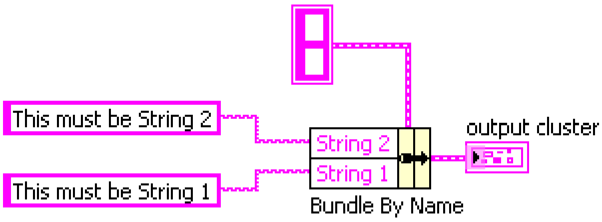


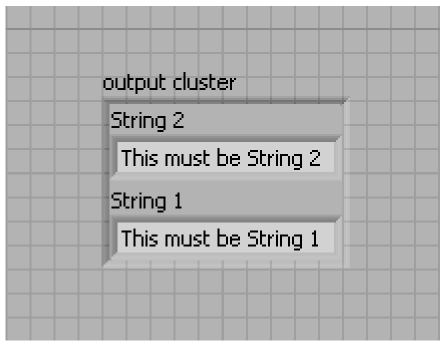
The **Bundle By Name** function is very useful when modifying existing clusters because it lets you know exactly which cluster element we are modifying. For example, consider a cluster that contains two string elements labeled “String 1” and “String 2.” If you use the **Bundle** function to modify the cluster, the function terminals appear in the form of pink abc’s. We do not know which terminal modifies “String 1” and which terminal modifies “String 2.”



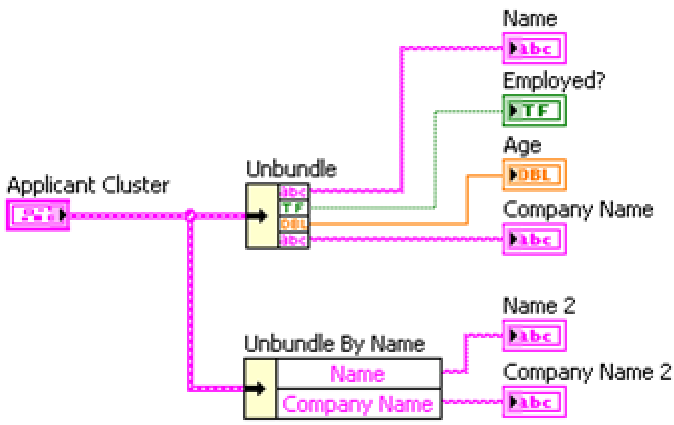


However, if we use the **Bundle By Name** function to modify the cluster, the function terminals display the element label so that you know which terminal modifies "String 1" and which terminal modifies "String 2."





Use the **Unbundle** function to disassemble a cluster into its individual elements. Use the **Unbundle by Name** function to return specific cluster elements you specify by name. We can also resize these functions for multiple elements in the same manner as the **Bundle** and **Bundle By Name** functions.

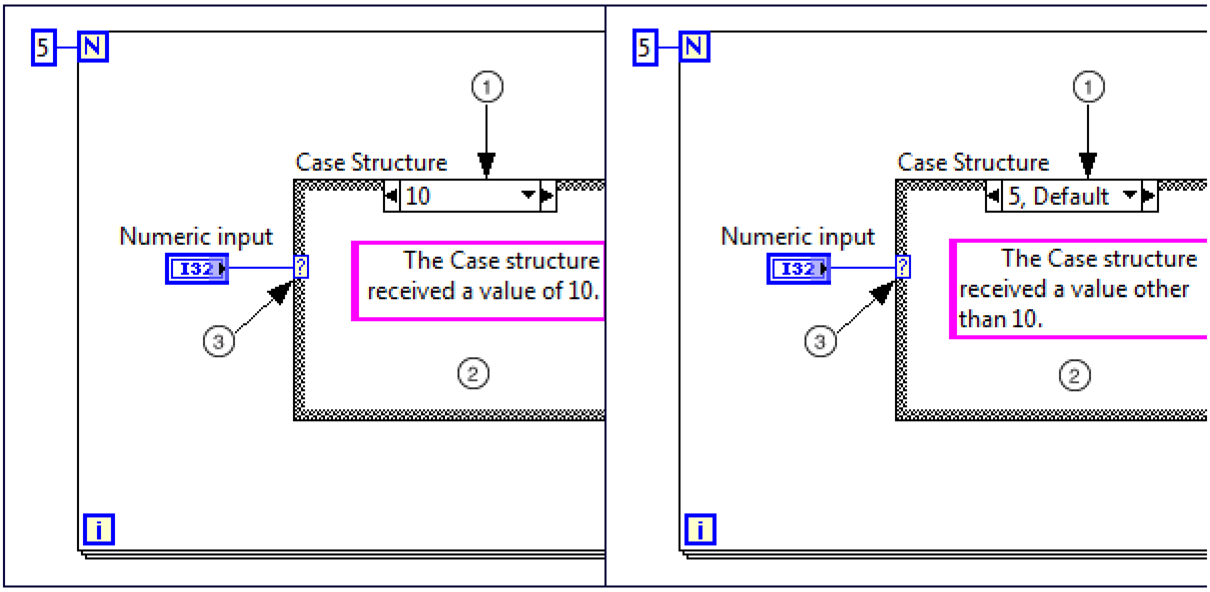


2. CASE STRUCTURES IN LABVIEW

A Case Structure is a branching control mechanism that allows different executions depending on the value of the label. The Case Structure is analogous to the Case block in Java or C++ in which, based on what case value the input variable matched, the case structure will choose the correct cases for execution.

Functions -> Programming -> Structures -> Case Structure

Components of a Case Structure



(**Selector label**—Displays the value(s) for which the associated case executes. You can specify a single value or a range of values. You also can use the selector label to specify a default case.

(**Subdiagram(case)**—Contains the code that executes when the value wired to the case selector matches the value that appears in the selector label. To modify the number or order of subdiagrams, right-click the border of the Case structure and select the appropriate option.

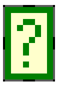
(**Case selector**—Selects which case to execute based on the value of the input data. The input data can be a Boolean, string, integer, enumerated type or error cluster. The data type you wire to the case selector determines the allowed cases you can enter in the selector label.

A Case structure has two or more subdiagrams, or cases.

Only one subdiagram is visible at a time, and the structure executes only one case at a time. An input value determines which subdiagram executes. The Case structure is similar to switch statements or if...then...else statements in text-based programming languages.

The case selector label at the top of the Case structure contains the name of the selector value that corresponds to the case in the center and decrement and increment arrows on each side.

Click the decrement and increment arrows to scroll through the available cases. We also can click the down arrow next to the case name and select a case from the pull-down menu.



Wire an input value, or selector, to the selector terminal to determine which case executes.

We must wire an integer, Boolean value, string, or enumerated type value to the selector terminal. We can position the selector terminal anywhere on the left border of the Case structure. If the data type of the selector terminal is Boolean, the structure has a True case and a False case. If the selector terminal is an integer, string, or enumerated type value, the structure can have any number of cases.

Note: By default, string values we wire to the selector terminal are case sensitive. To allow case-insensitive matches, wire a string value to the selector terminal, right-click the border of the Case structure, and select Case Insensitive Match from the shortcut menu.

If we do not specify a default case for the Case structure to handle out-of-range values, we must explicitly list every possible input value. For example, if the selector is an integer and you specify cases for 1, 2, and 3, we must specify a default case to execute if the input value is 4 or any other unspecified integer value.

Note: We cannot specify a default case if we wire a Boolean control to the selector. If we right-click the case selector label, Make this the Default Case does not appear in the shortcut menu. Make the Boolean control TRUE or FALSE to determine which case to execute.

Right-click the Case structure border to add, duplicate, remove, or rearrange cases, and to select a default case.

Selecting a Case:

Figure below shows a VI that uses a Case structure to execute different code depending on whether a user selects °C or °F for temperature units. The top block diagram shows the True case in the foreground. In the middle block diagram, the False case is selected. To select a case, enter the value in the case selector identifier or use the Labeling tool to edit the values. After we select another case, that case displays on the block diagram, as shown in the bottom block diagram of figure below.

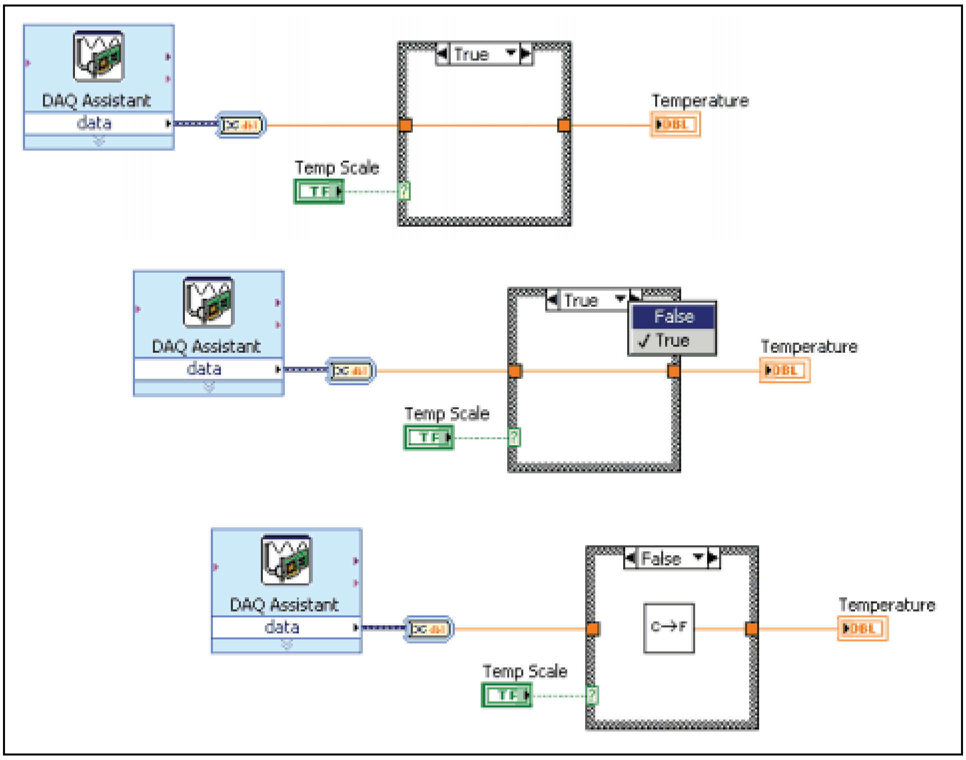


Fig.1 Changing the Case View of a Case Structure

If we enter a selector value that is not the same type as the object wired to the selector terminal, the value appears red. This indicates that the VI will not run until we delete or edit the value. Also, because of the possible round-off error inherent in floating-point arithmetic, we cannot use floating-point numbers as case selector values. If we wire a floating-point value to the case, LabVIEW rounds the value to the nearest integer. If we type a floating-point value in the case selector label, the value appears red to indicate that we must delete or edit the value before the structure can execute.

3.SEQUENCE STRUCTURES IN LABVIEW

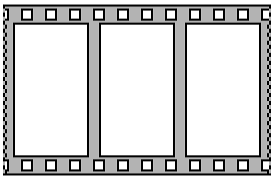
A sequence structure contains one or more sub diagrams, or frames, that execute in sequential order. Within each frame of a sequence structure, as in the rest of the block diagram, data dependency determines the execution order of nodes.

There are two types of sequence structures—the Flat Sequence structure and the Stacked Sequence structure. Use sequence structures sparingly because they hide code. Rely on data flow rather than sequence structures to control the order of execution. With sequence structures, we break the left-to-right data flow paradigm whenever you use a sequence local variable.

The output tunnels of a sequence structure can have only one data source, unlike Case structures. The output can emit from any frame. As with Case structures, data at input tunnels is available to all frames in either the Flat Sequence or the Stacked Sequence structure.

Flat Sequence Structure

The Flat Sequence structure, shown as follows, executes frames from left to right and when all data values wired to a frame are available. The data leaves each frame as the frame finishes executing. This means the input of one frame can depend on the output of another frame.

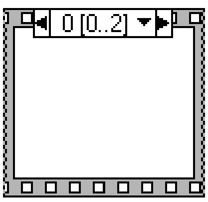


When we add or delete frames in a Flat Sequence structure, the structure resizes automatically.

If we change a Flat Sequence to a Stacked Sequence and then back to a Flat Sequence, LabVIEW moves all input terminals to the first frame of the sequence. The final Flat Sequence operates the same as the Stacked Sequence. After we change the Stacked Sequence to a Flat Sequence with all input terminals on the first frame, we can move wires to where they were located in the original Flat Sequence.

Stacked Sequence Structure

The Stacked Sequence structure, shown as follows, stacks each frame so we see only one frame at a time and executes frame 0, then frame 1, and so on until the last frame executes.



The Stacked Sequence structure returns data only after the last frame executes. Use the Stacked Sequence structure if we want to conserve space on the block diagram.

Unlike when we pass data between frames in the Flat Sequence structure, we need to use sequence locals to pass data from frame to frame in the Stacked Sequence structure.

The sequence selector identifier, shown as follows, at the top of the Stacked Sequence structure contains the current frame number and range of frames.

Use the sequence selector identifier to navigate through the available frames and rearrange frames. The frame label in a Stacked Sequence structure is similar to the case selector label of the Case structure. The frame label contains the frame number in the center and decrement and increment arrows on each side.

We cannot enter values in the frame label. When you add, delete, or rearrange frames in a Stacked Sequence structure, LabVIEW automatically adjusts the numbers in the frame labels.

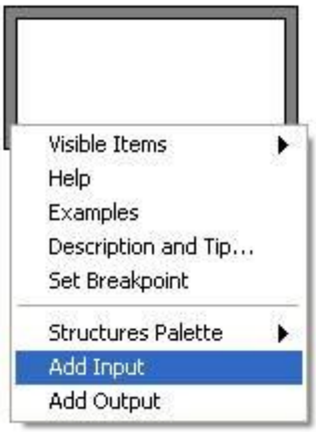
4.FORMULA NODES IN LABVIEW

The Formula Node in the LabVIEW software is a convenient, text-based node we can use to perform complicated mathematical operations on a block diagram using the C++ syntax structure. It is most useful for equations that have many variables or are otherwise complicated. The text-based code simplifies the block diagram and increases its readability. Furthermore, we can copy and paste existing code directly into the Formula Node rather than recreating it graphically.

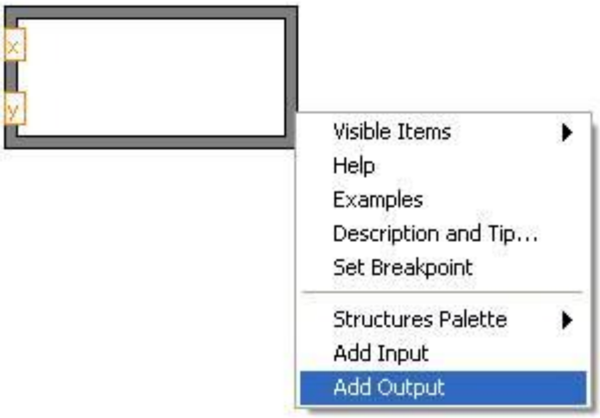
Using the Formula Node

Complete the following steps to create a VI that computes different formulas depending on whether the product of the inputs is positive or negative.

1. Selecting **File**»**New VI** to open a blank VI.
2. Place a Formula Node on the block diagram.
    1. Right-click on the diagram and navigate to **Programming**»**Structures**»**Formula Node**.
    2. Click and drag the cursor to place the Formula Node on the block diagram.
3. Right-click the border of the Formula Node and select **Add Input** from the shortcut menu.



4. Label the input variable 'x'.
5. Repeat steps 3 and 4 to add another input and label it 'y'.
6. Right-click the border of the Formula Node and select **Add Output** from the shortcut menu.



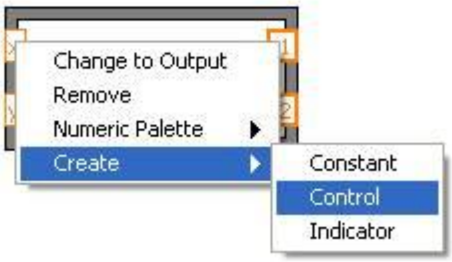
7. Create two outputs and name them z1 and z2, respectively.

Note: It is considered good programming practice to keep the inputs on the left border and the outputs on the right border of the Formula Node. This helps us to follow the data flow in our VI and keep our code organized.

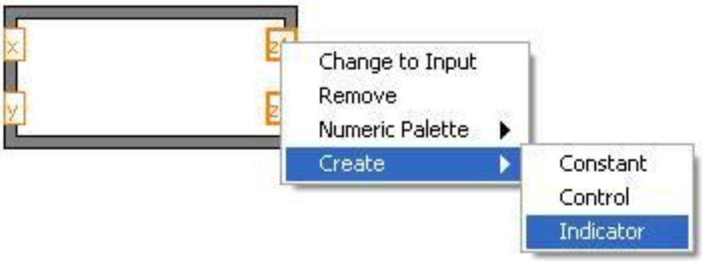
8. Enter the expressions below in the Formula Node. Make sure that we complete each command with a semicolon. Notice, however, that the if statement does not require a semicolon after the first line.

```
if (x*y>0)
z1 = 3*x**2 - 2*y**3;
else z1 = 0;
z2 = sinh(z1);
```

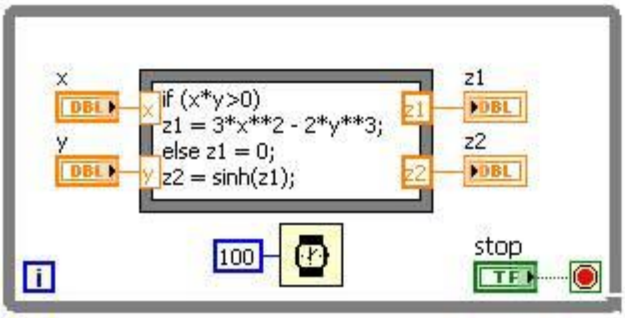
9. Create controls and indicators for the inputs and outputs.
   1. Right-click on each input and select **Create»Control** from the shortcut menu.



2.
   2. Right-click on each output and select **Create»Indicator** from the shortcut menu.

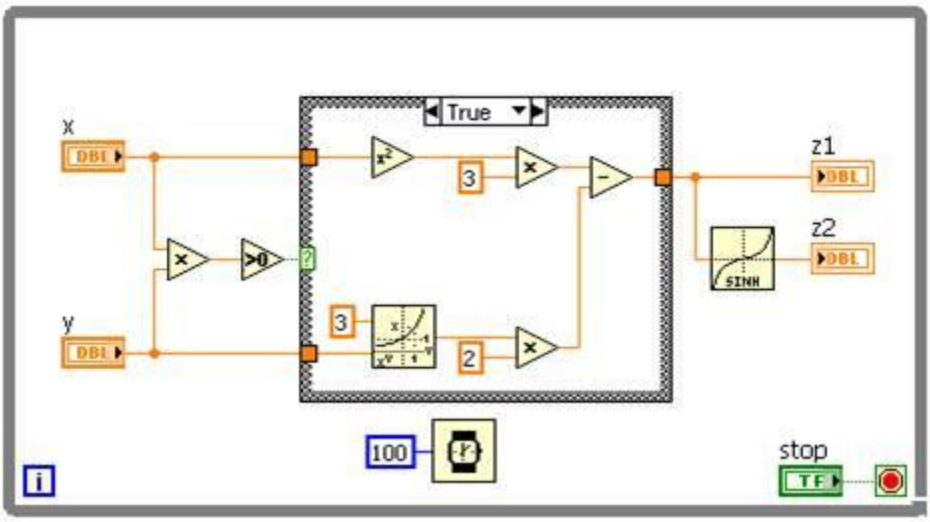


10. Place a While Loop with a stop button around the Formula Node and the controls. Be sure to include a Wait (ms) function inside the loop to conserve memory usage. Your block diagram should appear as follows.



11. Click the **Run** button to run the VI. Change the values of the input controls to see how the outputs change.

In this case, the Formula Node helps minimize the space required on the block diagram. Accomplishing the same task without the use of a Formula Node requires the following code.



5. FILE I/O in LABVIEW

Use the File I/O VIs and functions to open and close files, read from and write to files, create directories and files we specify in the path control, retrieve directory information, and write strings, numbers, arrays, and clusters to files.

Use the VIs and functions on this palette to perform common I/O operations and other types of file I/O operations. we can read or write various types of data, such as characters or lines in text files, numeric values in spreadsheet text files, or data in binary files.

The File I/O palette and the Advanced File palette include functions to control each file I/O operation individually. Use these functions to create or open a file, read data from or write data to the file, and close the file. We also can use the functions to create directories; move, copy, or delete files; list directory contents; change file characteristics; or manipulate paths.

Table:1 Palette Object Description

Palette Object	**Description**
Build Path	Creates a new path by appending a name (or relative path) to an existing path.
Close File	Closes an open file specified by **refnum** and returns the path to the file associated with the refnum.
Format Into File	Formats string, numeric, path, or Boolean data as text and writes the text to a file. If we wire a file refnum to the **file** input, writing begins at the current file position. To append to an existing file, open the file and set the file position to the end of the file by using the Set File Position function. Otherwise, the function will open the file and write to the beginning of the file. This function does not work for files inside an LLB.
Open/Create/Replace File	Opens an existing file, creates a new file, or replaces an existing file, programmatically or interactively using a file dialog box. This function does not work for files inside an LLB.
Read Delimited Spreadsheet	Reads a specified number of lines or rows from a numeric text file beginning at a specified character offset and converts the data to a 2D, double-precision array of numbers, strings, or integers. You must manually select the polymorphic instance you want to use.
Read from Binary File	Reads binary data from a file and returns it in **data**. How the data is read depends on the format of the specified file. This function does not work for files inside an LLB.
Read From Measurement File	Reads data from a text-based measurement file (.lvm) or binary measurement file (.tdm or .tdms).

Read from Text File	Reads a specified number of characters or lines from a byte stream file. This function does not work for files inside an LLB.
Scan From File	Scans text in a file for string, numeric, path, and Boolean data, converts the text to a data type, and returns a duplicated refnum and the converted outputs in the order scanned. This function does not work for files inside an LLB.
Strip Path	Returns the **name** of the last component of a path and the **stripped path** that leads to that component.
Write Delimited Spreadsheet	Converts a 2D or 1D array of strings, signed integers, or double-precision numbers to a text string and writes the string to a new byte stream file or appends the string to an existing file. Wire data to the **2D data** input or **1D data** input to determine the polymorphic instance to use or manually select the instance. Use this VI to transpose or separate data. **Note** To format with Microsoft Excel, use ActiveX with LabVIEW or the Report Generation Toolkit for Microsoft Office.
Write to Binary File	Writes binary data to a new file, appends data to an existing file, or replaces the contents of a file. This function does not work for files inside an LLB.
Write To Measurement File	Writes data to text-based measurement files (.lvm), binary measurement files (.tdm or .tdms), or Microsoft Excel files (.xlsx).
Write to Text File	Writes a string or an array of strings as lines to a file. This function does not work for files inside an LLB.

6. STRING FUNCTIONS IN LABVIEW

Use the String functions to concatenate two or more strings, extract a subset of strings from a string, convert data into strings, and format a string for use in a word processing or spreadsheet application.

Table 2:Palette Object Description for String Function

Palette Object	**Description**
Array To Spreadsheet String	Converts an array of any dimension to a table in string form, containing tabs separating column elements, a platform-dependent EOL character separating rows, and, for arrays of three or more dimensions, headers separating pages.
Build Text	Creates an output string from a combination of text and parameterized inputs. If the input is not a string, this Express VI converts the input into a string based on the configuration of the Express VI.
Carriage Return Constant	Consists of a constant string containing the ASCII CR value.
Concatenate Strings	Concatenates input strings and 1D arrays of strings into a single output string. For array inputs, this function concatenates each element of the array.
Empty String Constant	Consists of a constant string that is empty (length zero).
End of Line Constant	Consists of a constant string containing the platform-dependent end-of-line value.
Format Date/Time String	Displays a timestamp value or a numeric value as time in the format you specify using time format codes.
Format Into String	Formats string, path, enumerated type, time stamp, Boolean, or numeric data as text.
Line Feed Constant	Consists of a constant string containing the ASCII LF value.

Match Pattern	Searches for **regular expression** in **string** beginning at **offset**. If the function finds a match, it splits **string** into three substrings. A regular expression requires a specific combination of characters for pattern matching. This function gives us fewer options for matching strings but performs more quickly than the Match Regular Expression function.
Match Regular Expressi on	Searches for a regular expression in the **input string** beginning at the **offset** we enter. If the function finds a match, it splits the string into three substrings and any number of submatches. Resize the function to view any submatches found in the string.
Normaliz e End Of Line	Converts the line endings of the input string to the line ending format we specify. If we do not specify a line ending format, this VI converts the line endings of the string to the line endings that the current platform expects. Use this VI to make your strings readable by different platforms or by the command line of the current platform.
Replace Substrin g	Inserts, deletes, or replaces a substring at the offset you specify in **string**.
Scan From String	Scans the input string and converts the string according to **format string**.
Search and Replace String	Replaces one or all instances of a substring with another substring. To include the **multiline** input and enable advanced regular expression searches, right-click the function and select **Regular Expression**.
Space Constant	Use this constant to supply a one-character space string to the block diagram.
Spreadsh eet String To Array	Converts the **spreadsheet string** to an **array** of the dimension and representation of **array type**. This function works for arrays of strings and arrays of numbers.
String Constant	Use this constant to supply a constant text string to the block diagram.
String Length	Returns in **length** the number of characters (bytes) in **string**.
String Subset	Returns the **substring** of the input **string** beginning at **offset** and containing **length** number of characters.

Tab Constant	Consists of a constant string containing the ASCII HT (horizontal tab) value.
To Lower Case	Converts all alphabetic characters in **string** to lowercase characters. Evaluates all numbers in **string** as ASCII codes for characters. This function does not affect non-alphabetic characters.
To Upper Case	Converts all alphabetic characters in **string** to uppercase characters. Evaluates all numbers in **string** as ASCII codes for characters. This function does not affect non-alphabetic characters.
Trim Whitespace	Removes all ASCII white space (spaces, tabs, carriage returns, and linefeeds) from the beginning, end, or both ends of **string**. The Trim Whitespace VI does not remove double byte characters.

7. LAB VIEW APPLICATIONS IN ELECTRONIC INSTRUMENTATION

1)Automated Manufacturing test of a component/sub-system/system.

2)Automated Product design validation of a component/sub-system/system.

3)Control and/or monitoring of a machine/piece of industrial equipment/process.

4)Condition monitoring of a machine/piece of industrial equipment.

Here's some applications of LabVIEW:

- Wind turbine monitoring
- Oil and gas pressure monitoring
- Power quality monitoring
- Machine control monitoring
- Commercial and industrial robotics
- Customized control and measurement
- Machine condition monitoring
- Data acquisition
- Signal processing
- Instrumentation control
- Aerospace and transportation systems control and monitoring
- Embedded systems testing
- Device and component prototyping
- Industrial automation
- Weather monitoring and warning systems
- UI design for instrumentation
- Industrial and commercial result/output generation systems