FATIMA MICHAEL COLLEGE OF ENGINEERING & TECHNOLOGY
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An ISO 9001:2008 Certified Institution

EC2351

MEASUREMENTS AND INSTRUMENTATION

Branch: ECE **Year & Semester: III & 6th**

Name of the Staff: G.Sasi, ASP/ECE

Syllabus

EC2351 MEASUREMENTS AND INSTRUMENTATION **L T P C**
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UNIT I BASIC MEASUREMENT CONCEPTS **9**
Measurement systems – Static and dynamic characteristics – units and standards of measurements – error :- accuracy and precision, types, statistical analysis – moving coil, moving iron meters – multimeters – Bridge measurements : – Maxwell, Hay, Schering, Anderson and Wien bridge.
UNIT II BASIC ELECTRONIC MEASUREMENTS **9**
Electronic multimeters – Cathode ray oscilloscopes – block schematic – applications – special oscilloscopes :– delayed time base oscilloscopes, analog and digital storage oscilloscope, sampling oscilloscope – Q meters – Vector meters – RF voltage and power measurements – True RMS meters.
UNIT III SIGNAL GENERATORS AND ANALYZERS **9**
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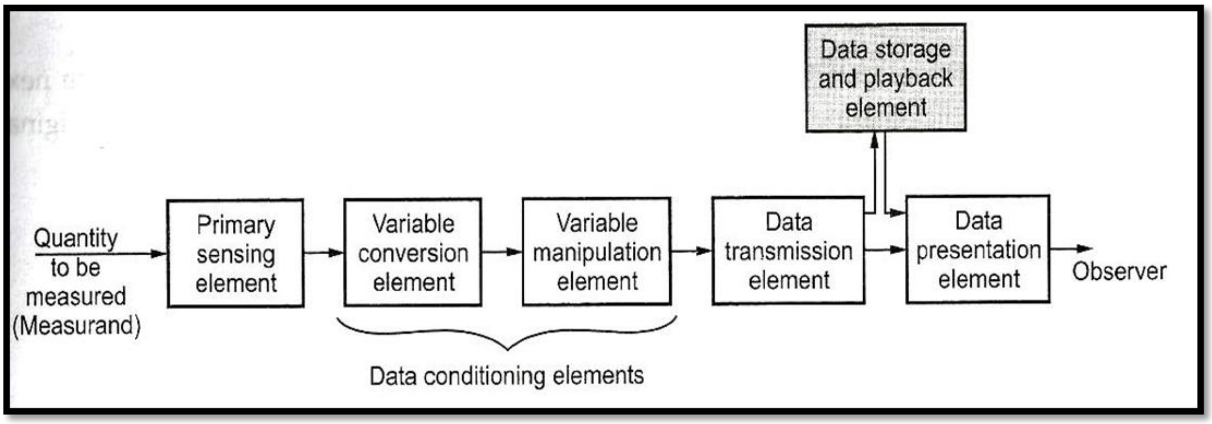
UNIT IV DIGITAL INSTRUMENTS **9**
Comparison of analog and digital techniques – digital voltmeter – multimeters – frequency counters – measurement of frequency and time interval – extension of frequency range – Automation in digital instruments, Automatic polarity indication, automatic ranging, automatic zeroing, fully automatic digital instruments, Computer controlled test systems, Virtual instruments.
UNIT V DATA ACQUISITION SYSTEMS AND FIBER OPTIC MEASUREMENT9
Elements of a digital data acquisition system – interfacing of transducers – multiplexing – data loggers –computer controlled instrumentation – IEEE 488 bus – fiber optic measurements for power and system loss – optical time domains reflectometer.

TOTAL: 45 PERIODS

TEXT BOOKS
1. Albert D.Helfrick and William D.Cooper – Modern Electronic Instrumentation and Measurement Techniques, Pearson / Prentice Hall of India, 2007.
2. Ernest O. Doebelin, Measurement Systems- Application and Design, TMH, 2007.
REFERENCES
1. Joseph J.Carr, Elements of Electronics Instrumentation and Measurement, Pearson Education, 2003.
2. Alan. S. Morris, Principles of Measurements and Instrumentation, 2nd Edition, Prentice Hall of India, 2003.
3. David A. Bell, Electronic Instrumentation and measurements, Prentice Hall of India Pvt Ltd, 2003.
4. B.C. Nakra and K.K. Choudhry, Instrumentation, Meaurement and Analysis, 2nd Edition, TMH, 2004.
5. James W. Dally, William F. Riley, Kenneth G. McConnell, Instrumentation for Engineering Measurements, 2nd Edition, John Wiley, 2003.

UNIT I - BASIC MEASUREMENT CONCEPTS:

Functional elements of Instruments



- Primary sensing element
- Variable conversion element
- Data presentation element

Primary sensing element

The quantity under measurement makes its first contact with primary sensing element of a measurement system here, the primary sensing element transducer. This transducer converts measured into an analogous electrical signal.

Variable conversion element

The output of the primary sensing element is the electrical signal. It may be a voltage a frequency or some other electrical parameter. But this output is not suitable for this system.

For the instrument to perform the desired function, it may be necessary to convert this output to some other suitable form while retaining the original signal. Consider an example, suppose output is an analog signal form and the next of system accepts input signal only in digital form . Therefore we have to use and to digital converter in this system.

Variable manipulation element

The main function of variable manipulation element is to manipulation element is to manipulate the signal presented to it preserving the original nature of the signal. Here, manipulation means a change in numerical value of the signal.

Consider a small example, an electric amplifier circuit accepts a small voltage signal as input and produces an output signal which is also voltage but of greater amplifier. Thus voltage amplifier acts as a variable manipulation element.

Data presentation element

The information about the quantity under measurement has to be conveyed to the personal handling the instrument or system for control or analysis purposes. The information conveyed must be in the form of intelligible to the personnel. The above function is done by data presentation element.

The output or data of the system can be monitored by using visual display devices may be analog or digital device like ammeter, digital meter etc. In case the data to be record, we can use analog or digital recording equipment. In industries , for control and analysis purpose we can use computers.

The final stage in a measurement system is known as terminating stage . when a control device is used for the final measurement stage it is necessary to apply some feedback to the input signal to accomplish the control 0bjective.

The term signal conditioning includes many other functions in addition to variable conversion and variable manipulation. In fact the element that follows the primary sensing element in any instrument or instrumentation system should be called signal conditioning element.

When the element of an instrument is physically separated, it becomes necessary to transmit data from one to another. This element is called transmitting element. The signal conditioning and transmitting stage is generally known as intermediate stage.

Measurement system:

Measurement system any of the systems used in the process of associating numbers with physical quantities and phenomena. Although the concept of weights and measures today includes such factors as temperature, luminosity, pressure, and electric current, it once consisted of only four basic measurements: mass (weight), distance or length, area, and volume (liquid or grain measure). The last three are, of course, closely related. Basic to the whole idea of weights and measures are the concepts of uniformity, units, and standards.Uniformity, the essence of any system of weights and measures, requires accurate, reliable standards of mass and length .

Static Characteristics of Instrument Systems:

Output/Input Relationship

Instrument systems are usually built up from a serial linkage of distinguishable building blocks. The actual physical assembly may not appear to be so but it can be broken down into a representative diagram of connected blocks. In the Humidity sensor it is activated by an input physical parameter and provides an output signal to the next block that processes the signal into a more appropriate state.

A key generic entity is, therefore, the relationship between the input and output of the block. As was pointed out earlier, all signals have a time characteristic, so we must consider the behavior of a block in terms of both the static and dynamic states.

The behavior of the static regime alone and the combined static and dynamic regime can be found through use of an appropriate mathematical model of each block. The mathematical description of system responses is easy to set up and use if the elements all act as linear systems and where addition of signals can be carried out in a linear additive manner. If nonlinearity exists in elements, then it becomes considerably more difficult — perhaps even quite impractical — to provide an easy to follow mathemat- ical explanation. Fortunately, general description of instrument systems responses can be usually be adequately covered using the linear treatment.

The output/input ratio of the whole cascaded chain of blocks 1, 2, 3, etc. is given as:

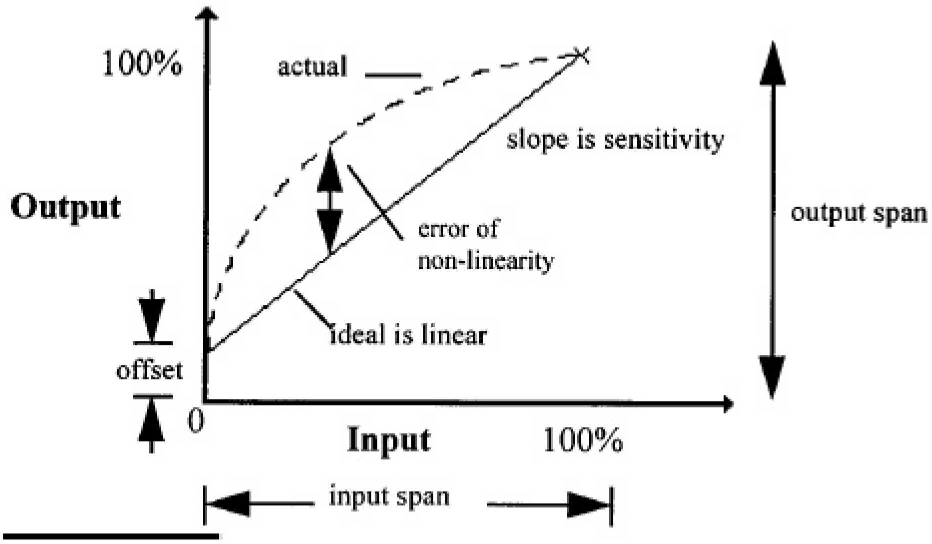
[output/input]total = [output/input]1× [output/input]2× [output/input]3 …

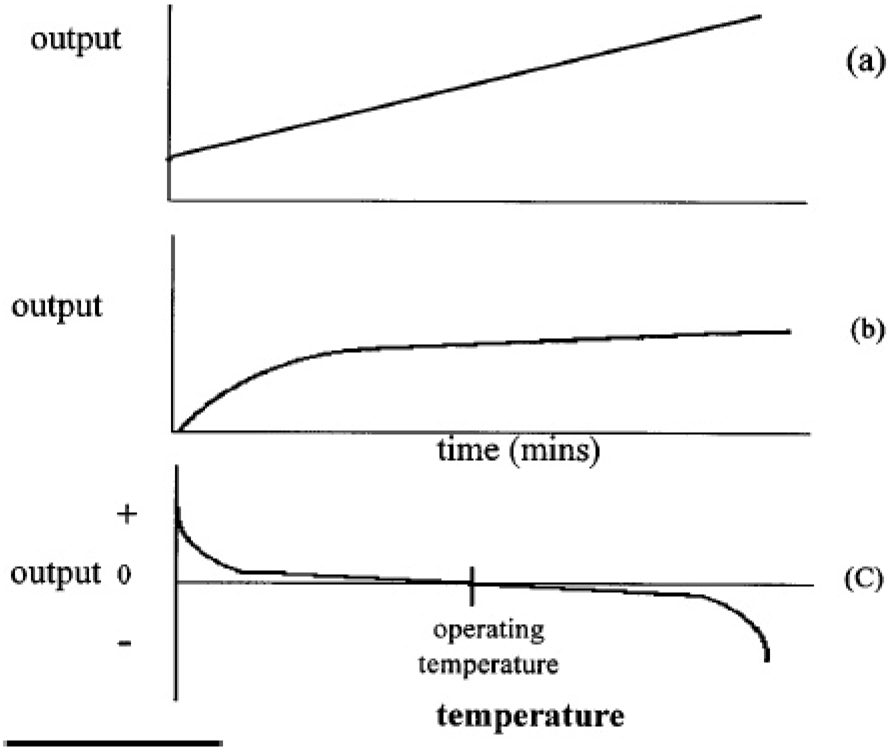
The output/input ratio of a block that includes both the static and dynamic characteristics is called the transfer function and is given the symbol G.

The equation forG can be written as two parts multiplied together. One expresses the static behavior of the block, that is, the value it has after all transient (time varying) effects have settled to their final state. The other part tells us how that value responds when the block is in its dynamic state. The static part is known as the transfer characteristic and is often all that is needed to be known for block description.

The static and dynamic response of the cascade of blocks is simply the multiplication of all individual blocks. As each block has its own part for the static and dynamic behavior, the cascade equations can be rearranged to separate the static from the dynamic parts and then by multiplying the static set and the dynamic set we get the overall response in the static and dynamic states. This is shown by the sequence of Equations.

Instruments are formed from a connection of blocks. Each block can be represented by a conceptual and mathematical model. This example is of one type of humidity sensor.





Drift :

It is now necessary to consider a major problem of instrument performance called instrument drift . This is caused by variations taking place in the parts of the instrumentation over time. Prime sources occur as chemical structural changes and changing mechanical stresses. Drift is a complex phenomenon for which the observed effects are that the sensitivity and offset values vary. It also can alter the accuracy of the instrument differently at the various amplitudes of the signal present.

Detailed description of drift is not at all easy but it is possible to work satisfactorily with simplified values that give the average of a set of observations, this usually being quoted in a conservative manner. The first graph (a) in Figure shows typical steady drift of a measuring spring component of a weighing balance. Figure (b) shows how an electronic amplifier might settle down after being turned on.

Drift is also caused by variations in environmental parameters such as temperature, pressure, and humidity that operate on the components. These are known as influence parameters. An example is the change of the resistance of an electrical resistor, this resistor forming the critical part of an electronic amplifier that sets its gain as its operating temperature changes.

Unfortunately, the observed effects of influence parameter induced drift often are the same as for time varying drift. Appropriate testing of blocks such as electronic amplifiers does allow the two to be separated to some extent. For example, altering only the temperature of the amplifier over a short period will quickly show its temperature dependence.

Drift due to influence parameters is graphed in much the same way as for time drift. Figure shows the drift of an amplifier as temperature varies. Note that it depends significantly on the temperature

Drift in the performance of an instrument takes many forms:

(a) drift over time for a spring balance;

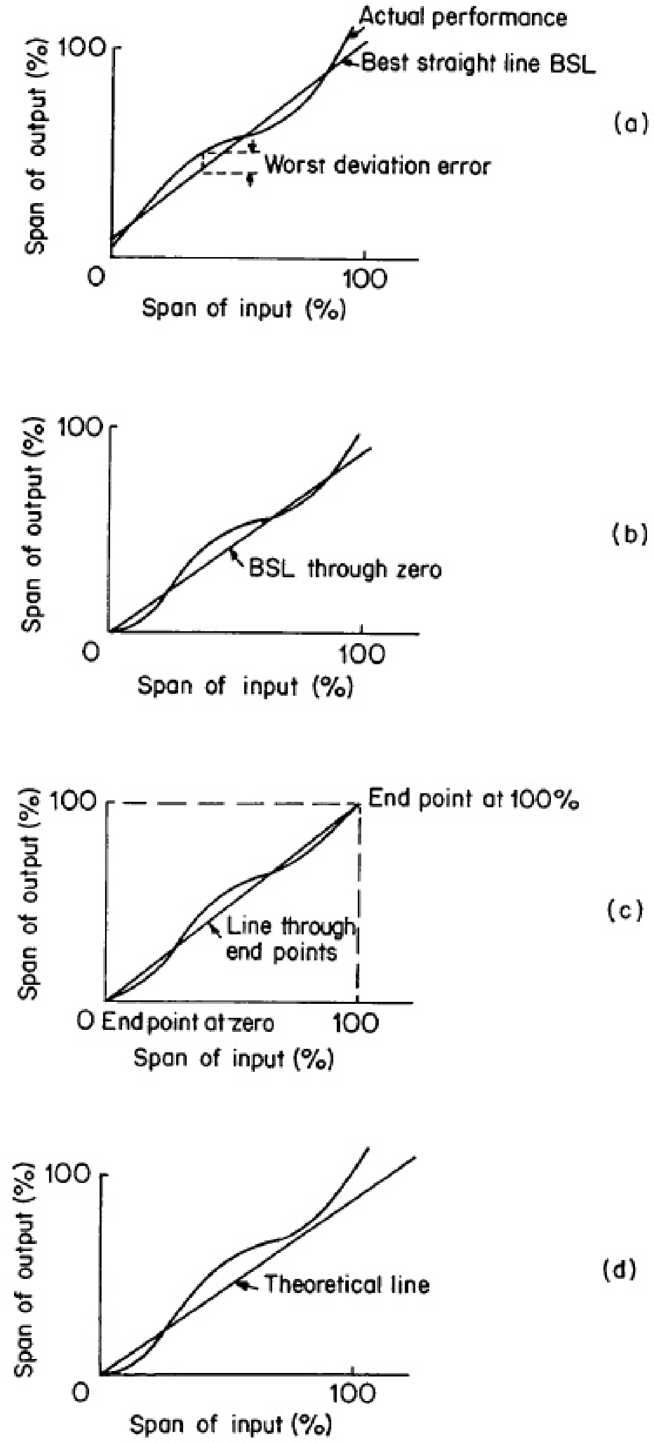
(b) how an electronic amplifier might settle over time to a final value after power is supplied;

(c) drift, due to temperature, of an electronic amplifier varies with the actual temperature of operation.

Dynamic Characteristics of Instrument Systems:

Dealing with Dynamic States:

Measurement outcomes are rarely static over time. They will possess a dynamic component that must be understood for correct interpretation of the results. For example, a trace made on an ink pen chart recorder will be subject to the speed at which the pen can follow the input signal changes.Drift in the performance of an instrument takes many forms: (a) drift over time for a spring



Error of nonlinearity can be expressed in four different ways: (a) best fit line (based on selected method used to decide this); (b) best fit line through zero; (c) line joining 0% and 100% points; and (d) theoretical line

To properly appreciate instrumentation design and its use, it is now necessary to develop insight into the most commonly encountered types of dynamic response and to develop the mathematical modeling basis that allows us to make concise statements about responses.

If the transfer relationship for a block follows linear laws of performance, then a generic mathematical method of dynamic description can be used. Unfortunately, simple mathematical methods have not been found that can describe all types of instrument responses in a simplistic and uniform manner. If the behavior is nonlinear, then description with mathematical models becomes very difficult and might be impracticable. The behavior of nonlinear systems can, however, be studied as segments of linear behavior joined end to end. Here, digital computers are effectively used to model systems of any kind provided the user is prepared to spend time setting up an adequate model.

Now the mathematics used to describe linear dynamic systems can be introduced. This gives valuable insight into the expected behavior of instrumentation, and it is usually found that the response can be approximated as linear.

The modeled response at the output of a blockGresult is obtained by multiplying the mathematical expression for the input signalGinput by the transfer function of the block under investigationGresponse, as shown in below equation

$$G_{result} = G_{input} \times G_{response}$$

To proceed, one needs to understand commonly encountered input functions and the various types of block characteristics. We begin with the former set: the so-called forcing functions.

Forcing Functions

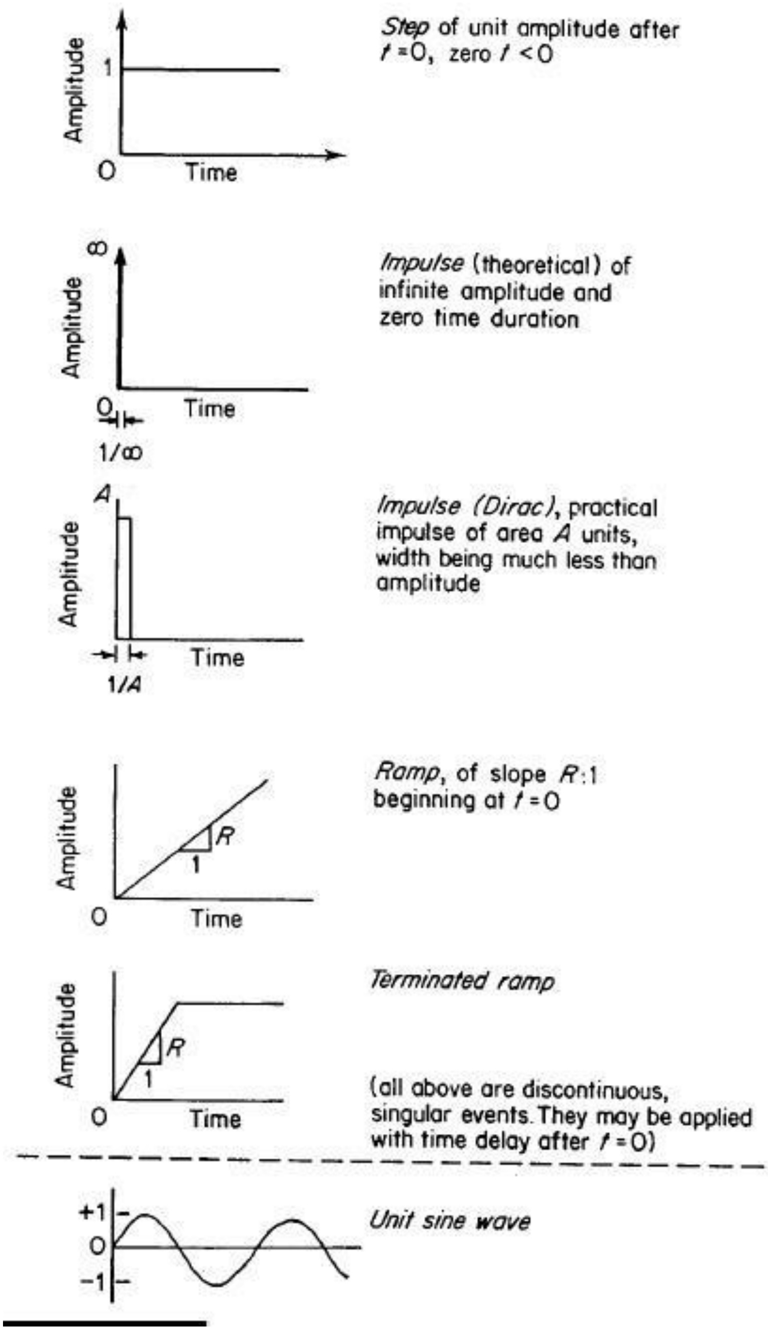
Let us first develop an understanding of the various types of input signal used to perform tests. The most commonly used signals are shown in Figure 3.12. These each possess different valuable test features. For example, the sine-wave is the basis of analysis of all complex wave-shapes because they can be formed as a combination of various sine-waves, each having individual responses that add to give all other wave- shapes. The step function has intuitively obvious uses because input transients of this kind are commonly encountered. The ramp test function is used to present a more realistic input for those systems where it is not possible to obtain instantaneous step input changes, such as attempting to move a large mass by a limited size of force. Forcing functions are also chosen because they can be easily described by a simple mathematical expression, thus making mathematical analysis relatively straightforward.

Characteristic Equation Development

The behavior of a block that exhibits linear behavior is mathematically represented in the general form of expression given as Equation

Here, the coefficientsa2,a1, anda0 are constants dependent on the particular block of interest. The left-hand side of the equation is known as the characteristic equation. It is specific to the internal properties of the block and is not altered by the way the block is used.

The specific combination of forcing function input and block characteristic equation collectively decides the combined output response. Connections around the block, such as feedback from the output to the input, can alter the overall behavior significantly: such systems, however, are not dealt with in this section being in the domain of feedback control systems.



Unit of measurement:

A unit of measurement is a definite magnitude of a physical quantity, defined and adopted by convention and/or by law, that is used as a standard for measurement of the same physical quantity.[1] Any other value of the physical quantity can be expressed as a simple multiple of the unit of measurement.For example, length is a physical quantity. The metre is a unit of length that represents a definite predetermined length. When we say 10 metres (or 10 m), we actually mean 10 times the definite predetermined length called "metre".The definition, agreement, and practical use of units of measurement have played a crucial role in human endeavour from early ages up to this day. Disparate systems of units used to be very common. Now there is a global standard, the International System of Units (SI), the modern form of the metric system.In trade, weights and measures is often a subject of governmental regulation, to ensure fairness and transparency. The Bureau international des poids et mesures (BIPM) is tasked with ensuring worldwide uniformity of measurements and their traceability to the International System of Units (SI). Metrology is the science for developing nationally and internationally accepted units of weights and measures.In physics and metrology, units are standards for measurement of physical quantities that need clear definitions to be useful. Reproducibility of experimental results is central to the scientific method. A standard system of units facilitates this. Scientific systems of units are a refinement of the concept of weights and measures developed long ago for commercial purposes.Science, medicine, and engineering often use larger and smaller units of measurement than those used in everyday life and indicate them more precisely. The judicious selection of the units of measurement can aid researchers in problem solving (see, for example, dimensional analysis).In the social sciences, there are no standard units of measurement and the theory and practice of measurement is studied in psychometrics and the theory of conjoint measurement.

Error Analysis :

Introduction

The knowledge we have of the physical world is obtained by doing experiments and making measurements. It is important to understand how to express such data and how to analyze and draw meaningful conclusions from it.In doing this it is crucial to understand that all measurements of physical quantities are subject to uncertainties. It is never possible to measure anything exactly. It is good, of course, to make the error as small as possible but it is always there. And in order to draw valid conclusions the error must be indicated and dealt with properly.Take the measurement of a person's height as an example. Assuming that her height has been determined to be 5' 8", how accurate is our result?Well, the height of a person depends on how straight she stands, whether she just got up (most people are slightly taller when getting up from a long rest in horizontal position), whether she has her shoes on, and how long her hair is and how it is made up. These inaccuracies could all be called errors of definition. A quantity such as height is not exactly defined without specifying many other circumstances.Even if you could precisely specify the "circumstances," your result would still have an error associated with it. The scale you are using is of limited accuracy; when you read the scale, you may have to estimate a fraction between the marks on the scale, etc.

If the result of a measurement is to have meaning it cannot consist of the measured value alone. An indication of how accurate the result is must be included also. Indeed, typically more effort is required to determine the error or uncertainty in a measurement than to perform the measurement itself. Thus, the result of any physical measurement has two essential components: (1) A numerical value (in a specified system of units) giving the best estimate possible of the quantity measured, and (2) the degree of uncertainty associated with this estimated value. For example, a measurement of the width of a table would yield a result such as 95.3 +/- 0.1 cm.

Significant Figures :

The significant figures of a (measured or calculated) quantity are the meaningful digits in it. There are conventions which you should learn and follow for how to express numbers so as to properly indicate their significant figures.

- Any digit that is not zero is significant. Thus 549 has three significant figures and 1.892 has four significant figures.
- Zeros between non zero digits are significant. Thus 4023 has four significant figures.
- Zeros to the left of the first non zero digit are not significant. Thus 0.000034 has only two significant figures. This is more easily seen if it is written as 3.4x10-5.
- For numbers with decimal points, zeros to the right of a non zero digit are significant. Thus 2.00 has three significant figures and 0.050 has two significant figures. For this reason it is important to keep the trailing zeros to indicate the actual number of significant figures.
- For numbers without decimal points, trailing zeros may or may not be significant. Thus, 400 indicates only one significant figure. To indicate that the trailing zeros are significant a decimal point must be added. For example, 400. has three significant figures, and 4×10^2 has one significant figure.
- Exact numbers have an infinite number of significant digits. For example, if there are two oranges on a table, then the number of oranges is 2.000... . Defined numbers

are also like this. For example, the number of centimeters per inch (2.54) has an infinite number of significant digits, as does the speed of light (299792458 m/s).

There are also specific rules for how to consistently express the uncertainty associated with a number. In general, the last significant figure in any result should be of the same order of magnitude (i.e.. in the same decimal position) as the uncertainty. Also, the uncertainty should be rounded to one or two significant figures. Always work out the uncertainty after finding the number of significant figures for the actual measurement.

For example,

9.82 +/- 0.02
10.0 +/- 1.5
4 +/- 1

The following numbers are all incorrect.

9.82 +/- 0.02385 is wrong but 9.82 +/- 0.02 is fine
10.0 +/- 2 is wrong but 10.0 +/- 2.0 is fine
4 +/- 0.5 is wrong but 4.0 +/- 0.5 is fine

In practice, when doing mathematical calculations, it is a good idea to keep one more digit than is significant to reduce rounding errors. But in the end, the answer must be expressed with only the proper number of significant figures. After addition or subtraction, the result is significant only to the place determined by the largest last significant place in the original numbers. For example,

89.332 + 1.1 = 90.432
should be rounded to get 90.4 (the tenths place is the last significant place in 1.1). After multiplication or division, the number of significant figures in the result is determined by the original number with the smallest number of significant figures. For example,

(2.80) (4.5039) = 12.61092

should be rounded off to 12.6 (three significant figures like 2.80).

Refer to any good introductory chemistry textbook for an explanation of the methodology for working out significant figures.

The Idea of Error :

The concept of error needs to be well understood. What is and what is not meant by "error"? A measurement may be made of a quantity which has an accepted value which can be looked up in a handbook (e.g.. the density of brass). The difference between the measurement and the accepted value is not what is meant by error. Such accepted values are not "right" answers. They are just measurements made by other people which have errors associated with them as well. Nor does error mean "blunder." Reading a scale backwards, misunderstanding what you are doing or elbowing your lab partner's measuring apparatus are blunders which can be caught and should simply be disregarded. Obviously, it cannot be determined exactly how far off a measurement is; if this could be done, it would be possible to just give a more accurate, corrected value. Error, then, has to do with uncertainty in measurements that nothing can be done about. If a measurement is repeated, the values obtained will differ and none of the results can be preferred over the others. Although it is not possible to do anything about such error, it can be characterized. For instance, the repeated measurements may cluster tightly together or they may spread widely. This pattern can be analyzed systematically.

Classification of Error :

Generally, errors can be divided into two broad and rough but useful classes: systematic and random.Systematic errors are errors which tend to shift all measurements in a systematic way so their mean value is displaced. This may be due to such things as incorrect calibration of equipment, consistently improper use of equipment or failure to properly account for some effect. In a sense, a systematic error is rather like a blunder and large systematic errors can and must be eliminated in a good experiment. But small systematic errors will always be present. For instance, no instrument can ever be calibrated perfectly.Other sources of systematic errors are external effects which can change the results of the experiment, but for which the corrections are not well known. In science, the reasons why several independent confirmations of experimental results are often required (especially using different techniques) is because different apparatus at different places may be affected by different systematic effects. Aside from making mistakes (such as thinking one is using the x10 scale, and actually using the x100 scale), the reason why experiments sometimes yield results which may be far outside the quoted errors is because of systematic effects which were not accounted for.

Random errors are errors which fluctuate from one measurement to the next. They yield results distributed about some mean value. They can occur for a variety of reasons.

- They may occur due to lack of sensitivity. For a sufficiently a small change an instrument may not be able to respond to it or to indicate it or the observer may not be able to discern it.
- They may occur due to noise. There may be extraneous disturbances which cannot be taken into account.
- They may be due to imprecise definition.
- They may also occur due to statistical processes such as the roll of dice.

Random errors displace measurements in an arbitrary direction whereas systematic errors displace measurements in a single direction. Some systematic error can be substantially eliminated (or properly taken into account). Random errors are unavoidable and must be lived with. Many times you will find results quoted with two errors. The first error quoted is usually the random error, and the second is called the systematic error. If only one error is quoted, then the errors from all sources are added together. (In quadrature as described in the section on propagation of errors.)A good example of "random error" is the statistical error associated with sampling or counting. For example, consider radioactive decay which occurs randomly at a some (average) rate. If a sample has, on average, 1000 radioactive decays per second then the expected number of decays in 5 seconds would be 5000. A particular measurement in a 5 second interval will, of course, vary from this average but it will generally yield a value within 5000 +/- . Behavior like this, where the error,

$$\Delta n = \sqrt{n_{expected}} \text{, (1)}$$

$$n_{measured} = n_{expected}$$

is called a Poisson statistical process. Typically if one does not know $n_{expected}$ it is assumed that,, in order to estimate this error.

A. Mean Value

Suppose an experiment were repeated many, say N, times to get,

$$x_1, x_2, \ldots x_k, \ldots x_N \text{,}$$

N measurements of the same quantity, x. If the errors were random then the errors in these results would differ in sign and magnitude. So if the average or mean value of our measurements were calculated,

$$\bar{x} = \frac{x_1 + x_2 + \ldots + x_k + \ldots + x_N}{N} = \frac{\sum_{k=1}^{N} x_k}{N}, \quad (2)$$

some of the random variations could be expected to cancel out with others in the sum. This is the best that can be done to deal with random errors: repeat the measurement many times, varying as many "irrelevant" parameters as possible and use the average as the best

$N \to \infty$

estimate of the true value of x. (It should be pointed out that this estimate for a given N will differ from the limit as the true mean value; though, of course, for larger N it will be closer to the limit.) In the case of the previous example: measure the height at different times of day, using different scales, different helpers to read the scale, etc.Doing this should give a result with less error than any of the individual measurements. But it is obviously expensive, time consuming and tedious. So, eventually one must compromise and decide that the job is done. Nevertheless, repeating the experiment is the only way to gain confidence in and knowledge of its accuracy. In the process an estimate of the deviation of the measurements from the mean value can be obtained.

B. Measuring Error

There are several different ways the distribution of the measured values of a repeated experiment such as discussed above can be specified.

- Maximum Error

The maximum and minimum values of the data set, x_{max} and x_{min}, could be specified. In these terms, the quantity,

$$\Delta x_{max} = \frac{x_{max} - x_{min}}{2}, \quad (3)$$

is the maximum error. And virtually no measurements should ever fall outside $\bar{x} \pm \Delta x_{max}$.

- Probable Error

The probable error, Δx_{prob}, specifies the range $\bar{x} \pm \Delta x_{prob}$ which contains 50% of the measured values.

- Average Deviation

The average deviation is the average of the deviations from the mean,

$$\Delta x_{av} = \frac{\sum_k |x_k - \bar{x}|}{N}. \quad (4)$$

For a Gaussian distribution of the data, about 58% will lie within

$\bar{x} \pm \Delta x_{av}$.

- Standard Deviation

For the data to have a Gaussian distribution means that the probability of obtaining the result x is,

$$P(x) = \frac{1}{\sigma\sqrt{2\pi}} e^{\frac{-(x-x_0)^2}{2\sigma^2}}, \quad (5)$$

where x_0 is most probable value and σ, which is called the standard deviation, determines the width of the distribution. Because of the law of large numbers this assumption will tend to be valid for random errors. And so it is common practice to quote error in terms of the standard deviation of a Gaussian distribution fit to the observed data distribution. This is the way you should quote error in your reports.

It is just as wrong to indicate an error which is too large as one which is too small. In the measurement of the height of a person, we would reasonably expect the error to be +/-1/4" if a careful job was done, and maybe +/-3/4" if we did a hurried sample measurement. Certainly saying that a person's height is 5' 8.250"+/-0.002" is ridiculous (a single jump will compress your spine more than this) but saying that a person's height is 5' 8"+/- 6" implies that we have, at best, made a very rough estimate!

C. Standard Deviation

The mean is the most probable value of a Gaussian distribution. In terms of the mean, the standard deviation of any distribution is,

$$\sigma = \sqrt{\frac{\sum_k (x_k - \bar{x})^2}{N}}. \quad (6)$$

The quantity σ^2, the square of the standard deviation, is called the variance. The best estimate of the true standard deviation is,

$$\sigma_x = \sqrt{\frac{\sum_k (x_k - \bar{x})^2}{N-1}} \quad . (7)$$

The reason why we divide by N to get the best estimate of the mean and only by N-1 for the best estimate of the standard deviation needs to be explained. The true mean value of x is not being used to calculate the variance, but only the average of the measurements as the best estimate of it. Thus, $(x_k - \bar{x})^2$ as calculated is always a little bit smaller than $(x_k - \bar{x}_{true})^2$, the quantity really wanted. In the theory of probability (that is, using the assumption that the data has a Gaussian distribution), it can be shown that this underestimate is corrected by using N-1 instead of N.If one made one more measurement of x then (this is also a property of a Gaussian distribution) it would have some 68% probability of lying within $\bar{x} \pm \sigma_x$. Note that this means that about 30% of all experiments will disagree with the accepted value by more than one standard deviation.However, we are also interested in the error of the mean, which is smaller than sx if there were several measurements. An exact calculation yields,

$$\sigma_{\bar{x}} = \frac{\sigma_x}{\sqrt{N}} = \sqrt{\frac{\sum_k (x_k - \bar{x})^2}{N(N-1)}} \quad , (8)$$

for the standard error of the mean. This means that, for example, if there were 20 measurements, the error on the mean itself would be = 4.47 times smaller then the error of each measurement. The number to report for this series of N measurements of x is $\bar{x} \pm \sigma_{\bar{x}}$ where $\sigma_{\bar{x}} = \frac{\sigma_x}{\sqrt{N}}$. The meaning of this is that if the N measurements of x were repeated there would be a 68% probability the new mean value of would lie within $\bar{x} \pm \sigma_{\bar{x}}$(that is between $\bar{x} + \sigma_{\bar{x}}$and $\bar{x} - \sigma_{\bar{x}}$). Note that this also means that there is a 32% probability that it will fall outside of this range. This means that out of 100 experiments of this type, on the average, 32 experiments will obtain a value which is outside the standard errors.

Examples :

Suppose the number of cosmic ray particles passing through some detecting device every hour is measured nine times and the results are those in the following table. Thus we have = 900/9 = 100 and σ_x^2 = 1500/8 = 188 or σ_x= 14. Then the probability that one more measurement of x will lie within 100 +/- 14 is 68%.The value to be reported for this series of measurements is 100+/-(14/3) or 100 +/- 5. If one were to make another series of nine measurements of x there would be a 68% probability the new mean would lie within the range 100 +/- 5.Random counting processes like this example obey a Poisson distribution for which $\sigma_x = \sqrt{\bar{x}}$. So one would expect the value of $\sigma_x$to be 10. This is somewhat less than the value of 14 obtained above; indicating either the process is not quite random or, what is more likely, more measurements are needed.

i	x_i	$(x_i - x_m)^2$
1	80	400
2	95	25
3	100	0
4	110	100
5	90	100
6	115	225
7	85	225
8	120	400
9	105	25
S	900	1500

The same error analysis can be used for any set of repeated measurements whether they arise from random processes or not. For example in the Atwood's machine experiment to measure g you are asked to measure time five times for a given distance of fall s. The mean value of the time is,

$$\bar{t} = \frac{\Sigma t_i}{5} = \frac{t_1 + t_2 + t_3 + t_4 + t_5}{5}, \quad (9)$$

and the standard error of the mean is,

$$\sigma\bar{t} = \sqrt{\frac{\sum(t_i - \bar{t})^2}{n(n-1)}}, (10)$$

where n = 5.

For the distance measurement you will have to estimate [[Delta]]s, the precision with which you can measure the drop distance (probably of the order of 2-3 mm).

Propagation of Errors :

Frequently, the result of an experiment will not be measured directly. Rather, it will be calculated from several measured physical quantities (each of which has a mean value and an error). What is the resulting error in the final result of such an experiment?

For instance, what is the error in Z = A + B where A and B are two measured quantities with errors ΔA ΔB and respectively?

A first thought might be that the error in Z would be just the sum of the errors in A and B. After all,

$$(A + \Delta A) + (B + \Delta B) = (A + B) + (\Delta A + \Delta B) (11)$$

and

$$(A - \Delta A) + (B - \Delta B) = (A + B) - (\Delta A + \Delta B). (12)$$

But this assumes that, when combined, the errors in A and B have the same sign and maximum magnitude; that is that they always combine in the worst possible way. This could only happen if the errors in the two variables were perfectly correlated, (i.e.. if the two variables were not really independent).If the variables are independent then sometimes the error in one variable will happen to cancel out some of the error in the other and so, on the average, the error in Z will be less than the sum of the errors in its parts. A reasonable way to try to take this into account is to treat the perturbations in Z produced by perturbations in its parts as if they were "perpendicular" and added according to the Pythagorean theorem,

$$\Delta Z = \sqrt{(\Delta A)^2 + (\Delta B)^2}. (13)$$

That is, if A = (100 +/- 3) and B = (6 +/- 4) then Z = (106 +/- 5) since $5 = \sqrt{3^2 + 4^2}$.

This idea can be used to derive a general rule. Suppose there are two measurements, A and B, and the final result is Z = F(A, B) for some function F. If A is perturbed by ΔA then Z will be perturbed by

$$\left(\frac{\partial F}{\partial A}\right)\Delta A$$

where (the partial derivative) [[partialdiff]]F/[[partialdiff]]A is the derivative of F with respect to A with B held constant. Similarly the perturbation in Z due to a perturbation in B is,

$$\left(\frac{\partial F}{\partial B}\right)\Delta B .$$

Combining these by the Pythagorean theorem yields

$$\Delta Z = \sqrt{\left(\frac{\partial F}{\partial A}\right)^2 (\Delta A)^2 + \left(\frac{\partial F}{\partial B}\right)^2 (\Delta B)^2} , \quad (14)$$

In the example of Z = A + B considered above,

$$\frac{\partial F}{\partial A} = 1 \text{ and } \frac{\partial F}{\partial B} = 1 ,$$

so this gives the same result as before. Similarly if Z = A - B then,

$$\frac{\partial F}{\partial A} = 1 \text{ and } \frac{\partial F}{\partial B} = -1 ,$$

which also gives the same result. Errors combine in the same way for both addition and subtraction. However, if Z = AB then,

$$\frac{\partial F}{\partial A} = B \text{ and } \frac{\partial F}{\partial B} = A ,$$

so

$$\Delta Z = \sqrt{B^2(\Delta A)^2 + A^2(\Delta B)^2} , \quad (15)$$

Thus

$$\frac{\Delta Z}{z} = \frac{\Delta Z}{AB} = \sqrt{\left(\frac{\Delta A}{A}\right)^2 + \left(\frac{\Delta B}{B}\right)^2} , \quad (16)$$

or the fractional error in Z is the square root of the sum of the squares of the fractional errors in its parts. (You should be able to verify that the result is the same for division as it is for multiplication.) For example,

$$(100 \pm 0.3)(6 \pm 0.4) = 600 \pm 600\sqrt{\left(\frac{0.3}{100}\right)^2 + \left(\frac{0.4}{6}\right)^2} = 600 \pm 40 .$$

It should be noted that since the above applies only when the two measured quantities are independent of each other it does not apply when, for example, one physical quantity is measured and what is required is its square. If Z = A2 then the perturbation in Z due to a perturbation in A is,

$$Z = \frac{\partial F}{\partial A}\Delta A = 2A\,\Delta A \quad . (17)$$

Thus, in this case,

$$(A \pm \Delta A)^2 = A^2 \pm 2A\,\Delta A = A^2\left(1 \pm 2\frac{\Delta A}{A}\right) \quad (18)$$

and not A2 (1 +/- ΔA/A) as would be obtained by misapplying the rule for independent variables. For example,

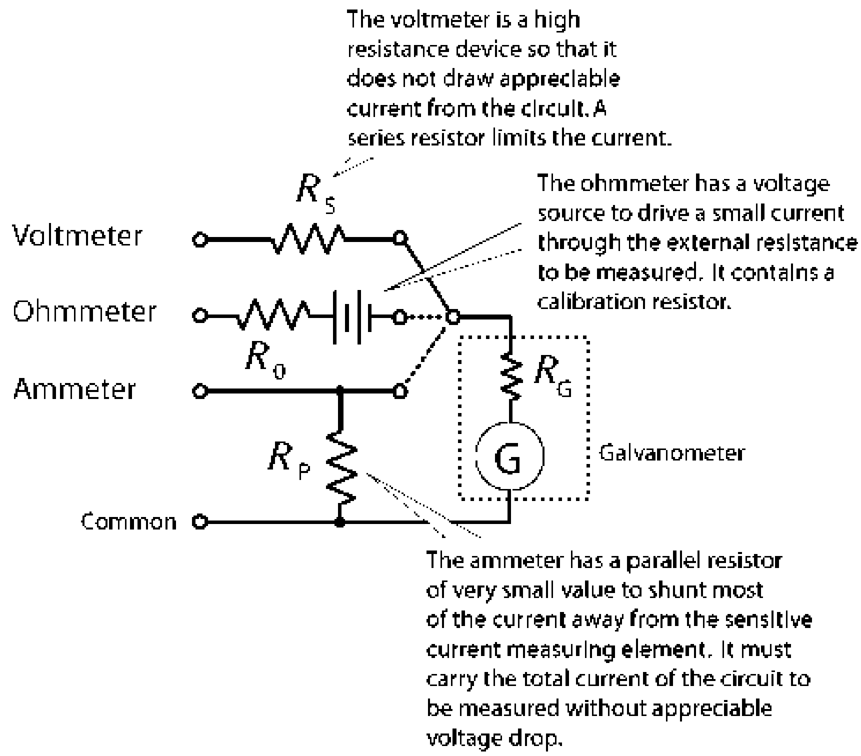
(10 +/- 1)2 = 100 +/- 20 and not 100 +/- 14.

If a variable Z depends on (one or) two variables (A and B) which have independent errors (ΔA and ΔB) then the rule for calculating the error in Z is tabulated in following table for a variety of simple relationships. These rules may be compounded for more complicated situations.

	Relation between Z and(A,B)	Relation between errors ΔZ and (ΔA, ΔB)
1	Z = A + B	$(\Delta Z)^2 = (\Delta A)^2 + (\Delta B)^2$
2	Z = A - B	$(\Delta Z)^2 = (\Delta A)^2 + (\Delta B)^2$
3	Z = AB	$\left(\frac{\Delta Z}{Z}\right)^2 = \left(\frac{\Delta A}{A}\right)^2 + \left(\frac{\Delta B}{B}\right)^2$
4	Z = A/B	$\left(\frac{\Delta Z}{Z}\right)^2 = \left(\frac{\Delta A}{A}\right)^2 + \left(\frac{\Delta B}{B}\right)^2$
5	Z = An	$\frac{\Delta Z}{Z} = n\frac{\Delta A}{A}$
6	Z = ln A	$\Delta Z = \frac{\Delta A}{A}$
7	Z = eA	$\frac{\Delta Z}{Z} = \Delta A$

Voltmeter:

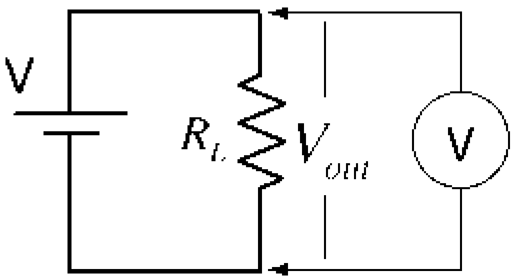
The design of a voltmeter, ammeter or ohmmeter begins with a current-sensitive element. Though most modern meters have solid state digital readouts, the physics is more readily demonstrated with a moving coil current detector called a galvanometer. Since the modifications of the current sensor are compact, it is practical to have all three functions in a single instrument with multiple ranges of sensitivity. Schematically, a single range "multimeter" might be designed as illustrated.



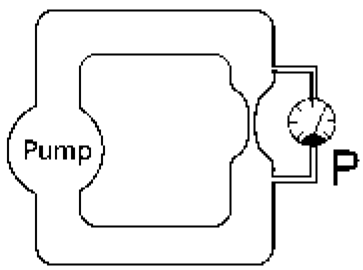
A voltmeter measures the change in voltage between two points in an electric circuit and therefore must be connected in parallel with the portion of the circuit on which the measurement is made. By contrast, an ammeter must be connected in series. In analogy with a water circuit, a voltmeter is like a meter designed to measure pressure difference. It is necessary for the voltmeter to have a very high resistance so that it does not have an appreciable affect on the current or voltage associated with the measured circuit. Modern solid-state meters have digital readouts, but the principles of operation can be better appreciated by examining the older moving coil meters based on galvanometer sensors.

Ammeter:

An ammeter is an instrument for measuring the electric current in amperes in a branch of an electric circuit. It must be placed in series with the measured branch, and must have very low resistance to avoid significant alteration of the current it is to measure. By contrast, an voltmeter must be connected in parallel. The analogy with an in-line flowmeter in a water circuit can help visualize why an ammeter must have a low resistance, and why connecting an ammeter in parallel can damage the meter. Modern solid-state meters have digital readouts, but the principles of operation can be better appreciated by examining the older moving coil metersbased on galvanometer sensors.

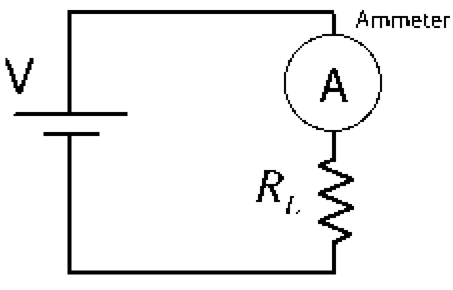


A voltmeter is connected in parallel to measure the voltage change across a circuit element.

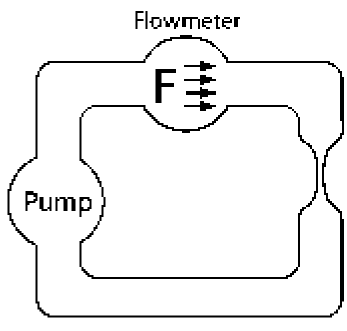


A pressure gauge is connected in parallel to measure the pressure drop across a region of resistance to flow.

A voltmeter is always connected in parallel with the part of the circuit for which you wish to measure voltage.



An ammeter is connected in series with a resistor to measure the current through the resistor.

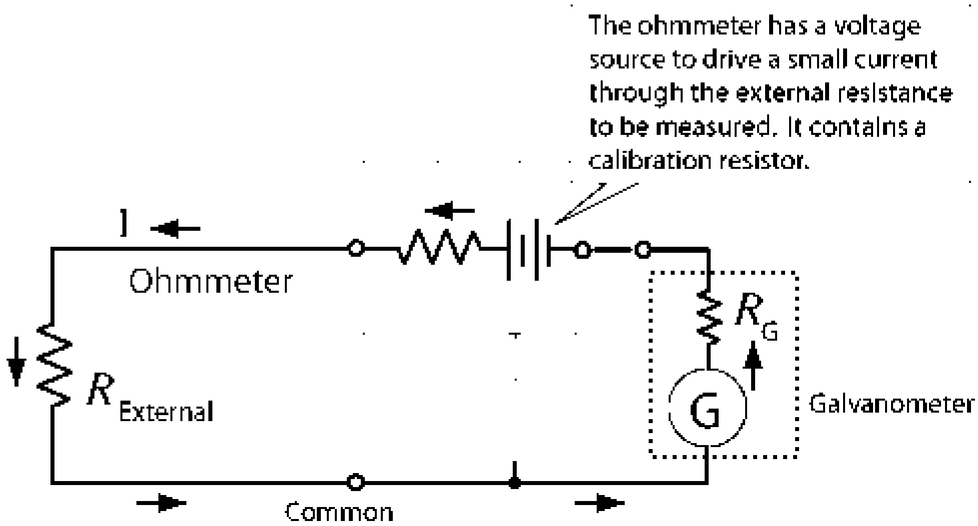


A meter for volume flowrate must be in series to measure the flow, but must not appreciably affect the flow.

An ammeter is always connected in series with the part of the circuit in which you wish to measure current.

Ohmmeter :

The standard way to measure resistance in ohms is to supply a constant voltage to the resistance and measure the current through it. That current is of course inversely proportional to the resistance according to Ohm's law,so that you have a non-linear scale. The current registered by the current sensing element is proportional to 1/R, so that a large current implies a small resistance. Modern solid-state meters have digital readouts, but the principles of operation can be better appreciated by examining the older moving coil meters based on galvanometer sensors.



RMS stands for Root Mean Square:

RMS, or Root Mean Square, is the measurement used for any time varying signal's effective value: It is not an "Average" voltage and its mathematical relationship to peak voltage varies depending on the type of waveform. By definition, RMS Value, also called the effective or heating value of AC, is equivalent to a DC voltage that would provide the same amount of heat generation in a resistor as the AC voltage would if applied to that same resistor. Since an AC signal's voltage rises and falls with time, it takes more AC voltage to produce a given RMS voltage. In other words the grid must produce about 169 volts peak AC which turns out to be 120 volts RMS (.707 x 169). The heating value of the voltage available is equivalent to a 120 volt DC source (this is for example only and does not mean DC and AC are interchangeable). The typical multi-meter is not a True RMS reading meter. As a result it will only produce misleading voltage readings when trying to measure anything other than a DC signal or sine wave. Several types of multi-meters exist, and the owner's manual or the manufacturer should tell you which type you have. Each handles AC signals differently, here are the three basic types. A rectifier type multi-meter indicates RMS values for sine waves only. It does this by measuring average voltage and multiplying by 1.11 to find RMS. Trying to use this type of meter with any waveform other than a sine wave will result in erroneous RMS readings. Average reading digital volt meters are just that, they measure average voltage for an AC signal. Using the equations in the next column for a sine wave, average voltage (V_{avg}) can be converted to Volts RMS (V_{rms}), and doing this allows the meter to display an RMS reading for a sinewave.A True RMS meter uses a complex RMS converter to read RMS for any type of AC waveform.

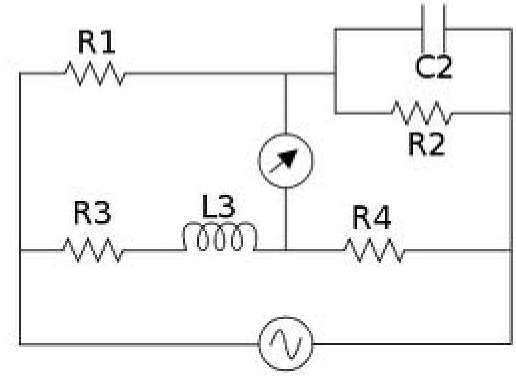
Bridge Measurements:

A Maxwell bridge (in long form, a Maxwell-Wien bridge) is a type of Wheatstone bridge used to measure an unknown inductance (usually of low Q value) in terms of calibrated resistance and capacitance. It is a real product bridge. With reference to the picture, in a typical application R1 and R4 are known fixed entities, and R2 and C2 are known variable entities. R2 and C2 are adjusted until the bridge is balanced.R3 and L3 can then be calculated based on the values of the other components:

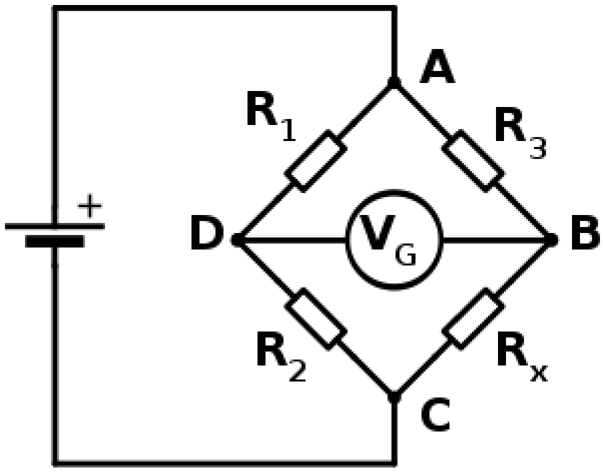
$$R_3 = \frac{R_1 \cdot R_4}{R_2}$$

$$L_3 = R_1 \cdot R_4 \cdot C_2$$

To avoid the difficulties associated with determining the precise value of a variable capacitance, sometimes a fixed-value capacitor will be installed and more than one resistor will be made variable. The additional complexity of using a Maxwell bridge over simpler bridge types is warranted in circumstances where either the mutual inductance between the load and the known bridge entities, or stray electromagnetic interference, distorts the measurement results. The capacitive reactance in the bridge will exactly oppose the inductive reactance of the load when the bridge is balanced, allowing the load's resistance and reactance to be reliably determined.



Wheatstone bridge



It is used to measure an unknown electrical resistance by balancing two legs of a bridge circuit, one leg of which includes the unknown component. Its operation is similar to the original potentiometer.

Operation :

Rx is the unknown resistance to be measured; R1, R2 and R3 are resistors of known resistance and the resistance of R2 is adjustable. If the ratio of the two resistances in the known leg (R2 / R1) is equal to the ratio of the two in the unknown leg (Rx / R3), then the voltage between the two midpoints (B and D) will be zero and no current will flow through the galvanometer Vg. R2 is varied until this condition is reached. The direction of the current indicates whether R2 is too high or too low.

Detecting zero current can be done to extremely high accuracy (see galvanometer). Therefore, if R1, R2 and R3 are known to high precision, then Rx can be measured to high precision. Very small changes in Rx disrupt the balance and are readily detected.At the point of balance, the ratio of R2 / R1 = Rx / R3

Therefore, $R_x = (R_2/R_1) \cdot R_3$

Alternatively, if R1, R2, and R3 are known, but R2 is not adjustable, the voltage difference across or current flow through the meter can be used to calculate the value of Rx, using Kirchhoff's circuit laws (also known as Kirchhoff's rules). This setup is frequently used in strain gauge and resistance thermometer measurements, as it is usually faster to read a voltage level off a meter than to adjust a resistance to zero the voltage.

$$I_3 \ - I_x \ + I_g = 0$$
$$I_1 \ - I_2 \ - I_g = 0$$

Then, Kirchhoff's second rule is used for finding the voltage in the loops ABD and BCD:

$$(I_3 \cdot R_3) - (I_g \cdot R_g) - (I_1 \cdot R_1) = 0$$
$$(I_x \cdot R_x) - (I_2 \cdot R_2) + (I_g \cdot R_g) = 0$$

The bridge is balanced and Ig = 0, so the second set of equations can be rewritten as:

$$I_3 \cdot R_3 = I_1 \cdot R_1$$
$$I_x \cdot R_x = I_2 \cdot R_2$$

Then, the equations are divided and rearranged, giving:

$$R_x = \frac{R_2 \cdot I_2 \cdot I_3 \cdot R_3}{R_1 \cdot I_1 \cdot I_x}$$

From the first rule, I3 = Ix and I1 = I2. The desired value of Rx is now known to be given as:

$$R_x = \frac{R_3 \cdot R_2}{R_1}$$

If all four resistor values and the supply voltage (VS) are known, the voltage across the bridge (VG) can be found by working out the voltage from each potential divider and subtracting one from the other. The equation for this is:

$$V_G = \frac{R_x}{R_3 + R_x} V_s - \frac{R_2}{R_1 + R_2} V_s$$

This can be simplified to:

$$V_G = \left(\frac{R_x}{R_3 + R_x} - \frac{R_2}{R_1 + R_2} \right) V_s$$

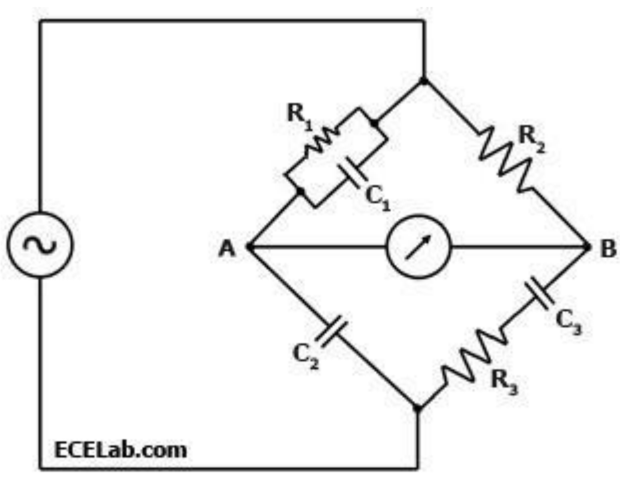
With node B being (VG) positive, and node D being (VG) negative.

Significance :

The Wheatstone bridge illustrates the concept of a difference measurement, which can be extremely accurate. Variations on the Wheatstone bridge can be used to measure capacitance, inductance, impedance and other quantities, such as the amount of combustible gases in a sample, with an explosimeter. The Kelvin bridge was specially adapted from the Wheatstone bridge for measuring very low resistances. In many cases, the significance of measuring the unknown resistance is related to measuring the impact of some physical phenomenon - such as force, temperature, pressure, etc. - which thereby allows the use of Wheatstone bridge in measuring those elements indirectly.

Schering Bridge:

A Schering Bridge is a bridge circuit used for measuring an unknown electrical capacitance and its dissipation factor. The dissipation factor of a capacitor is the the ratio of its resistance to its capacitive reactance. The Schering Bridge is basically a four-arm alternating-current (AC) bridge circuit whose measurement depends on balancing the loads on its arms. Figure 1 below shows a diagram of the Schering Bridge.



The Schering Bridge

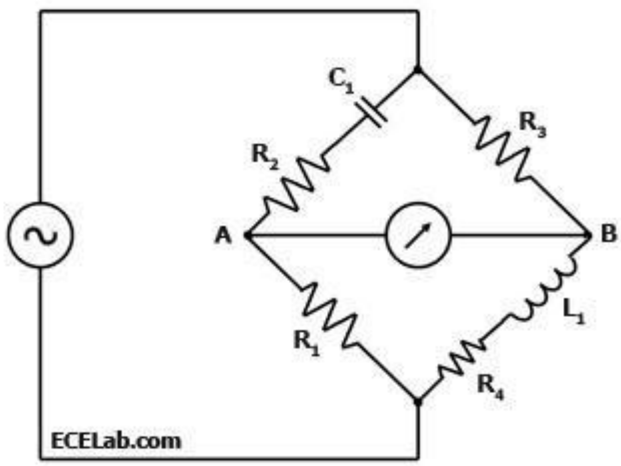
In the Schering Bridge above, the resistance values of resistors R1 and R2 are known, while the resistance value of resistor R3 is unknown. The capacitance values of C1 and C2 are also known, while the capacitance of C3 is the value being measured. To measure R3 and C3, the values of C2 and R2 are fixed, while the values of R1 and C1 are adjusted until the current through the ammeter between points A and B becomes zero. This happens when the voltages at points A and B are equal, in which case the bridge is said to be 'balanced'. When the bridge is balanced, Z1/C2 = R2/Z3, where Z1 is the impedance of R1 in parallel with C1 and Z3 is the impedance of R3 in series with C3. In an AC circuit that has a capacitor, the capacitor contributes a capacitive reactance to the impedance. The capacitive reactance of a capacitor C is $1/2\pi fC$.

As such, $Z1 = R1/[2\pi fC1((1/2\pi fC1) + R1)] = R1/(1 + 2\pi fC1R1)$ while $Z3 = 1/2\pi fC3 + R3$. Thus, when the bridge is balanced:
$2\pi fC2R1/(1+2\pi fC1R1) = R2/(1/2\pi fC3 + R3)$; or
$2\pi fC2(1/2\pi fC3 + R3) = (R2/R1)(1+2\pi fC1R1)$; or
$C2/C3 + 2\pi fC2R3 = R2/R1 + 2\pi fC1R2$.

When the bridge is balanced, the negative and positive reactive components are equal and cancel out, so
$2\pi fC2R3 = 2\pi fC1R2$ or
$R3 = C1R2 / C2$.

Similarly, when the bridge is balanced, the purely resistive components are equal, so
$C2/C3 = R2/R1$ or
$C3 = R1C2 / R2$.

A Hay Bridge is an AC bridge circuit used for measuring an unknown inductance by balancing the loads of its four arms, one of which contains the unknown inductance. One of the arms of a Hay Bridge has a capacitor of known characteristics, which is the principal component used for determining the unknown inductance value. Figure 1 below shows a diagram of the Hay Bridge.



The Hay Bridge :

As shown in Figure 1, one arm of the Hay bridge consists of a capacitor in series with a resistor (C1 and R2) and another arm consists of an inductor L1 in series with a resistor (L1 and R4). The other two arms simply contain a resistor each (R1 and R3). The values of R1and R3 are known, and R2 and C1 are both adjustable. The unknown values are those of L1 and R4.Like other bridge circuits, the measuring ability of a Hay Bridge depends on 'balancing' the circuit. Balancing the circuit in Figure 1 means adjusting R2 and C1 until the current through the ammeter between points A and B becomes zero. This happens when the voltages at points A and B are equal. When the Hay Bridge is balanced, it follows that Z1/R1 = R3/Z2 wherein Z1 is the impedance of the arm containing C1 and R2 while Z2 is the impedance of the arm containing L1 and R4.

Thus, Z1 = R2 + 1/(2πfC) while Z2 = R4 + 2πfL1.

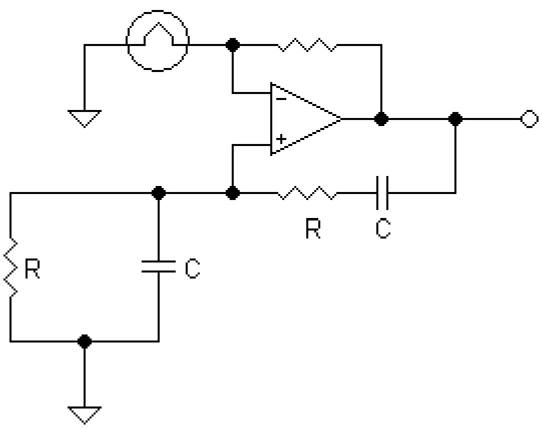
Mathematically, when the bridge is balanced,
[R2 + 1/(2πfC1)] / R1 = R3 / [R4 + 2πfL1]; or [R4 + 2πfL1] = R3R1 / [R2 + 1/(2πfC1)]; or R3R1 = R2R4 + 2πfL1R2 + R4/2πfC1 + L1/C1.

When the bridge is balanced, the reactive components are equal, so
2πfL1R2 = R4/2πfC1, or R4 = (2πf)2L1R2C1.

Substituting R4, one comes up with the following equation:
R3R1 = (R2+1/2πfC1)((2πf)2L1R2C1) + 2πfL1R2 + L1/C1; or L1 = R3R1C1 / (2πf)2R22C12 + 4πfC1R2 + 1); or
L1 = R3R1C1 / [1 + (2πfR2C1)2] after dropping the reactive components of the equation since the bridge is balanced.
Thus, the equations for L1 and R4 for the Hay Bridge in Figure 1 when it is balanced are:
L1 = R3R1C1 / [1 + (2πfR2C1)2]; and
R4 = (2πfC1)2R2R3R1 / [1 + (2πfR2C1)2]

Wien bridge :

A Wien bridge oscillator is a type of electronic oscillator that generates sine waves. It can generate a large range of frequencies. The circuit is based on an electrical network originally developed by Max Wien in 1891. The bridge comprises four resistors and two capacitors. It can also be viewed as a positive feedback system combined with a bandpass filter. Wien did not have a means of developing electronic gain so a workable oscillator could not be realized.The modern circuit is derived from William Hewlett's 1939 Stanford University master's degree thesis. Hewlett, along with David Packard co-founded Hewlett-Packard. Their first product was the HP 200A, a precision sine wave oscillator based on the Wien bridge. The 200A was one of the first instruments to produce such low distortion.



The frequency of oscillation is given by:

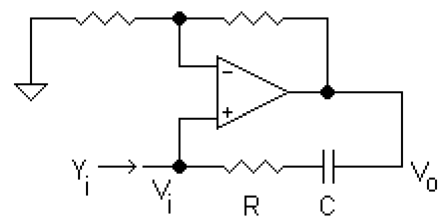
$$f = \frac{1}{2\pi RC}$$

Amplitude stabilization :

The key to Hewlett's low distortion oscillator is effective amplitude stabilization. The amplitude of electronic oscillators tends to increase until clipping or other gain limitation is reached. This leads to high harmonic distortion, which is often undesirable.Hewlett used an incandescent bulb as a positive temperature coefficient (PTC) thermistor in the oscillator feedback path to limit the gain. The resistance of light bulbs and similar heating elements increases as their temperature increases. If the oscillation frequency is significantly higher than the thermal time constant of the heating element, the radiated power is proportional to the oscillator power. Since heating elements are close to black body radiators, they follow the Stefan-Boltzmann law. The radiated power is proportional to T4,so resistance increases at a greater rate than amplitude. If the gain is inversely proportional to the oscillation amplitude, the oscillator gain stage reaches a steady state and operates as a near ideal class A amplifier,achieving very low distortion at the frequency of interest. At lower frequencies the time period of the oscillator approaches the thermal time constant of the thermistor element and the output distortion starts to rise significantly.

Light bulbs have their disadvantages when used as gain control elements in Wien bridge oscillators, most notably a very high sensitivity to vibration due to the bulb's microphonic nature amplitude modulating theoscillator output, and a limitation in high frequency response due to the inductive nature of the coiled filament. Modern Wien bridge oscillators have used other nonlinear elements, such as diodes, thermistors, field effect transistors, or photocells for amplitude stabilization in place of light bulbs. Distortion as low as 0.0008% (-100 dB) can be achieved with only modest improvements to Hewlett's original circuit.Wien bridge oscillators that use thermistors also exhibit "amplitude bounce" when the oscillator frequency is changed. This is due to the low damping factor and long time constant of the crude control loop, and disturbances cause the output amplitude to exhibit a decaying sinusoidal response. This can be used as a rough figure of merit, as the greater the amplitude bounce after a disturbance, the lower the output distortion under steady state conditions.

Analysis :



Input admittance analysis

If a voltage source is applied directly to the input of an ideal amplifier with feedback, the input current will be:

$$i_{in} = \frac{v_{in} - v_{out}}{Z_f}$$

Where vin is the input voltage, vout is the output voltage, and Zf is the feedback impedance. If the voltage gain of the amplifier is defined as:

$$A_v = \frac{v_{out}}{v_{in}}$$

And the input admittance is defined as:

$$Y_i = \frac{i_{in}}{v_{in}}$$

Input admittance can be rewritten as:

$$Y_i = \frac{1 - A_v}{Z_f}$$

For the Wien bridge, Zf is given by:

$$Z_f = R + \frac{1}{j\omega C}$$

$$Y_i = \frac{(1 - A_v)\left(\omega^2 C^2 R + j\omega C\right)}{1 + (\omega C R)^2}$$

If Av is greater than 1, the input admittance is a negative resistance in parallel with an inductance. The inductance is:

$$L_{in} = \frac{\omega^2 C^2 R^2 + 1}{\omega^2 C\,(A_v - 1)}$$

If a capacitor with the same value of C is placed in parallel with the input, the circuit has a natural resonance at:

$$\omega = \frac{1}{\sqrt{L_{in} C}}$$

Substituting and solving for inductance yields:

$$L_{in} = \frac{R^2 C}{A_v - 2}$$

If Av is chosen to be 3:

Lin = R2C

Substituting this value yields:

$$\omega = \frac{1}{RC}$$

Or:

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

Similarly, the input resistance at the frequency above is:

$$R_{in} = \frac{-2R}{A_v - 1}$$

For A_v=3 and $R_{in=}$-R

If a resistor is placed in parallel with the amplifier input, it will cancel some of the negative resistance. If the net resistance is negative, amplitude will grow until clipping occurs. Similarly, if the net resistance is positive, oscillation amplitude will decay. If a resistance is added in parallel with exactly the value of R, the net resistance will be infinite and the circuit can sustain stable oscillation at any amplitude allowed by the amplifier.Notice that increasing the gain makes the net resistance more negative, which increases amplitude. If gain is reduced to exactly 3 when a suitable amplitude is reached, stable, low distortion oscillations will result. Amplitude stabilization circuits typically increase gain until a suitable output amplitude is reached. As long as R, C, and the amplifier are linear, distortion will be minimal.

QUESTION BANK

UNIT I

BASIC MEASUREMENT CONCEPTS

PART – A (2 Marks)

1. **What are the basic elements of a generalized measurement system?**

   The basic elements are,

   - Primary sensing element which is generally a transducer.
   - Data conditioning element which further consists of variable conversion element and variable manipulation element.
   - Data transmission and presentation elements which include data transmission system and data display system.

2. **List any four static characteristics of a measuring system.**

   The various static characteristics of a measuring system are accuracy, precision, error, resolution, sensitivity, reproducibility, stability, linearity etc.

3. **Define the term accuracy.**

   The accuracy is defined as the degree of clones with which the instrument reading approaches the true value of the quantity to be measured. It indicates the ability of an instrument to indicate the true of the quantity.

4. **Define the term precision.**

   It is the measure of the consistency or repeatability of measurements. It denotes the amount by which the individual readings are departed about the average of number of readings.

5. **What is an error?**

   The algebraic difference between the indicated value and the true value of the quantity to be measured is called an error.

6. **What is calibration?**

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

7. **Classify the errors that may occur in an instrument ?**

   i. Gross errors
   ii. Systematic errors
   iii. Instrumental errors
   iv. Environmental errors

v. Observational errors
vi. Random Errors

8. What are the sources of errors in DC voltage measurement?

In DC voltage measurement, the various possible errors are,

- The friction in moving system
- The heat generated changes the resistance of working coil, causing errors.
- The aging of permanent magnet and control spring.

9. What is a transfer instrument?

A transfer instrument is one which is calibrated with DC source and used without any modifications for AC measurements. It has same accuracy for AC and DC measurements.

10. Write the two conditions to be satisfied to make an AC bridge balance

$$|Z1\ Z4| = |Z2\ Z3|$$

$$\theta 1 + \theta 4 = \theta 2 + \theta 3$$

11. Define Standard deviation?

The standard deviation or root mean square deviation of a sample is both mathematically more convenient and statistically more meaningful for analyzing grouped data than is the average deviation by definition, the standard deviation of a sample is given by

$$S = \sqrt{\frac{\in(\bar{x}-x_i)^2}{n}} = \sqrt{\frac{\in d_i^2}{n}}$$

12. What are all the different types of standard?

- International standards
- Primary standards
- Secondary standards
- Working standards

13. Define arithmetic mean?

The average value or arithmetic mean value is the most probable value obtained from a series of readings of a given quantity. As a general rule the more readings the more closely the computed average represents the most probable value. The average value $\bar{x}$ is calculated by taking the sum of all the readings and divided by the number of readings, so that

$$\bar{x} = \frac{\in x_1}{n} = \frac{x_1+x_2+x_3+\cdots+x_n}{n}$$

Where $\bar{x}$ = the average value or arithmetic mean

x_1 = the value of the i th reading

n = the number of readings

14. What is meant by average deviation?

The mean or average is a measure how much the data is varied from the average value. The mean $\overline{D}$ is calculated by adding all the absolute values of deviations of a set of measured values and dividing this sum by the number of observations 'n' so that

$$\overline{D} = \frac{|d_1|+|d_2|+\cdots|d_n|}{n}$$

$$\overline{D} = \frac{\in|d_i|}{n}$$

15. What is a standard?

A standard is a physical representation of a unit of measurement. A known accurate measure of physical quantity is termed as standard.

16. Define systematic error?

The errors that occur due to instrument errors, environmental errors and observational errors are classified into instrumental errors.

17. What are all the types of systematic errors?

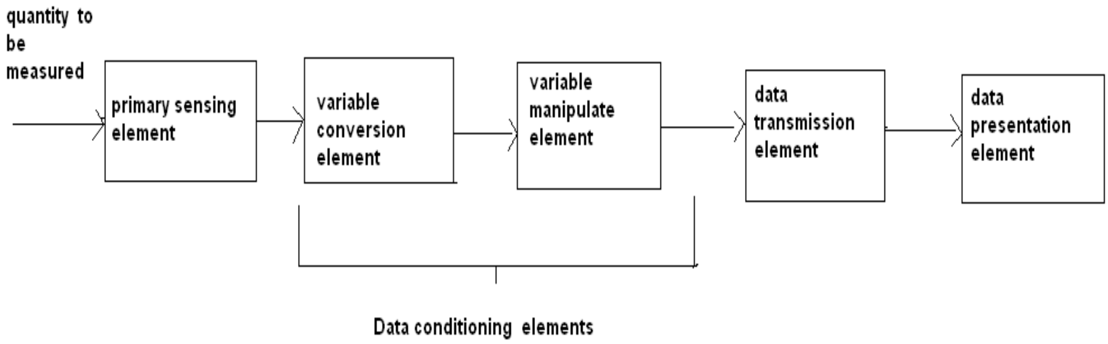
- Instrumental errors
- Environmental errors
- Observational errors

PART-B (16 Marks)

1. Explain and detail about functional elements of an instruments? (AU APRIL 2007,2009) (16)

Figures shows the functional elements of a generalized measurement system most of the measurement systems. Contain three main functional elements.

1. Primary sensing element
2. Variable conversion element
3. Data presentation element



Primary sensing element:

The quantity under measurement makes its first contact with primary sensing element of a measurement system here, the primary sensing element transducer. This transducer converts measured into an analogous electrical signal.

Variable conversion element:

The output of the primary sensing element is the electrical signal. It may be a voltage a frequency or some other electrical parameter. But this output is not suitable for this system.

For the instrument to perform the desired function, it may be necessary to convert this output to some other suitable form while retaining the original signal. Consider an example, suppose output is an analog signal form and the next of system accepts input signal only in digital form . Therefore we have to use and to digital converter in this system.

Variable manipulation element:

The main function of variable manipulation element is to manipulation element is to manipulate the signal presented to it preserving the original nature of the signal. Here, manipulation means a change in numerical value of the signal.

Consider a small example, an electric amplifier circuit accepts a small voltage signal as input and produces an output signal which is also voltage but of greater amplifier. Thus voltage amplifier acts as a variable manipulation element.

Data presentation element:

The information about the quantity under measurement has to be conveyed to the personal handling the instrument or system for control or analysis purposes. The information conveyed must be in the form of intelligible to the personnel. The above function is done by data presentation element.

The output or data of the system can be monitored by using visual display devices may be analog or digital device like ammeter, digital meter etc. In case the data to be record, we can use analog or digital recording equipment. In industries , for control and analysis purpose we can use computers.

The final stage in a measurement system is known as terminating stage . when a control device is used for the final measurement stage it is necessary to apply some feedback to the input signal to accomplish the control 0bjective.

The term signal conditioning includes many other functions in addition to variable conversion and variable manipulation. In fact the element that follows the primary sensing element in any instrument or instrumentation system should be called signal conditioning element.

When the element of an instrument is physically separated, it becomes necessary to transmit data from one to another. This element is called transmitting element. The signal conditioning and transmitting stage is generally known as intermediate stage.

2. Explain the static and dynamic character of an instrument ? (AU APRIL 2005,2006)

(16)

Static characteristics:

The static characteristics of a instrument are consider for instrument which are used to measure an unvarying process condition. All the static perform ate characteristics are obtained by one form another of a process called calibration. The main static characteristics are accrued sensitivity, resolution, precision, drift static error, dead zone etc.

Accuracy:

It is a measure of the closeness with which an instrument measure the true value of a quantity.

Precision:

It is a measure of the consistency or repeatability of a series of measurements. Although accuracy implies precision, precision does not necessarily imply accuracy. A precise instrument can be very inaccurate. The precision of a given measurement can be very inaccurate. The precision of a given measurement can be given by

Precision = $1-\left|\frac{x_{i-\bar{x}}}{x_i}\right|$

Sensitivity:

It is a measure of the change in reading of an instrument for a given change in the measured quantity.

Resolution:

It is the smallest change in the measured quantity that will produce a deductible change in the instrument reading.

Error:

Error is the deviation from the true value of the measured quantity . Error can be expressed as absolute quantity or as a percentage.

% error = $\left|\frac{x_e - x_m}{x_e}\right|$

Range:

The range of an instrument describes the limits of magnitude over which a quantity may be measured. It is normally specified by stating its lower and upper limits. For example an ammeter whose scale reads from 0 to 1 MB is said to have a range from 0 to 1 MA

Span:

The span of an instrument is the algebraic difference between the upper and lower limits of the instrument range for a- 10 MA to + 10 MA galvanometer, the span is than 20 MA

Drift:

It is the variation of the measured value with time. Perfect reproducibility means that the instrument has no drift. There are 3 types of drifts. They are zero drift and zonal drift. Drift is an undesirable quality in industrial instruments because it is rarely apparent and cannot be easily compensated for thus it must be carefully guarded by continuous prevention, inspection and maintenance.

Dead zone:

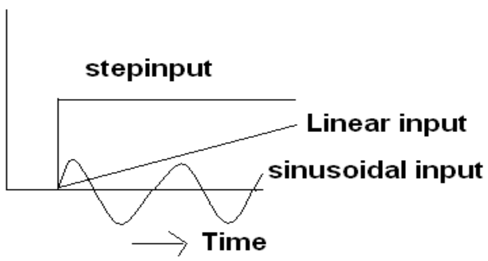
It is defined as the largest change of input quantity for which there is no output of the instrument. The factors which produce dead zone are hysterias and back lash in the instrument.

Threshold:

It is clear that if the input into the instrument is increased very gradually from 0, there will be some minimum value below which no output change can be detected. This minimum value defines the threshold of the instrument.

Dynamic characteristics:

The dynamic behavior of an instrument can be determined by applying some form of known and predetermined input to its primary element and then study the output , movement of the pointer generally the behavior is judged for three types of inputs.



Step change:

In this case the input is changed suddenly to a finite value and then remains constant.

Linear change:

In this case the input changes linearly with time.

Sinusoidal change:

In this case the magnitude of the input changes in accordance with a sinusoidal function of constant amplitude.

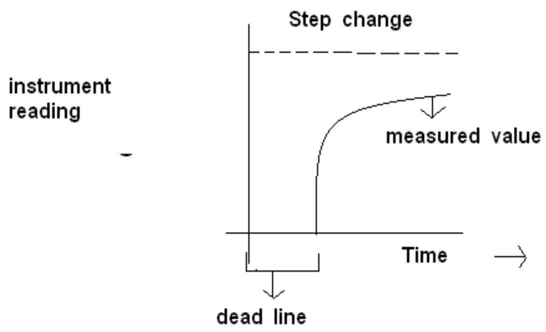
The dynamic characteristics of any instrument is defined and evaluated by the following terms.

Speed of response:

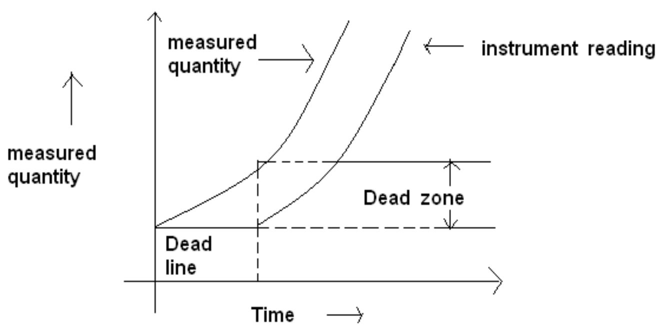
It is the rapidly with which an instrument responds to changes in the measured quantity.

Lag:

It Is the retardation or delay in the response of an instrument to changes in the measured quantity. The measuring lag can be either of retardation type in which case the response of the instrument beings immediately on a changes in measured variable.



Generally the dead time lag is very small. Instrument having appreciable dead time are not satisfactory for measuring a variable than fluctuates rapidly. The dead time can also be caused by a finite dead zone in the instrument as a result of friction. In such cases, the instrument does not respond for a certain time delay and acts only sufficiently to overcome the starting friction. The dead time due to this cause depends on how fast the measured variable is changing and on the extent of the instrument dead zone. It is shown in figure.



3. Explain in detail about error? (AU APRIL 2003,2004,2006.DEC 2005) (16)

A major skill in taking measurements is the ability to interpret results in terms of possible errors. No matter how carefully the measurements are taken and no matter how accurate the instruments that are used, some error will always be present. The tree systematic errors and random errors.

Grass error:

This class of errors mainly covers human mistakes in reading or using instruments and in recording and calculating measured values. As long a human beings are involved. Some gross errors will definitely be committed. Although complete elimination of gross errors is probably impossible, are should try to anticipate and correct them. Some gross errors are easily detected while others may be very difficult to detect. The experiment may grossly misread the scale.

- Great care should be taken in reading and recording the data.
- Two , three or even more readings should be taken for the quantity under measurement. These readings should be taken preferably by different experimenters and the readings should be taken at a different reading point to avoid re-reading with the same error. Never place complete dependence on one reading but take at least three separate readings. Preferably under conditions in which instruments are switched off- on.

Systematic error:

These types of errors are divided into three categories such as instrumental errors. Environmental errors and observational errors.

Instrumental errors:

These errors arise due to inherent short comings in the instruments misuse of the instruments and loading effects.

Environmental errors:

These errors are due to conditions external to the measuring device including conditions in the area surrounding the instrument. These may be effects of temperature, pressure, humidity, dust, vibrations or of external magnetic or electrostatic fields.

The connective measures employed to eliminate to reduce these undesirable effects.

Random errors:

This occurs are due to unknown causes and are observed when the magnitude and polarity of a measurement future in an unpredictable manner. Some of the more common random errors are:

(i) Rounding error:

This occurs when readings are between scale graduations and the reading is rounded up or down to the nearest graduation.

(ii) Periodic error:

This occurs when an analog meter reading swings or fluctuates about the correct reading. In addition, the meter reading quickly changes in the immediate vicinity of the corrected value, but changes slowly at the extremes of the swing. Since it could be easier to read the meter when it is slowly changing, the correct value would be less likely read than an incorrect value.

The other random errors are due to noise backlash and ambient influence. Random errors cannot normally be predicted or corrected but they can be minimized by skilled observes using a well maintained quality instrument.

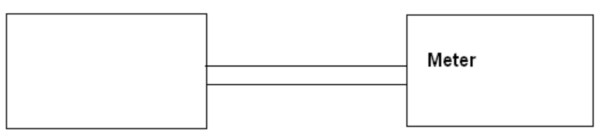
4. Define calibration and explain difference methods of calibration? (AU APRIL 2007, Dec 2008) (16)

All measuring instruments are to prove themselves their ability to measure reliably and accurately . For this the results of measurement are to be compared with higher standards which are traceable to national or international standards. The procedure involved is termed as calibration.

Calibration is thus a set of operations that establish the relationship between the values that are indicated by the measuring instrument and corresponding known values of measured.

Primary calibration:

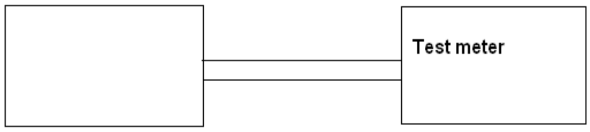
If the instrument is calibrated against primary standards, then the calibration is called primary calibration. After the primary calibration, the instrument can be used as a secondary calibration instrument.



Primary Standard

Secondary calibration:

The secondary calibration instrument is used as secondary for further calibration of other devices of lesser accuracy. This type of instruments are used in general laboratory practice as well as in the industry because they are practical calibration sources.



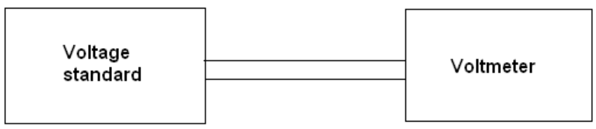
Secondary Standard

Secondary calibration can further be classified into two types.

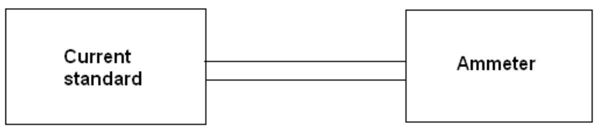
- ❖ Direct calibration
- ❖ Indirect calibration

Direct calibration:

Direct calibration with a known input source is in general of the same order of accuracy as primary calibration. So, the instrument which are calibrated directly are also used as secondary calibration instruments.



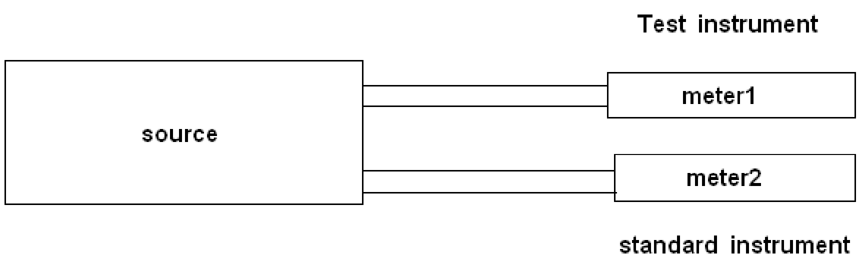
Calibration of voltmeter



Calibration of Ammeter

Indirect calibration:

This procedure is based on the equivalence of two different devices adopting same similarity concept.



5. What is the standard and write and detail about different types of standards? (AU APRIL DEC 2003) (16)

A standard is a physical representation of a unit of measurement. A known accurate measure of physical quantity is termed as standard. These standards are used to determine the values of other physical quantities by the comparison methods.

In fact , a unit is realized by reference to a material standard or to natural phenomena, including physical and atomic constants. For example, the fundamental unit of length in the international system (SI) is the meter defined as the distance between two fine lines engraved on gold plugs near the ends of a platinum-iridium alloy at 0° c and mechanically supported in a prescribed manner.

Based on the functions an application. Standards are classified into four categories as

(i) International standards
(ii) Primary standards
(iii) Secondary standards
(iv) Working standards

International standards:

International standards are defined by international agreement. They are periodically evaluated and checked by absolute measurement in terms of fundamental units of physics . They represent certain units of measurement to the closest possible accuracy attainable by the science and technology of measurement . These international standards are not available to ordinary users for measurements and calibrations.

International ohms:

It is defined as the resistance offered by a column of mercury having a mass of 14.4521gms, uniform cross sectional area and length of 106.300 cm, to the flow at constant current at the melting point of ice.

International Amperes:

It is an unvarying current , which when passed through a solution of silver nitrate in water deposits silver at the rate of 0.00111gm/5.

Primary standards:

The Principle function of primary standards is the calibration and verification of secondary standards. Primary standards are maintained at the National standards Laboratories in different countries. They are not available for use outside the National Laboratory. These Primary standards are absolute standards of high accuracy that can be used as ultimate reference standard .

Secondary standards:

Secondary standards are basic reference standards used by measurement and calibration laboratories in industries. These secondary standards are maintained by the particular industry to which they belong. Each industry has its own secondary standard to the National Standards Laboratory for calibration, the National Standards Laboratory returns the secondary standards to the particular industrial laboratory with a certification of measuring accuracy in terms of a primary standards.

Working standards:

Working standards are the principal tools of a measurement laboratory. These standards are used to check and calibrate laboratory instruments for accuracy and performance for example, manufactures of electronic components Such as capacitors resistors etc. Use a standard called a working standard for checking the component values being manufactured a standard resistor for checking of resistance value manufactured.

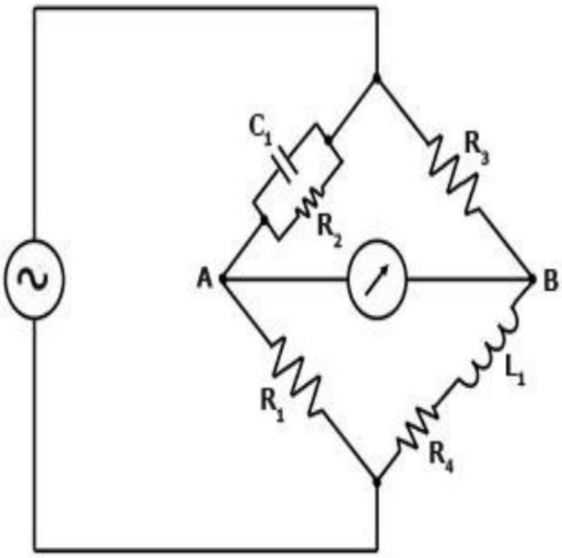
6. Maxwell inductance Bridge. (AU DEC 2004, APRIL 2005) (16)

The maxwell bridge is used to measure unknown inductance in terms of calibrated resistance and capacitance. Calibration-grade inductors are more difficult to manufacture than capacitors of similar precision, and so the use of a simple "symmetrical" inductance bridge is not always practical.

Because the phase shifts of inductors and capacitors are exactly opposite each other, capacitive impedance can balance out an inductive impedance if they are located in opposite legs of a bridge, as they are here.

Another advantage of using a Maxwell bridge to measure inductance rather than a symmetrical inductance bridge is the elimination of measurement error due to mutual inductance between two inductors.

Magnetic fields can be difficult to shield, and even a small amount of coupling between coils in a bridge can introduce substantial errors in certain conditions. With no second inductor to react with in the Maxwell bridge, this problem is eliminated.



As shown in Figure, one arm of the Maxwell bridge consists of a capacitor in parallel with a resistor (C1 and R2) and another arm consists of an inductor L1 in series with a resistor (L1 and R4). The other two arms just consist of a resistor each (R1 and R3). The values of R1 and R3 are known, and R2 and C1 are both adjustable. The unknown values are those of L1 and R4.

Like other bridge circuits, the measuring ability of a Maxwell Bridge depends on 'balancing' the circuit. Balancing the circuit in Figure 1 means adjusting C1 and R2 until the current through the bridge between points A and B becomes zero. This happens when the voltages at points A and B are equal.

When the Maxwell Bridge is balanced, it follows that Z1/R1 = R3/Z2 wherein Z1 is the impedance of C2 in parallel with R2, and Z2 is the impedance of L1 in series with R4. Mathematically, $Z1 = R2 + 1/(2\pi fC1)$; while $Z2 = R4 + 2\pi fL1$.

Thus, when the bridge is balanced,

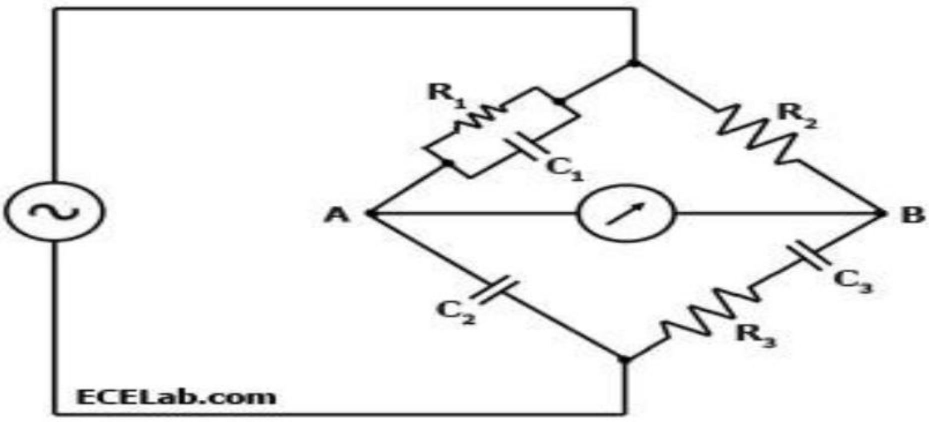
$(R2 + 1/(2\pi fC1)) / R1 = R3 / [R4 + 2\pi fL1]$; or

$R1R3 = [R2 + 1/(2\pi fC1)] [R4 + 2\pi fL1]$;

When the bridge is balanced, the negative and positive reactive components cancel out, so R1R3 = R2R4, or R4 = R1R3/R2

7.Schering bridge (AU APRIL DEC 2006) (16).

A **Schering Bridge** is a bridge circuit used for measuring an unknown electrical capacitance and its dissipation factor. The dissipation factor of a capacitor is the the ratio of its resistance to its capacitive reactance. The Schering Bridge is basically a four-arm alternating-current (AC) bridge circuit whose measurement depends on balancing the loads on its arms. Figure 1 below shows a diagram of the Schering Bridge.



In the Schering Bridge above, the resistance values of resistors R1 and R2 are known, while the resistance value of resistor R3 is unknown. The capacitance values of C1 and C2 are also known, while the capacitance of C3 is the value being measured. To measure R3 and C3, the values of C2 and R2 are fixed, while the values of R1 and C1 are adjusted until the current through the ammeter between points A and B becomes zero. This happens when the voltages at points A and B are equal, in which case the bridge is said to be 'balanced'.

When the bridge is balanced, Z1/C2 = R2/Z3, where Z1 is the impedance of R1 in parallel with C1 and Z3 is the impedance of R3 in series with C3. In an AC circuit that has a capacitor, the capacitor contributes a capacitive reactance to the impedance. The capacitive reactance of a capacitor C is $1/2\pi fC$.

As such, $Z1 = R1/[2\pi fC1((1/2\pi fC1) + R1)] = R1/(1 + 2\pi fC1R1)$ while $Z3 = 1/2\pi fC3 + R3$.

Thus, when the bridge is balanced:

$2\pi fC2R1/(1+2\pi fC1R1) = R2/(1/2\pi fC3 + R3)$; or

$2\pi fC2(1/2\pi fC3 + R3) = (R2/R1)(1+2\pi fC1R1)$; or

$C2/C3 + 2\pi fC2R3 = R2/R1 + 2\pi fC1R2$.

When the bridge is balanced, the negative and positive reactive components are equal and cancel out, so

$2\pi fC2R3 = 2\pi fC1R2$ or

R3 = C1R2 / C2.

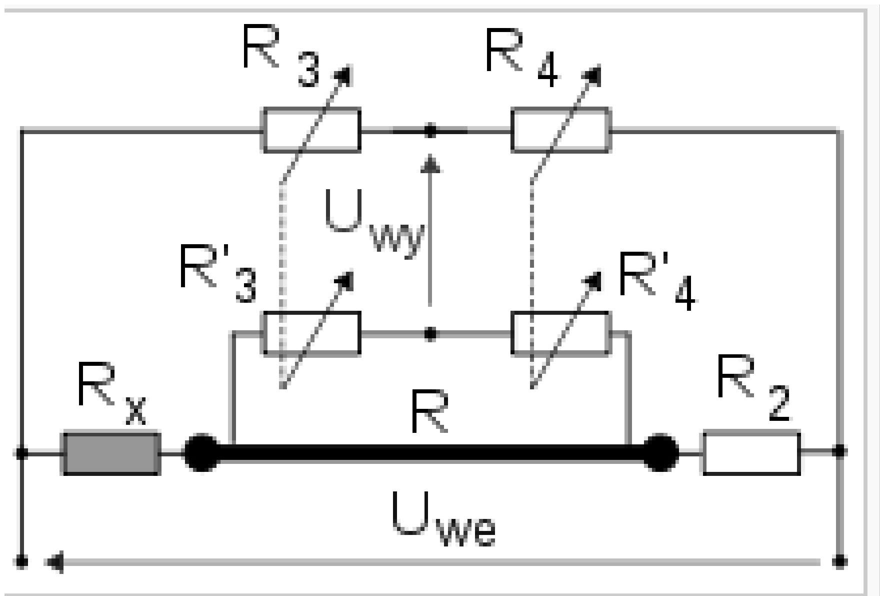
Similarly, when the bridge is balanced, the purely resistive components are equal, so

C2/C3 = R2/R1 or

C3 = R1C2 / R2.

Note that the balancing of a Schering Bridge is independent of frequency.

7. Kelvin double Bridge (AU APRIL 2007) (8).



A **Kelvin bridge** (also called a **Kelvin double bridge** and some countries **Thomson bridge**) is a measuring instrument invented by William Thomson, 1st Baron Kelvin. It is used to measure an unknown electrical resistance below 1 Ω. Its operation is similar to the Wheatstone bridge except for the presence of additional resistors. These additional low value resistors and the internal configuration of the bridge are arranged to substantially reduce measurement errors introduced by voltage drops in the high current (low resistance) arm of the bridge

ACCURACY

There are some commercial devices reaching accuracies of 2% for resistance ranges from 0.000001 to 25 Ω. Often, ohmmeters include Kelvin bridges, amongst other measuring instruments, in order to obtain large measure ranges, for example, the Valhalla 4100 ATC Low-Range Ohmmeter.

The instruments for measuring sub-ohm values are often referred to as low-resistance ohmmeters, milli-ohmmeters, micro-ohmmeters, etc

PRINCIPLE OF OPERATION

The measurement is made by adjusting some resistors in the bridge, and the balance is achieved when:

Resistance R should be as low as possible (much lower than the measured value) and for that reason is usually made as a short thick rod of solid copper. If the condition $R3 \cdot R`4 = R`3 \cdot R4$ is met (and value of R is low), then the last component in the equation can be neglected and it can be assumed that:

Which is equivalent to the Wheatstone bridge

9. PMMC Instruments (DEC 2010) (16)

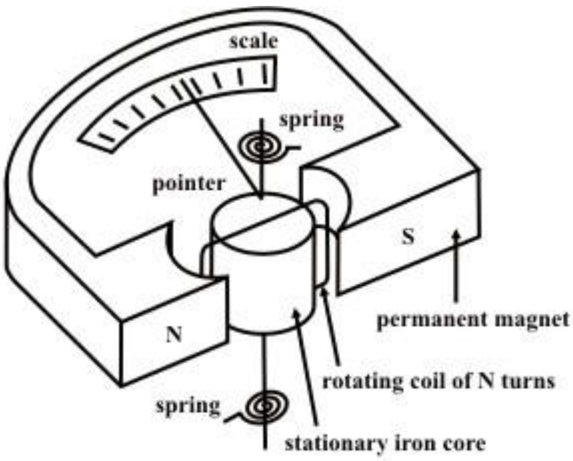


Fig. 42.1(a): Permanent Magnet Moving Coil Instrument.

The permanent magnet moving coil instruments are most accurate type for direct current measurements. The action of these instruments is based on the motoring principle. When a current carrying coil is placed in the magnetic field produced by permanent magnet, the coil experiences a force and moves.

As the coil is moving and the magnet is permanent, the instrument is called permanent magnet moving coil instrument. This basic principle is called D'Arsonval principle. The amount of force experienced by the coil is proportional to the current passing through the coil.

The moving coil is either rectangular or circular in shape. It has number of turns of fine wire. The coil is suspended so that it is free to turn about its vertical axis.

The coil is placed in uniform, horizontal and radial magnetic field of a permanent magnet in the shape of a horse-shoe. The iron core is spherical if coil is circular and is cylindrical if the coil is rectangular. Due to iron core, the deflecting torque increase, increasing the sensitivity of the instrument

The controlling torque is provided by two phosphor bronze hair springs.

The damping torque is provided by eddy current damping. It is obtained by movement of aluminum former, moving in the magnetic field of the permanent

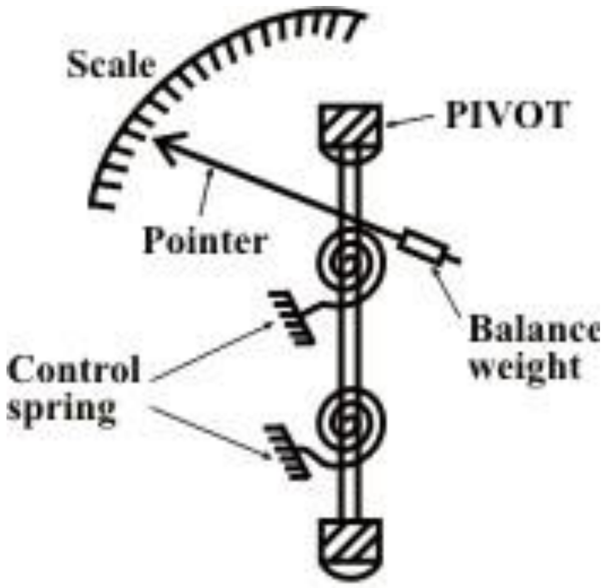


Fig. 42.1(b)

The pointer is carried by the spindle and it moves over a graduated scale. The pointer has light weight so that it deflects rapidly. The mirror is placed below the pointer to get the accurate reading by removing the parallax.

The weight of the instrument is normally counter balanced by the weights situated diametrically opposite and rapidly connected to it. The scale markings of the basic d.c PMMC instruments are usually linearly spaced as the deflecting torque and hence the pointer deflections are directly proportional to the current passing through the coil.

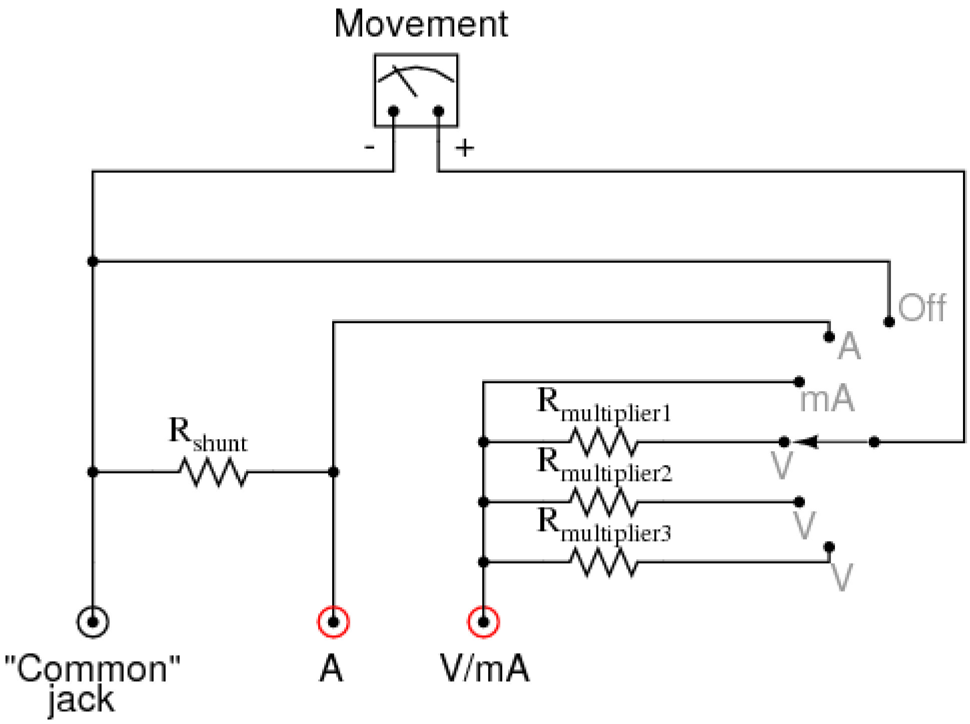
The top view of PMMC instrument is shown in the below image.

The various advantages of PMMC instruments are, It has uniform scale.

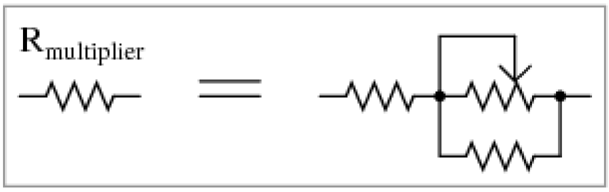
With a powerful magnet, its torque to weight ratio is very high. So operating current of PMMC is small. The sensitivity is high. The eddy currents induced in the metallic former over which coil is wound, provide effective damping. It consumes low power, of the order of 25 W to 200 mW. It has high accuracy. Instrument is free from hysteresis error. Extension of instrument range is possible. Not affected by external magnetic fields called stray magnetic fields

Unit-II BASIC **ELECTRONIC MEASUREMENTS**

Electronic Multimeters :



"$R_{multiplier}$" resistors are actually rheostat networks



Introduction: The cathode-ray oscilloscope (CRO) is a common laboratory instrument that provides accurate time and aplitude measurements of voltage signals over a wide range of frequencies. Its reliability, stability, and ease of operation make it suitable as a general purpose laboratory instrument. The heart of the CRO is a cathode-ray tube shown schematically in Fig. 1.

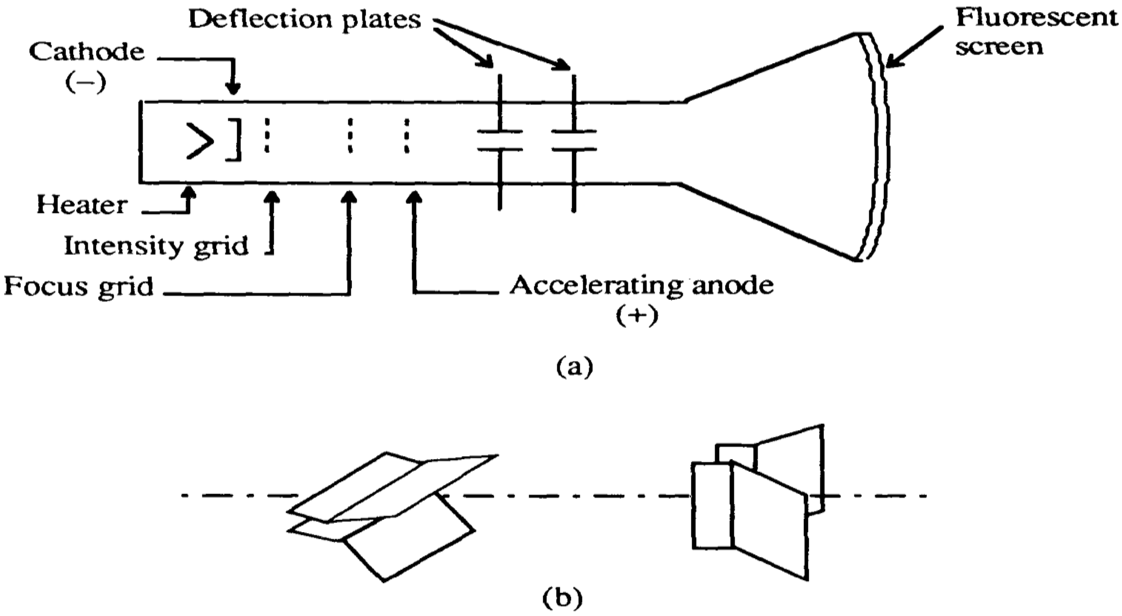


Figure 1. Cathode-ray tube: (a) schematic, (b) detail of the deflection plates.

The cathode ray is a beam of electrons which are emitted by the heated cathode (negative electrode) and accelerated toward the fluorescent screen. The assembly of the cathode, intensity grid, focus grid, and accelerating anode (positive electrode) is called an electron gun. Its purpose is to generate the electron beam and control its intensity and focus. Between the electron gun and the fluorescent screen are two pair of metal plates - one oriented to provide horizontal deflection of the beam and one pair oriented ot give vertical deflection to the beam. These plates are thus referred to as the horizontal and vertical deflection plates. The combination of these two deflections allows the beam to reach any portion of the fluorescent screen. Wherever the electron beam hits the screen, the phosphor is excited and light is emitted from that point. This coversion of electron energy into light allows us to write with points or lines of light on an otherwise darkened screen.In the most common use of the oscilloscope the signal to be studied is first amplified and then applied to the vertical (deflection) plates to deflect the beam vertically and at the same time a voltage that increases linearly with time is applied to the horizontal (deflection) plates thus causing the beam to be deflected horizontally at a uniform (constant> rate. The signal applied to the verical plates is thus displayed on the screen as a function of time. The horizontal axis serves as a uniform time scale.The linear deflection or sweep of the beam horizontally is accomplished by use of a sweep generator that is incorporated in the oscilloscope circuitry. The voltage output of such a generator is that of a sawtooth wave as shown in Fig. 2. Application of one cycle of this voltage difference, which increases linearly with time, to the horizontal plates causes the beam to be deflected linearly with time across the tube face. When the voltage suddenly falls to zero, as at points (a) (b) (c), etc...., the end of each sweep - the beam flies back to its initial position. The horizontal deflection of the beam is repeated periodically, the frequency of this periodicity is adjustable by external controls.

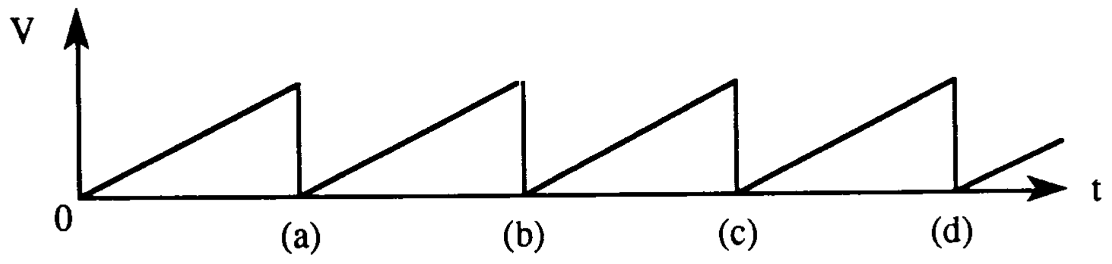


Figure. 2. Voltage difference V between horizontal plates as a function of time t.

To obtain steady traces on the tube face, an internal number of cycles of the unknown signal that is applied to the vertical plates must be associated with each cycle of the sweep generator. Thus, with such a matching of synchronization of the two deflections, the pattern on the tube face repeats itself and hence appears to remain stationary. The persistance of vision in the human eye and of the glow of the fluorescent screen aids in producing a stationary pattern. In addition, the electron beam is cut off (blanked) during flyback so that the retrace sweep is not observed.

CRO Operation: A simplified block diagram of a typical oscilloscope is shown in Fig. 3. In general, the instrument is operated in the following manner. The signal to be displayed is amplified by the vertical amplifier and applied to the vertical deflection plates of the CRT. A portion of the signal in the vertical amplifier is applied to the sweep trigger as a triggering signal. The sweep trigger then generates a pulse coincident with a selected point in the cycle of the triggering signal. This pulse turns on the sweep generator, initiating the sawtooth wave form. The sawtooth wave is amplified by the horizontal amplifier and applied to the horizontal deflection plates. Usually, additional provisions signal are made for applying an external triggering signal or utilizing the 60 Hz line for triggering. Also the sweep generator may be bypassed and an external signal applied directly to the horizontal amplifier.

CRO Controls :

The controls available on most oscilloscopes provide a wide range of operating conditions and thus make the instrument especially versatile. Since many of these controls are common to most oscilloscopes a brief description of them follows.

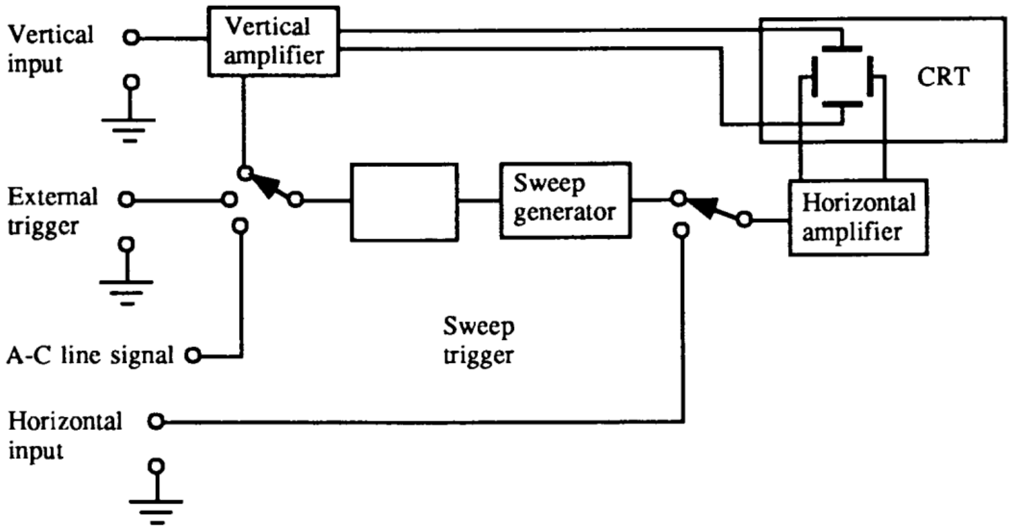


Figure 3. Block diagram of a typical oscilloscope.

CATHODE-RAY TUBE

Power and Scale Illumination: Turns instrument on and controls illumination of the graticule.

Focus: Focus the spot or trace on the screen.

Intensity: Regulates the brightness of the spot or trace.

VERTICAL AMPLIFIER SECTION

Position: Controls vertical positioning of oscilloscope display.

Sensitivity: Selects the sensitivity of the vertical amplifier in calibrated steps.

Variable Sensitivity: Provides a continuous range of sensitivities between the calibrated steps. Normally the sensitivity is calibrated only when the variable knob is in the fully clockwise position.

AC-DC-GND: Selects desired coupling (ac or dc) for incoming signal applied to vertical amplifier, or grounds the amplifier input. Selecting dc couples the input directly to the amplifier; selecting ac send the signal through a capacitor before going to the amplifier thus blocking any constant component.

HORIZONTAL-SWEEP SECTION

Sweep time/cm: Selects desired sweep rate from calibrated steps or admits external signal to horizontal amplifier.Sweep time/cm Variable: Provides continuously variable sweep rates. Calibrated position is fully clockwise.Position: Controls horizontal position of trace on screen.Horizontal Variable: Controls the attenuation (reduction) of signal applied to horizontal aplifier through Ext. Horiz. connector.

TRIGGER

The trigger selects the timing of the beginning of the horizontal sweep.

Slope: Selects whether triggering occurs on an increasing (+) or decreasing (-) portion of trigger signal.

Coupling: Selects whether triggering occurs at a specific dc or ac level.

Source: Selects the source of the triggering signal.

INT - (internal) - from signal on vertical amplifier
EXT - (external) - from an external signal inserted at the EXT. TRIG. INPUT.
LINE - 60 cycle triger

Level: Selects the voltage point on the triggering signal at which sweep is triggered. It also allows automatic (auto) triggering of allows sweep to run free (free run).

CONNECTIONS FOR THE OSCILLOSCOPE

Vertical Input: A pair of jacks for connecting the signal under study to the Y (or vertical) amplifier. The lower jack is grounded to the case.

Horizontal Input: A pair of jacks for connecting an external signal to the horizontal amplifier. The lower terminal is graounted to the case of the oscilloscope.

External Tigger Input: Input connector for external trigger signal.

Cal. Out: Provides amplitude calibrated square waves of 25 and 500 millivolts for use in calibrating the gain of the amplifiers.

Accuracy of the vertical deflection is + 3%. Sensitivity is variable.

Horizontal sweep should be accurate to within 3%. Range of sweep is variable.

Operating Instructions: Before plugging the oscilloscope into a wall receptacle, set the controls as follows:

(a) Power switch at off
(b) Intensity fully counter clockwise
(c) Vertical centering in the center of range

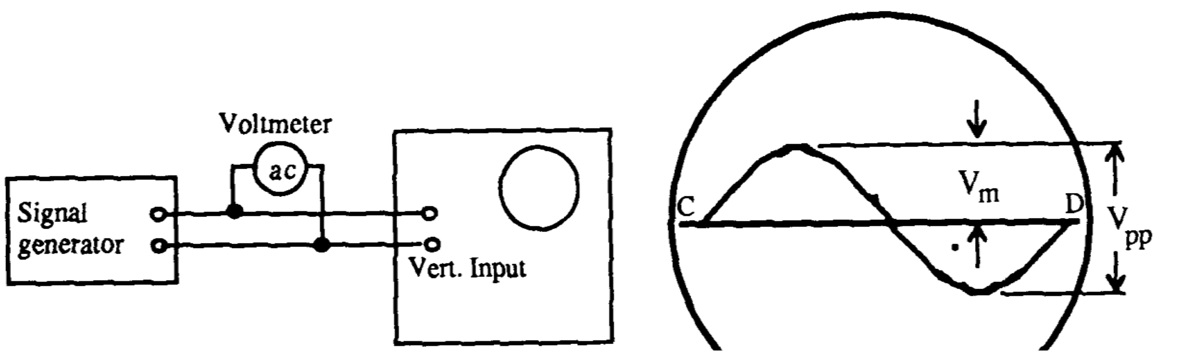
(d) Horizontal centering in the center of range
(e) Vertical at 0.2
(f) Sweep times 1

Plug line cord into a standard ac wall recepticle (nominally 118 V). Turn power on. Do not advance the Intensity Control.Allow the scope to warm up for approximately two minutes, then turn the Intensity Control until the beam is visible on the screen.

PROCEDURE:

I. Set the signal generator to a frequency of 1000 cycles per second. Connect the output from the gererator to the vertical input of the oscilloscope. Establish a steady trace of this input signal on the scope. Adjust (play with)all of the scope and signal generator controls until you become familiar with the functionof each. The purpose fo such "playing" is to allow the student to become so familiar with the oscilloscope that it becomes an aid (tool) in making measurements in other experiments and not as a formidable obstacle. Note: If the vertical gain is set too low, it may not be possible to obtain a steady trace.

II. Measurements of Voltage: Consider the circuit in Fig. 4(a). The signal generator is used to produce a 1000 hertz sine wave. The AC voltmeter and the leads to the vertical input of the oscilloscope are connected across the generator's output. By adjusting the Horizontal Sweep time/cm and trigger, a steady trace of the sine wave may be displayed on the screen. The trace represents a plot of voltage vs. time, where the vertical deflection of the trace about the line of symmetry CD is proportional to the magnitude of the voltage at any instant of time.



To determine the size of the voltage signal appearing at the output of terminals of the signal generator, an AC (Alternating Current) voltmeter is connected in parallel across these terminals (Fig. 4a). The AC voltmeter is designed to read the dc "effective value" of the voltage. This effective value is also known as the "Root Mean Square value" (RMS) value of the voltage.The peak or maximum voltage seen on the scope face (Fig. 4b) is Vm volts and is represented by the distance from the symmetry line CD to the maximum deflection. The relationship between the magnitude of the peak voltage displayed on the scope and the effective or RMS voltage (VRMS) read on the AC voltmeter is

VRMS = 0.707 Vm (for a sine or cosine wave).

Thus

$$V_m = \frac{V_{RMS}}{0.707}$$

Agreement is expected between the voltage reading of the multimeter and that of the oscilloscope. For a symmetric wave (sine or cosine) the value of V_m may be taken as 1/2 the peak to peak signal Vpp.The variable sensitivity control a signal may be used to adjust the display to fill a concenient range of the scope face. In this position, the trace is no longer calibrated so that you cannot just read the size of the signal by counting the number of divisions and multiplying by the scale factor. However, you can figure out what the new calibration is an use it as long as the variable control remains unchanged.Caution: The mathematical prescription given for RMS signals is valid only for sinusoidal signals. The meter will not indicate the correct voltage when used to measure non-sinusoidal signals.

III. Frequency Measurements: When the horizontal sweep voltage is applied, voltage measurements can still be taken from the vertical deflection. Moreover, the signal is displayed as a function of time. If the time base (i.e. sweep) is calibrated, such measurements as pulse duration or signal period can be made. Frequencies can then be determined as reciprocal of the periods.Set the oscillator to 1000 Hz. Display the signal on the CRO and measure the period of the oscillations. Use the horizontal distance between two points such as C to D in Fig. 4b.Set the horizontal gain so that only one complete wave form is displayed.Then reset the horizontal until 5 waves are seen. Keep the time base control in a calibrated position. Measure the distance (and hence time) for 5 complete cycles and calculate the frequency from this measurement. Compare you result with the value determined above.Repeat your measurements for other frequencies of 150 Hz, 5 kHz, 50 kHz as set on the signal generator.

IV. Lissajous Figures: When sine-wave signals of different frequencies are input to the horizontal and vertical amplifiers a stationary pattern is formed on the CRT when the ratio of the two frequencies is an intergral fraction such as 1/2, 2/3, 4/3, 1/5, etc. These stationary patterns are known as Lissajous figures and can be used for comparison measurement of frequencies.Use two oscillators to generate some simple Lissajous figures like those shown in Fig.5. You will find it difficult to maintain the Lissajous figures in a fixed configuration because the two oscillators are not phase and frequency locked. Their frequencies and phase drift slowly causing the two different signals to change slightly with respect to each other.

V. Testing what you have learned: Your instructor will provide you with a small oscillator circuit. Examine the input to the circuit and output of the circuit using your oscilloscope. Measure such quantities as the voltage and frequency of the signals. Specify if they are sinusoidal or of some other wave character. If square wave, measure the frequency of the wave. Also, for square waves, measure the on time (when the voltage is high) and off time (when it is low).

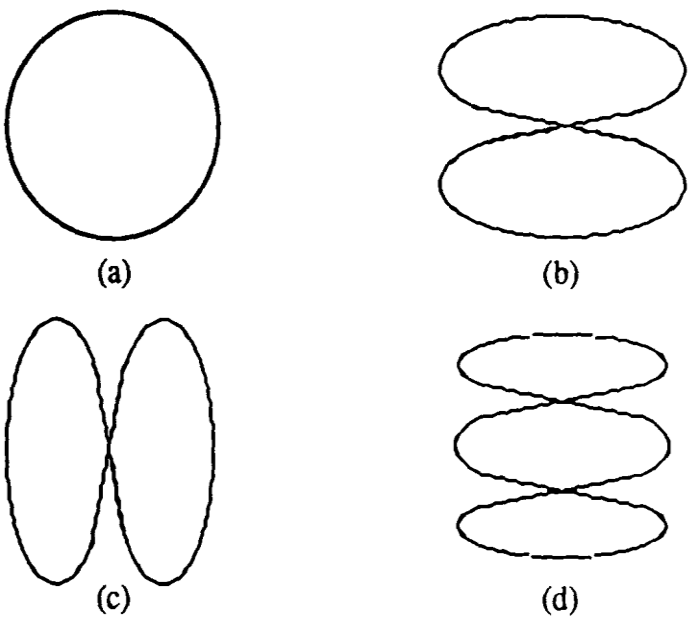


Figure 5. Lissajous figures for horizontal-to-vertical frequency ratios of: (a) 1:1, (b) 2:1, (c) 1:2, and (d) 3:1.

Q meter :

Introduction:

For many years, the Q meter has been an essential piece of equipment for laboratories engaged in the testing of radio frequency circuits. In modem laboratories, the Q meter has been largely replaced by more exotic (and more expensive) impedance measuring devices and today, it is difficult to find a manufacturer who still makes a Q meter. For the radio amateur, the Q meter is still a very useful piece of test equipment and the writer has given some thought to how a simple Q meter could be made for the radio shack. For those who are unfamiliar with this type of instrument, a few introductory notes on the definition of Q and the measurement of Q, are included.

WHAT IS Q AND HOW IS IT MEASURED?

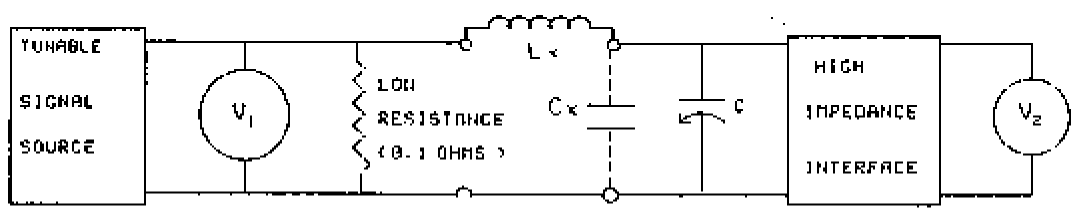
The Q factor or quality factor of an inductance is commonly expressed as the ratio of its series reactance to its series resistance. We can also express the Q factor of a capacitance as the ratio of its series reactance to its series resistance although capacitors are generally specified by the D or dissipation factor which is the reciprocal of Q.A tuned circuit, at resonance, is considered to have a Q factor. In this case, Q is equal to the ratio of either the inductive reactance, or the capacitive reactance, to the total series loss resistance in the tuned circuit. The greater the loss resistance and the lower the Q, the greater the power lost on each cycle of oscillation in the tuned circuit and hence the greater the power needed to maintain oscillation.

Another way to derive Q is as follows:

$Q = fo/\Delta f$ where fo is the resonant frequency and Δf is the 3 dB bandwidth.Sometimes we talk of loaded Q (such as in transmitter tank circuits) and, in this case, resistance for calculation of Q is the unloaded tuned circuit series resistance plus the additional loss resistance reflected in series into the circuit from its coupled load. There are other ways of expressing Q factor. It can be expressed approximately as the ratio of equivalent shunt resistance to either the inductive or the capacitive reactance. Series loss resistance can be converted to an equivalent shunt resistance using the following formula:

R(shunt) = R (series). (Q^2 + 1)

Finally, Q factor of a resonant circuit is equal to its voltage magnification factor and Q can also be expressed as the ratio of voltage developed across its reactive elements to the voltage injected in series with the circuit to produce the developed voltage. To measure Q factor, Q meters make use of this principle.A basic Q meter is shown in Figure 1. Terminals are provided to connect the inductance (Lx) to be measured and this is resonated by a variable tuning capacitor (C). Terminals are also provided to add capacitance (Cx), if required. The tuned circuit is excited from a tunable signal source which develops voltage across a resistor in series with the tuned circuit. The resistor must have a resistance small compared to the loss resistance of the components to be measured so that its value can be ignored. A resistance of a mere fraction of an ohm is necessary. Metering is provided to measure the AC injection voltage across the series resistor and the AC output voltage across the terminals of the tuning capacitor. The output measurement must be a high input impedance circuit to prevent loading of the tuned circuit by the metering circuit.



Basic Q Meter

At resonance of Lx and Cx, Q = V2/V1

*Meter V2 is Calibrated to read voltage referred to that across C.

Q is measured by adjusting the source frequency and/or the tuning capacitor for a peak in output voltage corresponding to resonance. Q factor is calculated as the ratio of output voltage measured across the tuned circuit to that injected into it. In practice, the signal source level is generally set for a calibrate point on the meter which measures injected voltage and Q is directly read from calibration on the meter which measures output voltage.

Some of the uses of Q Meter:

The Q meter can be used for many purposes. As the name implies, it can measure Q and is generally used to check the Q factor of inductors. As the internal tuning capacitor has an air dielectric its loss resistance is negligible compared to that of any inductor and hence the Q measured is that of the inductor. The value of Q varies considerable with different types of inductors used over different ranges of frequency. Miniature commercial inductors, such as the Siemens B78108 types or the Lenox-Fugal Nanored types, made on ferrite cores and operated at frequencies up to 1MHz, have typical Q factors in the region of 50 to 100. Air wound inductors with spaced turns, such as found in transmitter tank circuits and operating at frequencies above 10 MHz, can be expected to have Q factors of around 200 to 500. Some inductors have Q factors as low as five or 10 at some frequencies and such inductors are generally unsuitable for use in selective circuits or in sharp filters. The Q meter is very useful to check these out.The tuning capacitor (C) of the Q meter has a calibrated dial marked in pico-farads so that, in conjunction with the calibration of the oscillator source, the value of inductance (Lx) can be derived. The tuned circuit is simply set to resonance by adjusting the frequency and/or the tuning capacitor for a peak in the output voltage meter and then calculating the inductance (Lx) from the usual formula:

$Lx = 1/4\pi^2 f^2 C$

For L in μH, C in pF and f in MHz this reduces to: $25330/f^2C$

Another use of the Q meter is to measure the value of small capacitors. Providing the capacitor to be tested is smaller than the tuning range of the internal tuning capacitor, the test sample can be easily measured. Firstly, the capacitor sample is resonated with a selected inductor by adjusting the source frequency and using the tuning capacitor set to a low value on its calibrated scale. The sample is then disconnected and using the same frequency as before, the tuning capacitor is reset to again obtain resonance. The difference in tuning capacitor calibration read for the two tests is equal to the capacitance of the sample. Larger values of capacitance can be read by changing frequency to obtain resonance on the second test and manipulating the resonance formula. A poorly chosen inductor is not the only cause of low Q in a tuned circuit as some types of capacitor also have high loss resistance which lowers the Q. Small ceramic capacitors are often used in tuned circuits and many of these have high loss resistance, varying considerably in samples often taken from the same batch. If ceramic capacitors must be used where high Q is required, it is wise to select them for low loss resistance and the Q meter can be used for this purpose. To do this, an inductor having a high Q, of at least 200, is used to resonate the circuit, first with the tuning capacitor (C) on its own and then with individual test sample capacitors in parallel. A drastic loss in the value of Q, when the sample is added, soon shows up which capacitor should not be used.

DISTRIBUTED COIL CAPACITANCE :

Direct measurement of Q in an inductor, as discussed in previous paragraphs. is based on the circuit having two components, inductance and capacitance. Inductors also have distributed capacitance (C_d) and if this represents a significant portion of the total tuning capacitance, the Q value read will be lower than its actual value. High distributed capacitance is common in large value inductors having closely wound turns or having multiple layers.

Actual Q can be calculated from Q_e, as read, from the following:

$$Q = Q_e \, (1 + C_d/C)$$

where C_d = Distributed capacitance
and C = Tuning Capacitance

Q value error is reduced by resonating with a large value of tuning capacitance, otherwise distributed capacitance can be measured and applied to the previous formula. Two methods of measuring distributed capacitance are described in the "Boonton Q Meter Handbook". The simplest of these is said to be accurate for distributed capacitance above 10 pF and this method is described as follows:

1. With the tuning capacitor (C) set to value C1 (say 50 pF), resonate with the sample inductor by adjusting the signal source frequency.

2. Set the signal source to half the original frequency and re-resonate by adjusting C to a new value of capacitance C2.

3. Calculate distributed capacitance as follows: $C_d = (C2 - 4C1)/3$

Another effect of distributed capacitance in the inductor is to make its inductance value (as calculated from the calibration of the tuning capacitance and the calibration of the signal source) appear higher than its actual value. Again, this error can be reduced by tuning with a large value of capacitance C and/or adding C_d to C in the calculation

Question Bank

UNIT-II

BASIC ELECTRONIC MEASUREMENTS

PART-A (2 Marks)

1. **What is vector voltmeter?**

   The voltmeter which is defined as the ratio of power stored in the element to the power dissipated in the element. It is also defined as the ratio of reactance to resistance of the reactive element. It is also defined as the ratio of reactance to resistance of the reactive element.

2. **What are the main parts of CRT?**

   The main parts of CRT are 1) Electron gun, 2) Deflection system, 3) Fluorescent screen,

   4) Glass tube and 5) Base.

3. **Define deflection sensitivity of CRO.**

   The horizontal deflection x is proportional to the horizontal deflection voltage V_x applied to x input.

4. **What is fluorescence?**

The material like phosphor converts electrical energy to light energy. Thus phosphor emits light when bombarded by the electrons. This emission of light due to excitation of phosphor is called fluorescence.

5. **What is the principle of sampling oscilloscope?**

Using sampling procedure, high frequency signal is converted to the low frequency signal. Thus instead of monitoring the input signal continuously it is sampled at the regular intervals. These samples are presented on the screen in the form of dots. Such samples are merged to reconstruct the input signal. The very high frequency more than 300 MHz performance can be achieved using sampling technique used in the sampling oscilloscope.

6. **What is an electronic voltmeter?**

The voltmeter which uses rectifiers, diodes and other supporting electronic circuits to produce a current proportional to the quantity to be measured is called electronic voltmeter.

7. **Define Q factor of coil.**

The Q factor is defined as the ratio of power stored in the element to the power dissipated in the element. It also defined as the ratio of reactance to the resistance of the reactive element. Thus for a coil it is defined as $Q = X_L / R$.

8. State the advantages of chopper amplifier.

- Chopper eliminates the need for a high gain DC amplifier which has drift and stability problems.
- The input impedance is very high for direct current.
- A very high gain of the order of 10^5 to 10^6 can be achieved.

9. What is the function of focusing anodes?

The electron beam consists of many electrons and all are similarly charged. Hence the electrons repel each other. Due to such repulsive forces, the beam tends to diverge. To compensate for such repulsive forces and produce sharp beam spot, an adjustable electrostatic field is created between the two cylindrical anodes which are called focusing anodes.

10. What is the function of trigger circuit?

It is necessary that the horizontal deflection starts at the same point of the input vertical signal, each time it sweeps. Hence to synchronize horizontal deflection with vertical deflection, the trigger circuit is used.

PART-B (16 Marks)

1. Vector voltmeter. (AU APRIL 2003,2005) (16).

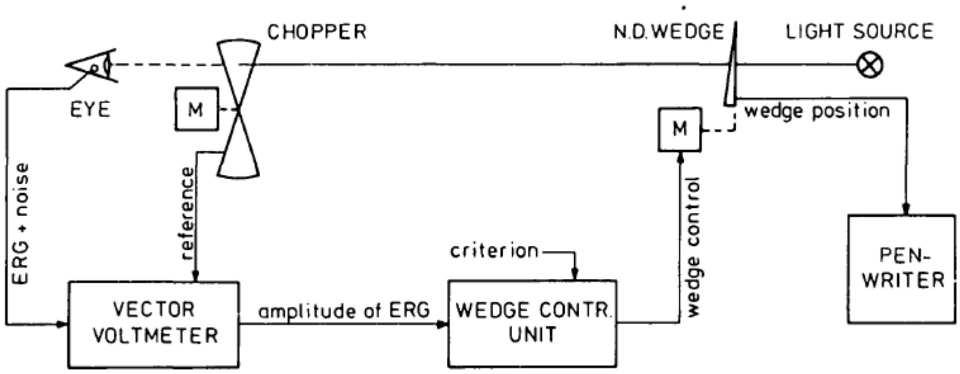
The vector voltmeter gives on line both, phase and amplitude of the ERG (electroretinogram) response to a flickering stimulus. This makes the apparatus useful in determining spectral sensitivity functions and dark adaptation curves of man and animals, in particular since it enables automatic measurement of these functions. The limitations and applications are briefly discussed.

In electroretinography (ERG) the use of the averaging computer has become standard practice for extracting the often low responses from the background electrical noise. If, besides the response magnitude the waveform of the response is of interest to the experimenter, the averaging technique indeed is a powerful tool.

There are situations however, in which the waveform of the ERG is not of primary interest, for instance for the determination of spectral sensitivity functions, or the recovery of sensitivity after exposure to a very bright light (dark adaptation curve).

The usual way to measure, e.g., spectral sensitivity, is to obtain a set of response vs. intensity curves and then, for each wavelength, to determine the intensity necessary to evoke a certain criterion response. In this particular application the averaging technique is an inefficient and time consuming way of data reduction:

One measures by hand the amplitude of the waveform from the obtained records while the waveform itself is disregarded in the final sensitivity curves. Besides being time consuming the averaging technique constitutes more problems.



Because of uncontrollable non stationarities (e.g., eye movements, blinks) the accuracy is considerably limited, as the averaged response is only available off-line after, e.g., plotting on a recorder. These drawbacks cumulate if one wants to measure dark adaptation curves, as they represent a change of sensitivity in time. Recently a so called "vector voltmeter" has become commercially available, which operates essentially with a flickering stimulus. The apparatus selectively amplifies the stimulus-locked part of the ERG, canceling at the same time the non stimulus locked background noise. In this respect its action is comparable to an averaging computer, but with the enormous advantage that it gives on line, the response magnitude as a DC (direct current) voltage.

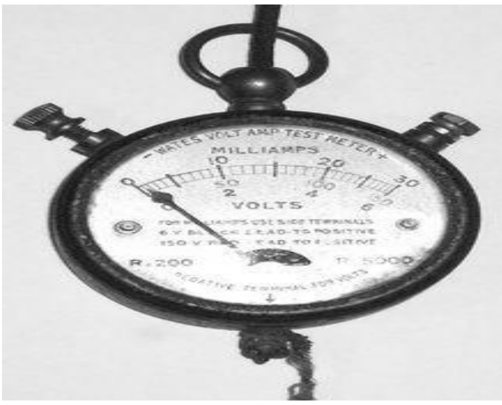
This opens the possibility of adjusting the light intensity such that a criterion response is obtained, which is analogous to what is done in psychophysical sensitivity measurements.

With an extension of the basic system even automatic measurement of the spectral sensitivity functions and the dark adaptation curve becomes possible. In this paper, a short description of the operation of the vector voltmeter in ERGs will be given, together with examples of its performance in the measurement of spectral sensitivity and dark adaptation.

2. Solid state electronic multimeter (AU APRIL2003,2005) (16).

A **multimeter** or a **multitester**, also known as a **volt/ohm meter** or **VOM**, is an electronic measuring instrument that combines several measurement functions in one unit. A typical multimeter may include features such as the ability to measure voltage, current and resistance. Multimeters may use analogor digital circuits—**analog multimeters** and **digital multimeters** (often abbreviated **DMM** or **DVOM**.) Analog instruments are usually based on amicroammeter whose pointer moves over a scale calibration for all the different measurements that can be made; digital instruments usually display digits, but may display a bar of a length proportional to the quantity measured.

A multimeter can be a hand-held device useful for basic fault finding and field service work or a bench instrument which can measure to a very high degree of accuracy. They can be used to troubleshoot electrical problems in a wide array of industrial and household devices such as electronic equipment, motor controls, domestic appliances, power supplies, and wiring systems.



The first moving-pointer current-detecting device was the galvanometer. These were used to measure resistance and voltage by using a Wheatstone bridge, and comparing the unknown quantity to a reference voltage or resistance. While useful in the lab, the devices were very slow and impractical in the field. These galvanometers were bulky and delicate.

The D'Arsonval/Weston meter movement used a fine metal spring to give proportional measurement rather than just detection, and built-in permanent field magnets made deflection independent of the 3D orientation of the meter. These features enabled dispensing with Wheatstone bridges, and made measurement quick and easy. By adding a series or shunt resistor, more than one range of voltage or current could be measured with one movement.

Multimeters were invented in the early 1920s as radio receivers and other vacuum tube electronic devices became more common.

QUANTITIES MEASURED

Contemporary multimeters can measure many quantities. The common ones are:

- Voltage, alternating and direct, in volts.
- Current, alternating and direct, in amperes. The frequency range for which AC measurements are accurate must be specified.
- Resistance in ohms.

ADDITIONALLY, SOME MULTIMETERS MEASURE:

- Capacitance in farads.
- Conductance in siemens.
- Decibels.
- Duty cycle as a percentage.
- Frequency in hertz.
- Inductance in henrys.
- Temperature in degrees Celsius or Fahrenheit, with an appropriate temperature test probe, often a thermocouple.

DIGITAL MULTIMETERS MAY ALSO INCLUDE CIRCUITS FOR:

- Continuity; beeps when a circuit conducts.
  Diodes (measuring forward drop of diode junctions, i.e., diodes and transistor junctions) and transistors (measuring current gain and other parameters).
- Battery checking for simple 1.5 volt and 9 volt batteries. This is a current loaded voltage scale. Battery checking (ignoring internal resistance, which increases as the battery is depleted), is less accurate when using a DC voltage scale.

VARIOUS SENSORS CAN BE ATTACHED TO MULTIMETERS TO TAKE MEASUREMENTS SUCH AS:

- Light level
- Acidity/Alkalinity(pH)
- Wind speed
- Relative humidity

DIGITAL

While a digital display can easily be extended in precision, the extra digits are of no value if not accompanied by care in the design and calibration of the analog portions of the multimeter. Meaningful high-resolution measurements require a good understanding of the instrument specifications, good control of the measurement conditions, and traceability of the calibration of the instrument.

ANALOG

Resolution of analog multimeters is limited by the width of the scale pointer, vibration of the pointer, the accuracy of printing of scales, zero calibration, number of ranges, and errors due to non-horizontal use of the mechanical display. Accuracy of readings obtained is also often compromised by miscounting division markings, errors in mental arithmetic, parallax observation errors, and less than perfect eyesight. Mirrored scales and larger meter movements are used to improve resolution; two and a half to three digits equivalent resolution is usual (and is usually adequate for the limited precision needed for most measurements).

3. Explain the working principle of Q meter. (AU APRIL, DEC 2006) (8)

The Q Meter has frequently been described as one of the most flexible instruments available with applications limited largely by the ingenuity of the person using it. It is our desire here to delineate some of those techniques, not normally encountered in everyday work, in the hope that wider dissemination of information gathered through many channels, will prove of some value. In order to approach our specific problems in a general way, it might be well to review some basic facts relative to the operation of the Q Meter. The Q Meter is always operated with a coil connected to its coil terminals.

If we are interested in measuring the Q of a coil, this coil will be connected to these terminals and it will be measured in one operation. If we are interested in making other measurements; (i.e., the Q of a capacitor, the impedance of a circuit, the parameters of a tuned circuit; etc.), we still need a coil, even though we are interested in that particular coil only as .a reference. This so-called “work coil“ would probably be a shielded unit to prevent stray coupling, hand-capacitance effects, etc.; and might be selected for its inductance, Q; etc., as needed for the particular application involved.

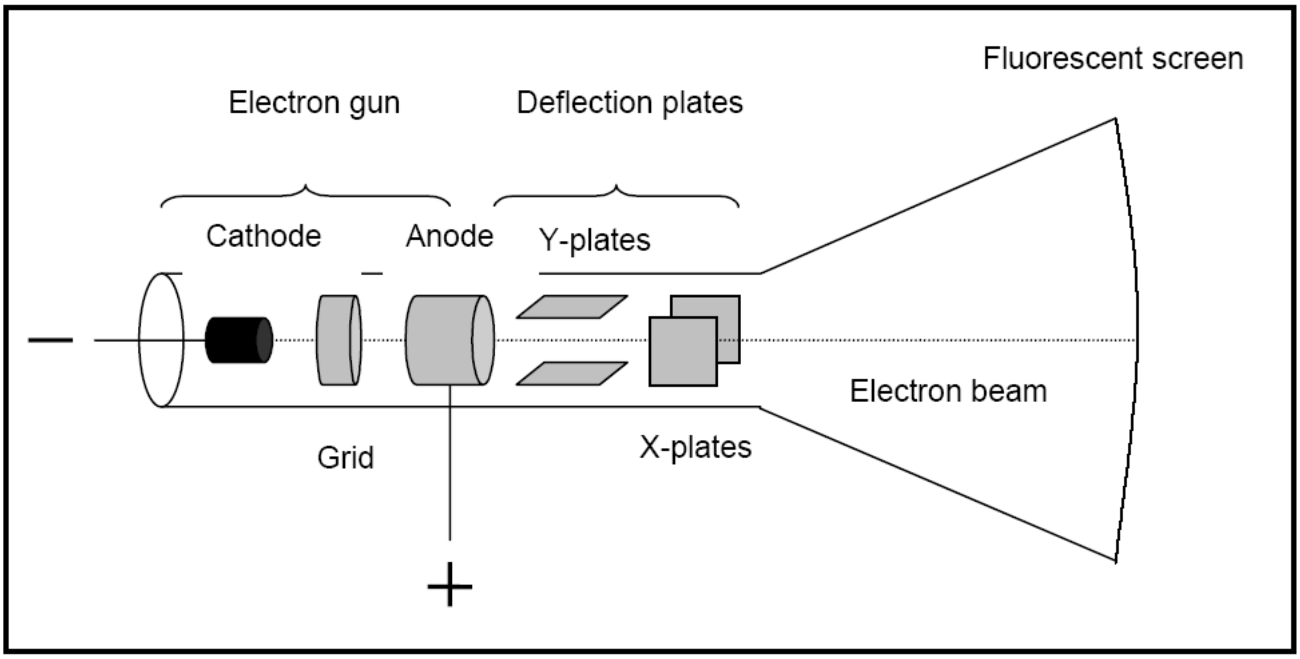
In making measurements (other than the Q of a coil} of circuit parameters there will be two steps involved. The first will be with the work coil mounted on the Q meter, where the resonating capacitance (Cl), circuit Q (Ql), and frequency will be recorded. The second will be with the unknown connected in addition to the work coil and once again the above reading will be noted, this time as C2 and *Qz.*

From this data the desired parameters can be determined using the appropriate formula selected from those shown in figure 2. High impedance circuits are measured by connecting them in *parallel* with the *Q Cupacibor;* i.e., across the “Capacitor” terminals, and using the formulas shown under the heading “Parallel Connection to Q Circuit”. If the unknown consists of more than one parameter, it should be noted that the *equivalent parallel* parameters are obtained in this manner.

Low impedance circuits are measured by connecting them in *series* with the “Low” side of the coil. In like manner the “Series Connection to Q Circuit” formulas are used to yield the *equivalent series $urdmeters* of the circuit involved. With the above in mind, it might be well to resolve some specific problems.

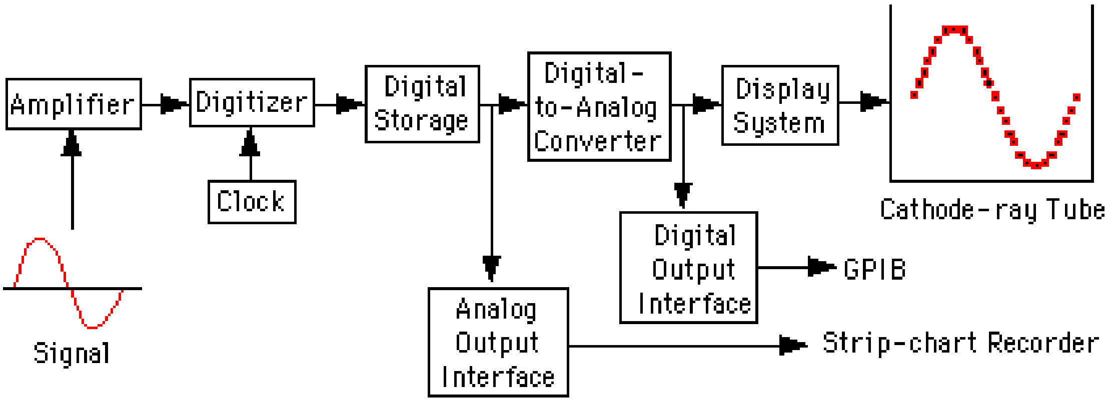
4. CRO (Cathode ray oscilloscope) (AU APRIL 2003, 2004 DEC 2005)

The device consists mainly of a vacuum tube which contains a cathode, anode, grid, X&Y-plates, and a fluorescent screen (see Figure below). When the cathode is heated (by applying a small potential difference across its terminals), it emits electrons. Having a potential difference between the cathode and the anode (electrodes), accelerate the emitted electrons towards the anode, forming an electron beam, which passes to fall on the screen.



When the fast electron beam strikes the fluorescent screen, a bright visible spot is produced. The grid, which is situated between the electrodes, controls the amount of electrons passing through it thereby controlling the intensity of the electron beam. The X&Y-plates, are responsible for deflecting the electron beam horizontally and vertically. A sweep generator is connected to the X-plates, which moves the bright spot horizontally across the screen and repeats that at a certain frequency as the source of the signal. The voltage to be studied is applied to the Y-plates. The combined sweep and Y voltages produce a graph showing the variation of voltage with time.

BLOCK DIAGRAM



TECHNICAL INFORMATION

Some technical parameters:

- Bandwidth: 0 –20 MHz to 0 –few GHz
- High input impedance
- Sensitivity: From µVcm-1to few 100 Vcm-1

Basic Classification:

- Manual
- Programmable
- Automatic
- Averaging

CLASSIFICATION

- Storage Oscilloscope
- Sampling Oscilloscope
- Digital Oscilloscope

Storage Oscilloscope:

- Special CRT which can store a waveform
- Used to capture and examine non-repetitive signals
- Used to store signals with low frequencies (10Hz)
- Highest frequency that can be recorded : 0.1MHz

Sampling Oscilloscope:

- Used in the case of repetitive waveforms
- Equivalent Sampling:
- One sample is taken from every period
- Shape of signal is acquired when displayed sequentially
- Frequency limit: 10-50GHz
- Sensitive to noise

Random Sampling:

- Samples of signal and time base are taken randomly
- Samples are displayed randomly rather in sequence
- No frequency limitation (theoretically)

Digital Oscilloscope:

- Contain memory facility for storage or precision measurement
- Uses input signal sampling, A to D conversion or DSP
- Plotters can be attached to oscilloscopes to obtain hard copies of recorded signal
- Instrumentation interfaces can be used for interconnected measurements

5. CRT (AU APRIL 2003) (8).

A **Video Display Controller** or **VDC** is an integrated circuit which is the main component in a video signal generator, a device responsible for the production of a TV video signal in a computing or game system. Some VDCs also generate a sound signal, but in that case it's not their main function. VDCs were most often used in the old home-computers of the 80s, but also in some early video game systems.

The VDC is always the main component of the video signal generator logic, but sometimes there are also other supporting chips used, such as RAM to hold the pixel data, ROM to hold character fonts, or perhaps some discrete logic such as shift registers were necessary to build a complete system. In any case, it's the VDC's responsibility to generate the timing of the necessary video signals, such as the horizontal and vertical synchronisation signals, and the blanking interval signal.

Most often the VDC chip is completely integrated in the logic of the main computer system, (its video RAM appears in the memory map of the main CPU), but sometimes it functions as a coprocessorthat can manipulate the video RAM contents independently

Video Display Controllers vs. Video Display Processors and Graphics processing units The difference between a **VDC** and the more modern Video Display Processor (**VDP**) is not that the VDCs could not generate graphics, but they did not have the special hardware accelerators to create2D and 3D images, while a typical 1990s VDP does have at least some form of hardware graphics acceleration. Also VDCs often had special hardware for the creation of "sprites", a function that in more modern VDP chips is done with the "Bit Blitter" using the "Bit blit" function.

One example of a typical Video Display Processor is the "VDP2 32-bit background and scroll plane video display processor" of the Sega Saturn. Another example is the **Advanced Graphics Architecture** (**AGA**) chip that was used for the improved graphics of the later generation Amiga computers.

This said, it is not completely clear when a "Video chip" is a "Video Display Controller" and when it is a "Video Display Processor". For example, the TMS9918 is sometimes called a "Video Display Controller" and sometimes a "Video Display Processor". In general however a "Video Display Processor" has some power to "Process" the contents of the Video RAM (filling an area of RAM for example), while a "Video Display Controller" only controls the timing of the Video synchronisation signals and the access to the Video RAM.

The Graphics processing unit (**GPU**) goes one step further than the VDP and normally also supports 3D functionality. It is the chip that is now used in modern personal computers.

Types of Video Display Controllers

Video Display controllers can be (arbitrarily) divided in several different types (here listed from simple to complex);

Video shifters, or "Video shift register based systems" (there is no generally agreed upon name for these type of devices) are the most simple type of video controllers; they are, (directly or indirectly) responsible for the video timing signals, but they normally do not access the Video RAM directly. They get the video data from the main CPU, a byte at a time, and convert it to a serial bitstream (hence the technical name "Video shifter"). This serial data stream is then used, together with the synchronisation signals, to output a (colour) video signal. The main CPU needs to do the bulk of the work. Normally these chips only support a very low resolution Raster graphics mode.

A **CRTC**, or Cathode Ray Tube Controller, generates the video timings and reads video data from a RAM attached to the CRTC, to output it via an external character generator ROM, (for text modes) or directly, (for high resolution graphics modes) to the video output shift register. Because the actual capabilities of the video generator depend to a large degree on the external logic, video generator based on a CRTC chip can have a wide range of capabilities. From very simple (text mode only) systems to very high resolution systems supporting a wide range of colours. Sprites however are normally not supported by these systems.

Video interface controllers are much more complex than CRT controllers, and the external circuitry that is needed with a CRTC is embedded in the video controller chip. Sprites are often supported, as are (RAM based) character generators and video RAM dedicated to colour attributes and pallette registers (Color lookup tables) for the high-resolution and/or text-modes.

Video coprocessors have their own internal CPU dedicated to reading (and writing) their own video RAM, and converting the contents of this video RAM to a video signal. The main CPU can give commands to the coprocessor, for example to change the video modes or to manipulate the video ram contents. The video coprocessor also controls the (most often RAM based) character generator, the colour attribute RAM, Palette registers and the Spite logic (as long as these exist of course).

UNIT III - SIGNAL GENERATORS AND ANALYZERS

FUNCTION GENERATOR:

A function generator is a device which produces simple repetitive waveforms. Such devices contain an electronic oscillator, a circuit that is capable of creating a repetitive waveform. (Modern devices may use digital signal processing to synthesize waveforms, followed by a digital to analog converter, or DAC, to produce an analog output). The most common waveform is a sine wave, but sawtooth, step (pulse), square,and triangular waveform oscillators are commonly available as are arbitrary waveform generators (AWGs). If the oscillator operates above the audio frequency range (>20 kHz), the generator will often include some sort of modulation function such as amplitude modulation (AM), frequency modulation (FM), or phase modulation (PM) as well as a second oscillator that provides an audio frequency modulation waveform.

Function generators are typically used in simple electronics repair and design; where they are used to stimulate a circuit under test. A device such as an oscilloscope is then used to measure the circuit's output. Function generators vary in the number of outputs they feature, frequency range, frequency accuracy and stability, and several other parameters.A function generator is a piece of electronic test equipment or software used to generate electrical waveforms. These waveforms can be either repetitive or single-shot, in which case some kind of triggering source is required (internal or external).Function Generators are used in development, testing and repair of electronic equipment, e.g. as a signal source to test amplifiers, or to introduce an error signal into a control loop.

Explanation

Analog function generators usually generate a triangle waveform as the basis for all of its other outputs. The triangle is generated by repeatedly charging and discharging a capacitor from a constant current source. This produces a linearly ascending or descending voltage ramp. As the output voltage reaches upper and lower limits, the charging and discharging is reversed using a comparator, producing the linear triangle wave. By varying the current and the size of the capacitor, different frequencies may be obtained. Sawtooth waves can be produced by charging the capacitor slowly, using a current, but using a diode over the current source to discharge quickly - the polarity of the diode changes the polarity of the resulting sawtooth, i.e. slow rise and fast fall, or fast rise and slow fall. A 50% duty cycle square wave is easily obtained by noting whether the capacitor is being charged or discharged, which is reflected in the current switching comparator's output. Other duty cycles (theoretically from 0% to 100%) can be obtained by using a comparator and the sawtooth or triangle signal. Most function generators also contain a non-linear diode shaping circuit that can convert the triangle wave into a reasonably accurate sine wave. It does so by rounding off the hard corners of the triangle wave in a process similar to clipping in audio systems. A typical function generator can provide frequencies up to 20 MHz. RF generators for higher frequencies are not function generators in the strict sense since typically produce pure or modulated sine signals only. Function generators, like most signal generators, may also contain an attenuator, various means of modulating the output waveform, and often the ability to automatically and repetitively "sweep" the frequency of the output waveform (by means of a voltage- controlled oscillator) between two operator-determined limits. This capability makes it very easy to evaluate the frequency response of a given electronic circuit.Some function generators can also generate white or pink noise.More advanced function generators use Direct Digital Synthesis (DDS) to generate waveforms. Arbitrary waveform generators use DDS to generate any waveform that can be described by a table of amplitudes.

Signal generator :

A signal generator, also known variously as function generator, pitch generator, arbitrary waveform generator, digital pattern generator or frequency generator is an electronic device that generates repeating or non-repeating electronic signals (in either the analog or digital domains). They are generally used in designing, testing, troubleshooting, and repairing electronic or electroacoustic devices; though they often have artistic uses as well. There are many different types of signal generators, with different purposes and applications (and at varying levels of expense); in general, no device is suitable for all possible applications. Traditionally, signal generators have been embedded hardware units, but since the age of multimedia-PCs, flexible, programmable software tone generators have also been available.

Basic Sweep Generator

A basic system for the sweep generator is shown in figure 1. A low-frequency sawtooth wave is generated from some form of oscillator or waveform generator. The instantaneous voltage of the sawtooth wave controls the frequency of an RF oscillator with its centre frequency set at the centre frequency of the device under test (filter or IF channel etc). Over a single sweep of frequency, RF output voltage from the device, as a function of time, is a plot of the filter response. By rectifying and RF filtering in a simple AM detector, the output is converted to a DC voltage varying as a function of time and this voltage is applied to the vertical input of the CRO. By synchronising the sweep of the CRO with the sawtooth output, the device response is plotted on the CRO screen.

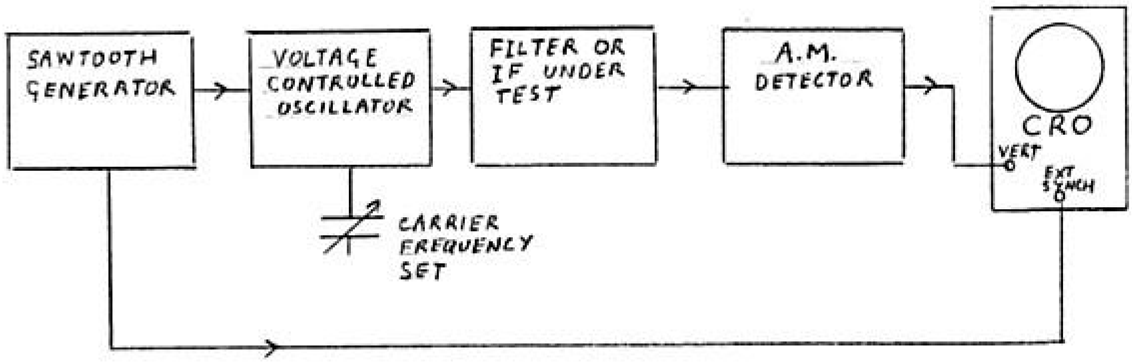
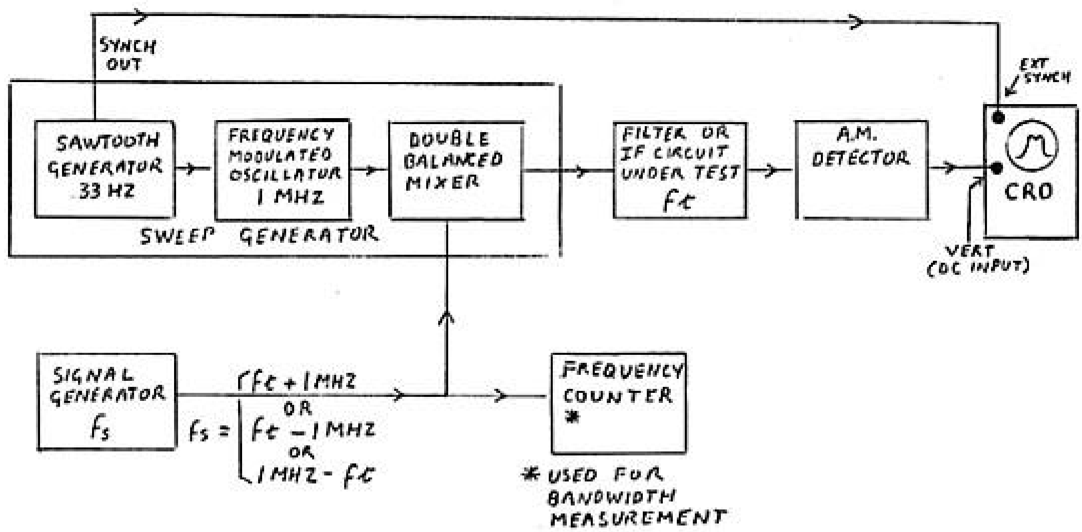


Figure 1 - Basic Sweep Generator arrangement

To achieve this for a range of frequencies, it is easiest to sweep a single frequency (say 1MHz) and heterodyne this to the test frequency required. The system developed is shown in the block diagram, figure 2. A 1MHz oscillator is frequency modulated by the output of a sawtooth generator operating at 33 Hz. The modulated output is beat with an external signal generator set to provide the difference frequency centered at the center frequency of the filter or IF circuit under test. The output of circuit under test is fed to a simple AM detector which provides varying DC output level to fed the CRO vertical input. By synchronising the CRO sweep circuit to the 33 Hz sweep generator, a plot of test circuit response is displayed in terms of amplitude verses frequency



Total Harmonic Distortion (THD) Analyzers:

It calculates the total distortion introduced by all the harmonics of the fundamental frequency wave. In most cases THD is the amount required to be calculated, rather than distortion caused by individual harmonics. This type of analysis is very important in systems (e.g. Audio) in which filters with extremely small passband/ stopband are desired, such as a notch filter in a parametric equalizer.

Block Diagram of a THD Analyzer

This is a specific type of THD analyzer, in which basically the fundamental frequency of the input wave is suppressed so as to remove it from the spectra of the meters used for distortion measurement, and the total gain of all the harmonics, is calculated, thus obtaining the total distortion caused by the harmonics.

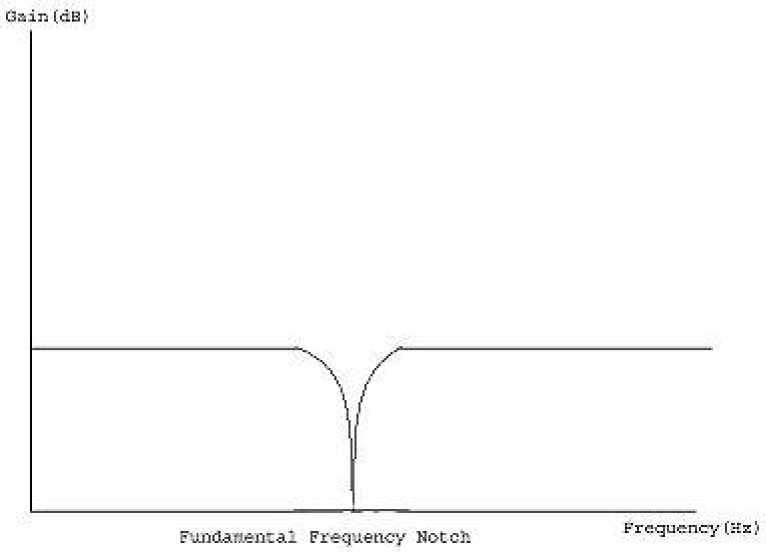


Fig.The frequency response of a Fundamental Suppression Analyzer

A block diagram of a Fundamental Suppression Analyzer is shown in Fig.1. This basic construction consists of three main sections: Input section with impedance matcher, a rejection amplifier section and an output metering circuit. Notice the feedback from the bridge amplifier to the pre-amp section, that enables the rejection circuit to work more accurately.

Working :

The applied input wave is impedance matched with the rejection circuit with the help of an attenuator and an impedance matcher. This signal is then applied to a pre-amplifier which raises the signal level to a desired value. The following section consists of a Wien bridge. The bridge is tuned to the fundamental frequency by frequency control and it is balanced for zero output by adjusting the bridge controls, thus giving a notch in the frequency response of the rejection section. After the Wien Bridge, a bridge amplifier follows that simply amplifies low harmonic voltage levels to measurable higher levels. A feedback loop is formed from Bridge Amp o/p to the Pre-Amp i/p thus eliminating even the slightest effect of fundamental frequency. This filtered output is then applied to a meter amplifier which can be an instrumentation amplifier. This amp raises the voltage levels to the compatibility of the meter scale/digital meter which follows. Thus the total voltage obtained at the meter output shows the amount of distortion present in the wave due to harmonics of fundamental.A spectrum analyzer or spectral analyzer is a device used to examine the spectral composition of some electrical, acoustic, or optical waveform. It may also measure the power spectrum.

Types :

There are analog and digital spectrum analyzers:

- An analog spectrum analyzer uses either a variable band-pass filter whose mid-frequency is automatically tuned (shifted, swept) through the range of frequencies of which the spectrum is to be measured or a superheterodyne receiver where the local oscillator is swept through a range of frequencies.
  - A digital spectrum analyzer computes the discrete Fourier transform (DFT), a mathematical process that transforms a waveform into the components of its frequency spectrum.

Some spectrum analyzers (such as "real-time spectrum analyzers") use a hybrid technique where the incoming signal is first down-converted to a lower frequency using superheterodyne techniques and then analyzed using fast fourier transformation (FFT) techniques. Typical functionality: Allows one to fix the window of frequencies to visualize and center the display on a chosen frequency. Controls the position and function of markers and indicates the value of power. Several spectrum analyzers have a "Marker Delta" function that can be used to measure Signal to Noise Ratio or Bandwidth.

Bandwidth/average

Is a filter of resolution. The spectrum analyzer captures the measure on having displaced a filter of small bandwidth along the window of frequencies.

Amplitude

The maximum value of a signal at a point is called amplitude. A spectrum analyzer that implements amplitude analysis is called a Pulse height analyzer.Manages parameters of measurement. It stores the maximum values in each frequency and a solved measurement to compare it.

Superheterodyne spectrum

analyzer: Operation

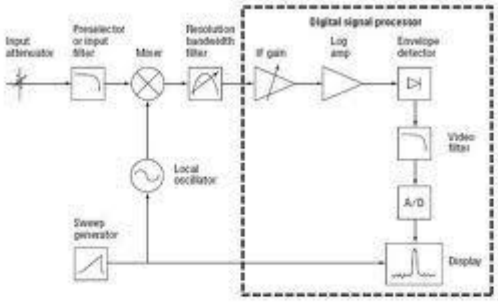


Figure 2. Block diagram of a superheterodyne spectrum analyzer

Usually, a spectrum analyzer displays a power spectrum over a given frequency range, changing the display as the properties of the signal change. There is a trade-off between how quickly the display can be updated and the frequency resolution, which is for example relevant for distinguishing frequency components that are close together. With a digital spectrum analyzer, the frequency resolution is $\Delta\nu = 1 / T$, the inverse of the time T over which the waveform is measured and Fourier transformed (according to Uncertainty principle). With an analog spectrum analyzer, it is dependent on the bandwidth setting of the bandpass filter. However, an analog spectrum analyzer will not produce meaningful results if the filter bandwidth (in Hz) is smaller than the square root of the sweep speed (in Hz/s)[citation needed], which means that an analog spectrum analyzer can never beat a digital one in terms of frequency resolution for a given acquisition time. Choosing a wider bandpass filter will improve the signal-to-noise ratio at the expense of a decreased frequency resolution.

With Fourier transform analysis in a digital spectrum analyzer, it is necessary to sample the input signal with a sampling frequency ν_s that is at least twice the highest frequency that is present in the signal, due to the Nyquist limit. A Fourier transform will then produce a spectrum containing all frequencies from zero to $\nu_s / 2$. This can place considerable demands on the required analog-to-digital converter and processing power for the Fourier transform. Often, one is only interested in a narrow frequency range, for example between 88 and 108 MHz, which would require at least a sampling frequency of 216 MHz, not counting the low-pass anti-aliasing filter. In such cases, it can be more economic to first use a superheterodyne receiver to transform the signal to a lower range, such as 8 to 28 MHz, and then sample the signal at 56 MHz. This is how an analog-digital-hybrid spectrum analyzer works. For use with very weak signals, a pre-amplifier can be used, although harmonic and intermodulationdistortion may lead to the creation of new frequency components that were not present in the original signal. A new method, without using a high local oscillator (LO) (that usually produces a high-frequency signal close to the signal) is used on the latest analyzer generation like Aaronia´s Spectran series. The advantage of this new method is a very low noise floor near the physical thermal noise limit of -174 dBm/Hz.A digital voltmeter typically consists of an analog to digital converter (A/D) with a digital display. The analog signal is converted into a digital code proportionate to the magnitude of the signal. Voltages from picovolts to megavolts are measurable, though the scale usually graduates in millivolts, volts, or kilovolts. Frequencies between zero and several megahertz may also be measured.

DVMs measure both alternating current (AC) and direct current (DC) in electronics. Common laboratory and commercial applications involve electromechanical machinery with a current flowing through wires and circuits. Often, a digital voltmeter is used to monitor a unit, such as a generator. Portable or handheld devices, such as the digital multimeter (DMM), for example, may combine several functions into one instrument measuring voltage, current, and resistance. This is the preferred tool of an electrician. Many DVMs integrate outputs for monitoring, controlling, transmitting, and printing of data. Advanced systems are often connected to computers, allowing for automation, optimization of processes, and prevention of malfunctions and critical failure safeties. Chemical plants can convert measurements to voltage, and control and monitor temperature, pressure, level, or flow. Medical equipment, such as x-ray machines, may use a digital voltmeter to make sure the voltage of the equipment is in the proper range.

Question Bank

UNIT-III

SIGNAL GENERATORS AND ANALYZERS

PART-A (2 Marks)

1. **What are the general requirements of signal generator? (AU APRIL 2007)**
   - The output frequency of signal generator should be very stable.
   - The amplitude of output signal of signal generator should be controllable from low values to relatively large values.
   - The amplitude of output signal must be stable.
   - The harmonic contents in the output should be as low as possible.
   - The output signal should be distortion free.
2. **What is an Oscillator?**

The oscillator uses an active device such as an operational amplifier. The output of an operational amplifier is fed back in phase with input. This positive feedback causes regenerative action resulting an oscillation.

3. **What is a function generator? (AU APRIL 2007)**

The function generator is an instrument which generates different types of waveforms. The frequency of these waveforms can be varied over wide range. The most required common waveforms like sine, square, saw tooth, triangular and DC pulses.

4. **What is sweep frequency generator? (AU NOV 2004)**

Sweep frequency generator provides a sinusoidal output voltage whose frequency varies smoothly and continuously over an entire frequency band. The process of frequency modulation may be accomplished electronically or mechanically.

5. **Give short notes on wave Analyzer. (AU APRIL 2010)**

A wave analyzer is an instrument designed to measure relative amplitudes of single frequency components in a complex waveform. It can analysis of waveforms includes the determination of amplitude, frequency and phase angle of the harmonic components.

6. **What are the two types of wave analyzer?**
   - Frequency selective wave analyzer.
   - Heterodyne wave analyzer.
7. **List few applications of wave analyzer**
   - To measure the harmonic distortion of an amplifier
   - To carry out complete harmonic analysis
   - To measure the signal energy with the well defined bandwidth.
8. **What is meant by harmonic distortion? (AU APRIL 2005,2006)**

The distortion caused due to the nonlinear behavior of the circuit elements is called harmonic distortion.

9. Give few useful applications of spectrum analyzer (AU APRIL 2006)

- Modulation measurement.
- Continuous wave signal frequency stability.
- Harmonic distortion measurement.
- Noise measurement.
- Examining pulse modulation.

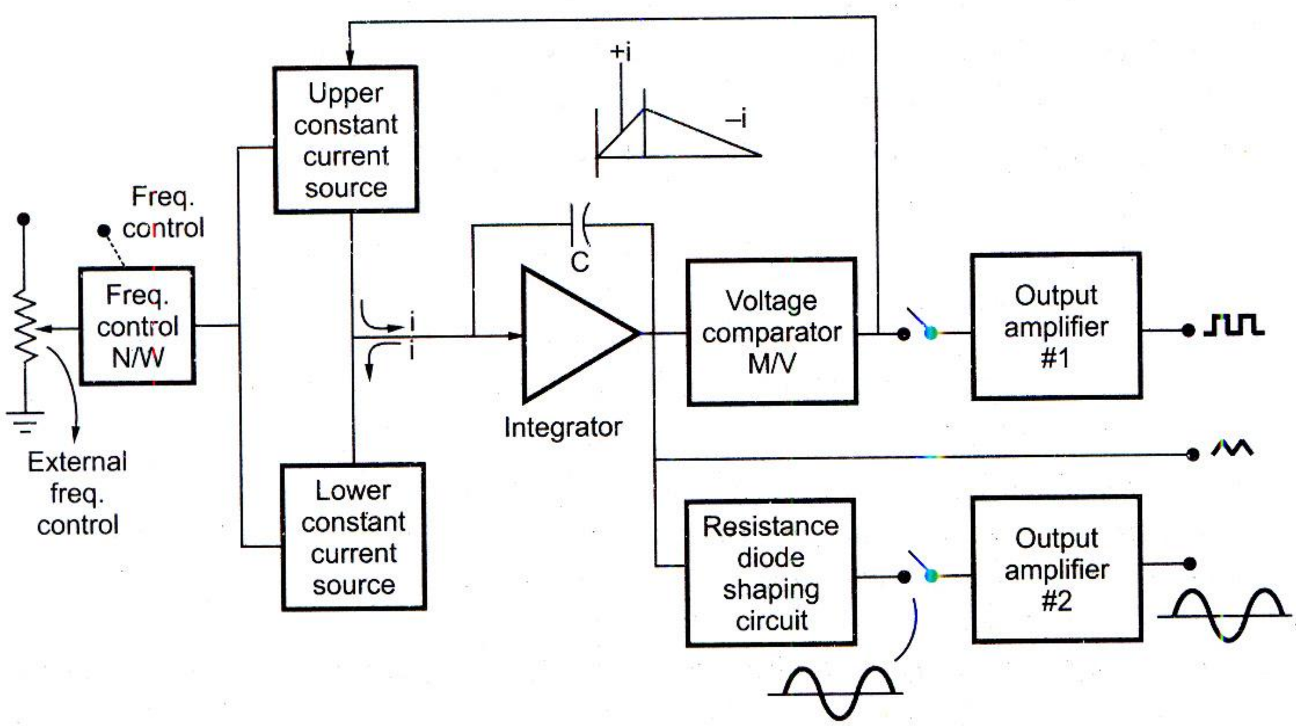
PART-B (16 Marks)

1. Describe the working of function generator with the block diagram. (AU APRIL, NOV 2004,2005) (16)

The function generator is an instrument which generates different types of waveforms. The frequency of these waveforms can be varied over wide range. The most required common waveforms like sine, square, saw tooth, triangular and DC pulses.

The function generator can be phase locked to a standard frequency of the source. Then all the output waveforms of the generator will have same accuracy and stability as that of standard source.

Block diagram



Frequency controlled voltage

It is used to regulate two current sources namely upper current source and lower current source.

Upper and lower current source

The upper current source supplies constant current to an integrator. The lower current source supplies opposite current to the integrator.

Integrator

The output voltage of integrator then increases linearly with time. Hence this controls frequency. The output of the integrator has triangular waveform. The frequency of this triangular waveform is determined by the magnitudes of the currents supplied by upper current source and lower current source.

Voltage comparator or multivibartor

This circuit changes the state of the network when the output voltage integrator equals the maximum predetermined upper level. To get square wave the output of the integrator is passed through comparator. The voltage comparator delivers square wave output voltage of same frequency as that of input triangular.

Diode resistance network

The sine wave is derived from triangular wave. The triangular wave is synthesized into sine wave using diode resistance network. In this shaper circuit the slope of triangular wave is changed as its amplitude changes. This results in a sine wave with less than 1 % distortion.

The two output amplifiers provide two simultaneous, individually selected outputs of any of the waveform functions.

The function of the signal generator is to supply signals of known amplitude and known frequency.

Features of function generators

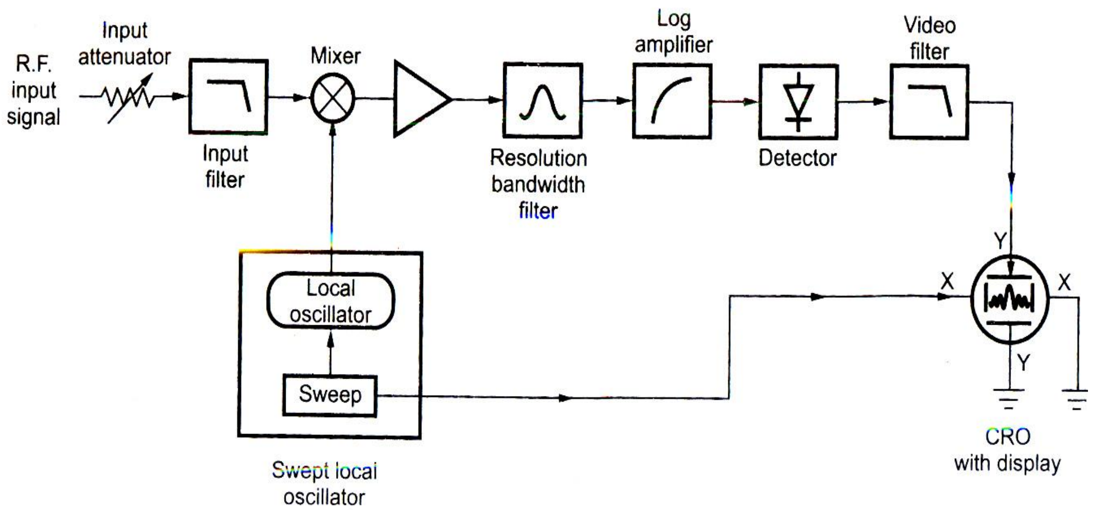
- The frequency range is 0.01Hz to 100 KHz.
- Accuracy $\pm$ 1 %
- Can be phase locked to another external signal source
- Can produce various waveforms.

2. Describe the working of spectrum analyzer with a block diagram. Explain the various applications of the spectrum analyzer. (AU APRIL 2007, NOV 2005, 2008) (16)

Based on the technique used, the spectrum analyzers can be classified as scanning type and non-scanning type. The scanning type analyzers use swept technique, while the non-scanning type is called real time spectrum analyzers.

Let us discuss the basic spectrum analyzer using swept technique. This analyzer uses a swept receiver of superhetrodyne type hence this analyzer is also called swept superhetrodyne spectrum analyzer.

Block diagram



The basic blocks of the swept superhetrodyne spectrum analyzer are

I. Wideband input mixer
II. Swept local oscillator driving wideband mixer
III. Resolution bandwidth filter, deciding intermediate frequency
IV. Detector and video filter.
V. Display

Input attenuator

The attenuator decides the level of the input signals so as to keep it within the operating range of other blocks of the instrument. Generally spectrum analyzer can handle 0 to 10 dBm.

Input filter

The input filter is used to reject unwanted signals. It suppresses the spurious signals. This is necessary because mixer responds to both sums and differences of frequencies. Mostly the filter is low pass filter.

Wideband input mixer

It multiplies the input signal from filter and the local oscillator signal. It provides two signals at the output which are proportional in amplitude to the input signal but having frequencies which are sum difference of frequencies of the input signal and the local oscillator signal.

Intermediate frequency (I.F) section

This is most important stage in analyzer where real analysis takes place. The stage function is to provide a wide selection of resolution bandwidth filters. These filters are described by their 3-dB bandwidth. These filters decide the resolving power of the analyzers.

Log amplifier

It processes incoming signal in a logarithmic fashion. The logarithmic processing allows a large range of incoming signals to be measured and compared

Detector

The detector used in the analyzer is called linear envelope detector. This is exactly similar to the detectors used in A.M. radios. The detector receives a signal from log amplifier which is compressed one. This is somewhat releases the large linear range requirement of a detector.

Video filter

These filters are used for post filtering or averaging the detector output. The bandwidth setting of video filter is same or larger than resolution bandwidth filter. If the signal is along with noise, averaging is necessary. Averaging removes the random noise and pure signal remains.

Display

The output of the video filter is given to CRO for display purpose.

Swept local oscillator

The swept local oscillator puts a limit on the stability and spectral purity in many performance areas.

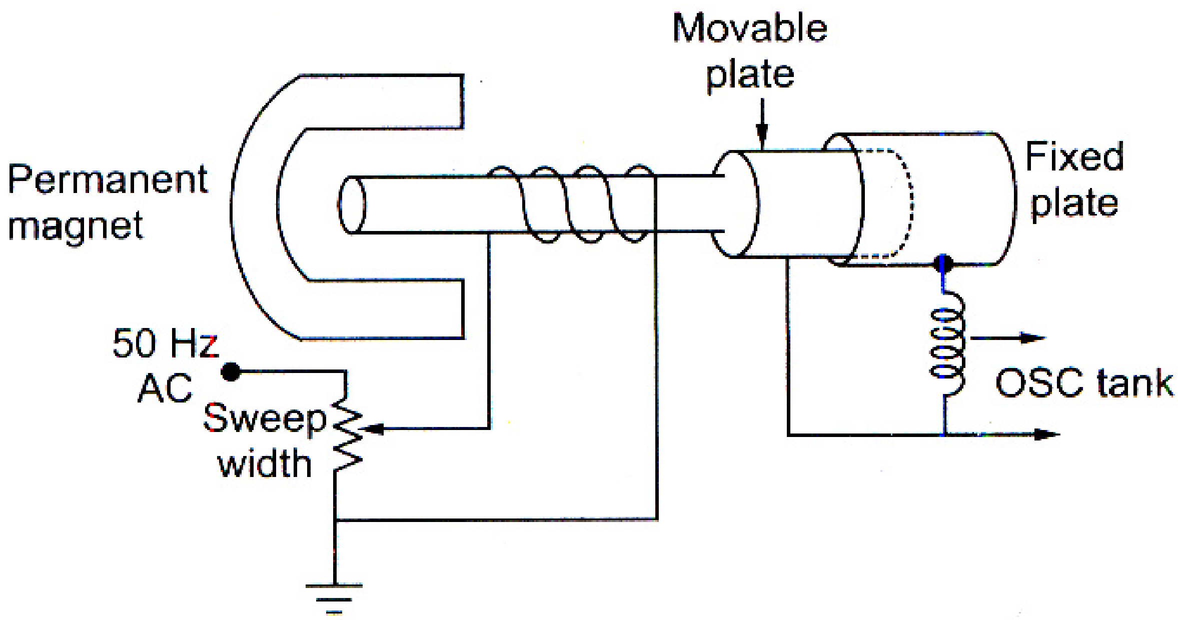
Applications

- Modulation measurement.

- Continuous wave signal frequency stability.
- Harmonic distortion measurement.
- Noise measurement.
- Examining pulse modulation.

3. Describe the working of a sweep frequency generator. (AU APRIL 2004, 2010, NOV 2003,2006) (16)

The sine wave generator discussed in earlier sections generates output voltage at a known and stable frequency. But in some applications such as measuring frequency response of amplifiers, filters and other networks, a variable frequency source is used. In such cases sweep frequency generators are used.



In the early days, the method for varying frequency electronically was not invented. Some other methods were used to get variable frequency source. Reactance tube modulator used was providing very little frequency variation, so most of the times, electro-mechanical systems such as motor driven capacitors were used.

The sweep generator is very much simple signal generator. In the simple signal generator, an oscillator is tuned to fixed single frequency. In the sweep generator, an oscillator is electronically tuned and by using voltage controlled oscillator variable frequency is obtained. As name indicates, a sweep voltage generator provides voltage, known as control voltage, to the voltage controlled oscillator (VCO).

The function of voltage controlled oscillator is to provide various frequency sweeps according to voltage provide by sweep voltage generator. But the relationship between sweep voltage and frequency is nonlinear. To obtain linearity, a compensation circuit is provided between sweep frequency voltage and oscillator tuning voltage. The compensation circuit is called linearizing circuit.

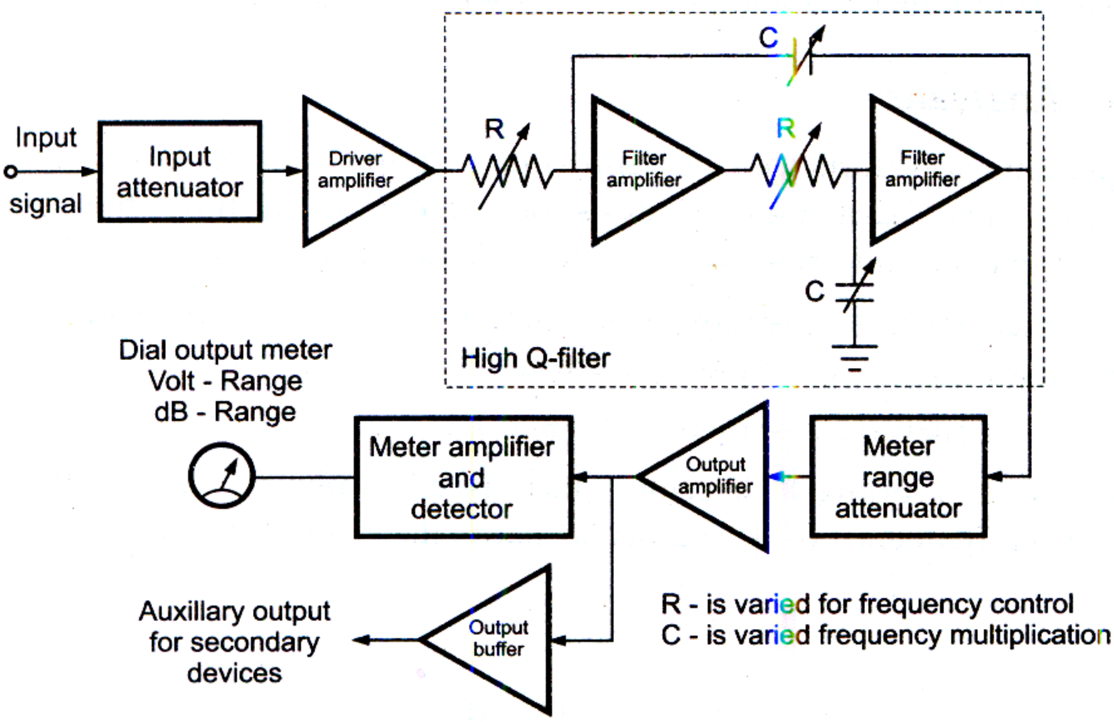
4. Give the principle of wave analyzer with the help of suitable diagrams. (AU APRIL, NOV 2009) (16)

A wave analyzer is an instrument designed to measure relative amplitudes of single frequency components in a complex waveform. It can analysis of waveforms includes the determination of amplitude, frequency and phase angle of the harmonic components.

Types of wave analyzer

- Frequency selective wave analyzer.
- Heterodyne wave analyzer.

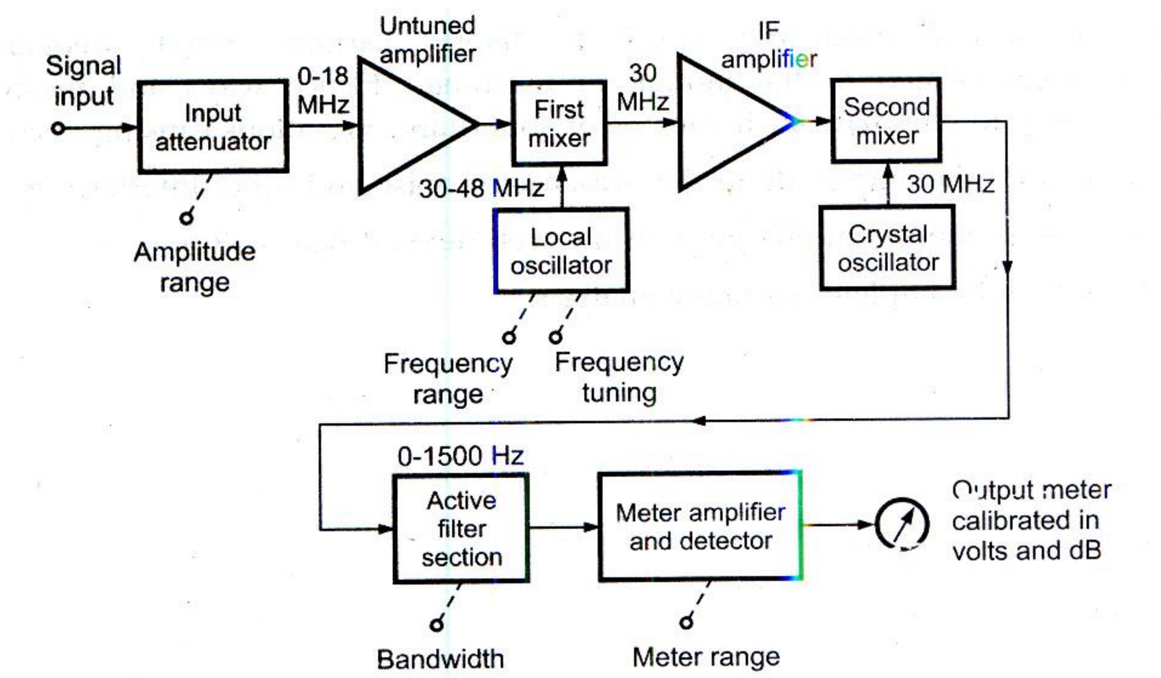
Frequency selective wave analyzer



- The waveform to be analyzed is passed through an adjustable attenuator. This acts as a range multiplier.
- The driver amplifier feeds the waveform to a high Q filter.
- This filter consists of cascade arrangement of RC resonant sections and filter amplifiers.
- The capacitors are used for range changing.
- The potentiometer is used to change frequency within the selected pass band.

- The entire AF range is covered in decade steps by the switching capacitors in the RC section
- The final amplifier stage supplies the selected signal to the meter circuit and to an untuned buffer amplifier.
- The function of buffer amplifier is to drive the output devices, such as the recorders, electronic counters etc.
- The analyzer input must have low input distortion.
- The meter has several voltage ranges as well as decibel scale marked on it. It is driven by an average reading rectifier type detector.

Heterodyne wave analyzer



- This is RF range analyzer works on the principle of mixing i.e. heterodyning.
- In this type of wave analyzer the input signal is heterodyned to a higher intermediate frequency (IF) by an internal local oscillator.
- Tuning the local oscillator shifts the various signal frequency components into the pass band of the IF amplifier.
- The output of the IF amplifier is then rectified and applied to the metering circuit.
- The input is applied first to the attenuator section. This gives the output frequency in the range of 0 to 18 MHz
- The untuned amplifier amplifies this signal and gives it to the first mixer.

- The first mixer heterodynes the input with the frequency from local oscillator. This oscillator has frequency range 30-48 MHz
- The output of the first mixer difference frequency of 30 MHz
- The IF amplifier amplifies this signal and gives it to the second mixer.
- The second mixer heterodynes the signal with 30 MHz frequency crystal oscillator.
- Thus at the output of second mixer the zero difference frequency is obtained.
- The active filter having controlled bandwidth and symmetrical slopes of 72 dB per octave, then passes the selected component to the meter amplifier and detector.
- The output from the meter detector is then used to obtain final indication on the output meter which is having a decibel calibrated scale.
- The output from detector may be applied to a recording device.

Applications of wave analyzer

- ➢ To measure the harmonic distortion of an amplifier
- ➢ To carry out complete harmonic analysis
- ➢ To measure the signal energy with the well defined bandwidth.

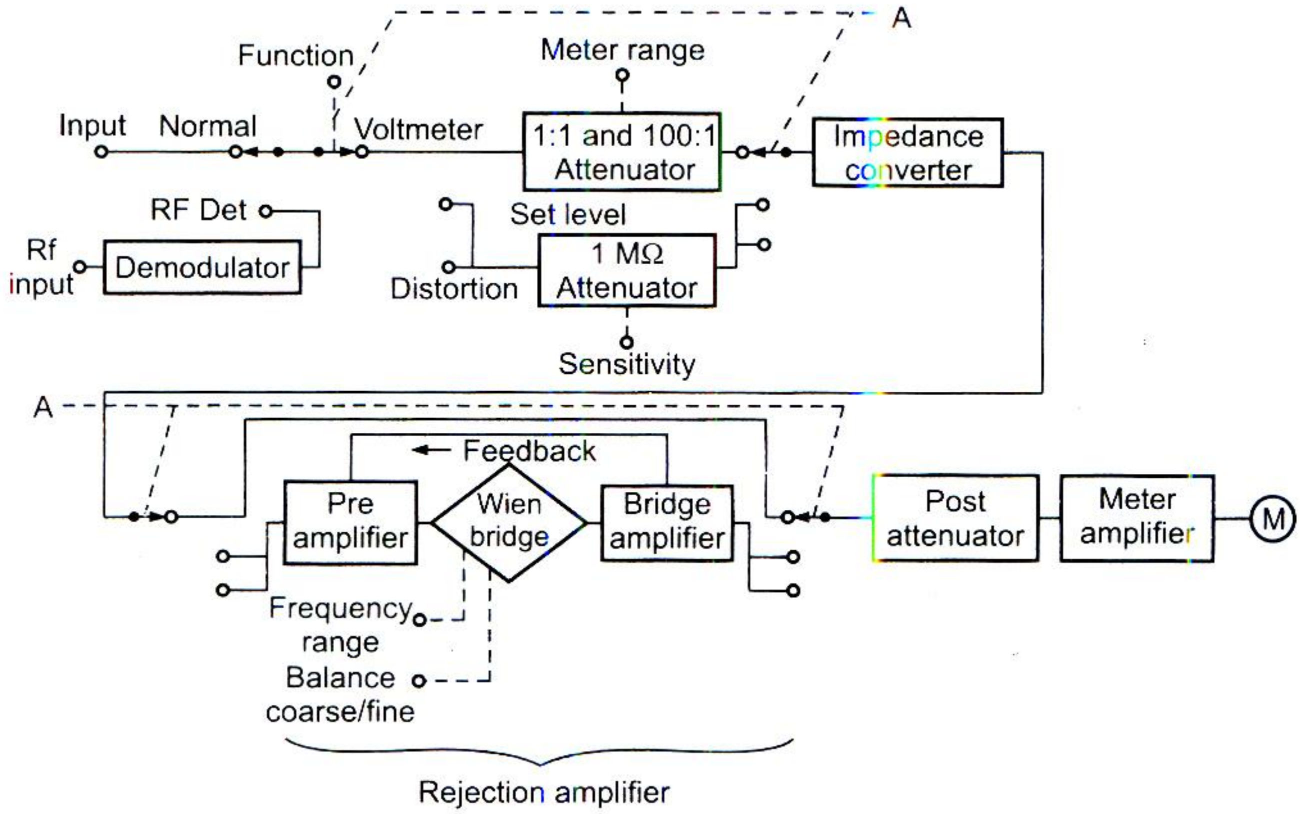
5. Explain the distortion analyzer with the help of suitable diagrams. (AU APRIL 2006, NOV 2007) (16)

The application of purely sinusoidal input signal to an amplifier should result in purely sinusoidal signal at the output. But practically output waveform is not exact replica of the input. This because of presence of various types of distortions. Such distortions due to the inherent nature of amplifier or nonlinear characteristics of various components used.

The distortion caused due to the nonlinear behavior of the circuit elements is called harmonic distortion.

Suppression Distortion Analyzer

This is used to measure the distortion factor (T.H.D) rather than the contribution by each component. In this analyzer, the input is applied to such a network that suppresses or rejects the fundamental component but passes all the harmonic frequency components for the measurement.



The analyzer consists of four major sections

I. Impedance converter
II. Rejection amplifier
III. Metering circuit
IV. Power supply.

- The **impedance converter provides** a low noise, high impedance input circuit.
- The **rejection amplifier** rejects the fundamental frequency,
- The **metering** circuit measures the harmonic distortion present which provides visual indication of the T.H.D in terms of a percentage of the total input voltage.
- The **power supply** provides the supply for proper circuit operation.

There are two modes of operation:

Voltmeter mode

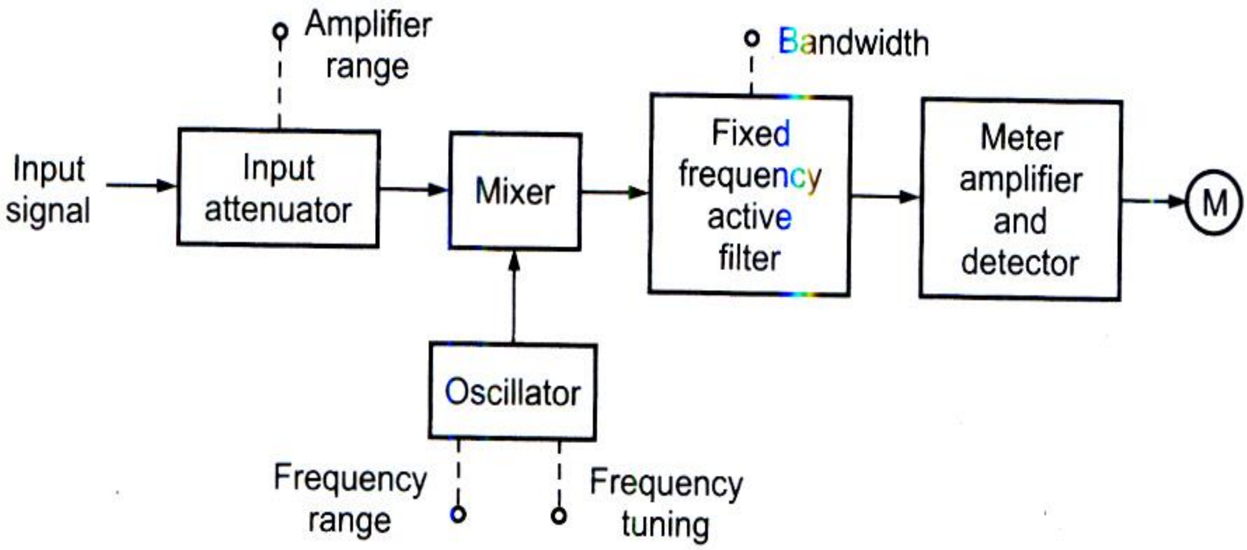
In this, it acts as a normal AC voltmeter. The input is applied through 1: 1 and 100: 1 attenuator which selects proper meter range. The impedance converter bypasses the rejection amplifier as shown dotted. The meter measures the RMS value of the AC input voltage.

Distortion mode

In this, output of the impedance converter is applied to the rejection amplifier. But before applying to the impedance converter, now the input signal is applied to 1 MΩ input attenuator which provides the 50 dB attenuation in 10 dB steps. This is controlled by a front panel control named sensitivity. Due to low noise high impedance provided by impedance converter, accurate measurement is possible.

- The rejection amplifier circuit consists of a pre-amplifier, a wien bridge and a bridge amplifier.
- The pre-amplifier provides further amplification at extremely low distortion levels.
- The bridge is connected as an interstate coupling between the pre-amplifier and the bridge amplifier.
- By the front panel, the bridge is tuned and balanced, no output results due to balancing. Hence fundamental frequency component is rejected.
- For other frequencies, wien bridge provides varying output which is amplified by bridge amplifier.
- The output is then given to meter circuit through post attenuator.

Heterodyne Harmonic Distortion Analyzer



- The variable frequency oscillator output is mixed with each harmonic of the input signal, with the help of balanced mixer, either the sum or difference frequency is made equal to the frequency of the filter.
- The quartz crystal type highly selective filters can be used as each harmonic frequency is converted to a constant frequency.
- This allows selecting constant frequency signal related to a particular harmonic and passing it to the metering circuit.
- The balanced mixer consists of a balanced modulator and it eliminates original frequency of the harmonic. Generation of low harmonic distortion is the advantage of the balanced modulator.

- In some cases, the meter reading is calibrated directly in terms of voltage while in some cases the harmonics are compared with a reference voltage, which is representation of the fundamental component.
- As the calibration in terms of voltage is the feature of direct reading heterodyne harmonic distortion analyzer, they are also called frequency voltmeters.
- These instruments are also called carrier frequency voltmeters and selective level voltmeters.

UNIT IV- DIGITAL INSTRUMENTS

Comparison of analog and digital techniques :

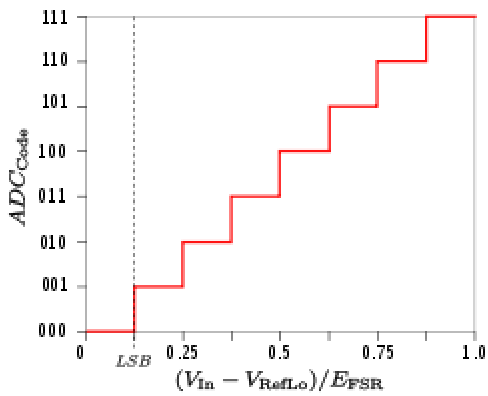
An analog-to-digital converter (abbreviated ADC, A/D or A to D) is a device that converts a continuous quantity to a discrete digital number. The reverse operation is performed by a digital-to-analog converter (DAC).

Typically, an ADC is an electronic device that converts an input analog voltage (or current) to a digital number proportional to the magnitude of the voltage or current. However, some non-electronic or only partially electronic devices, such as rotary encoders, can also be considered ADCs.

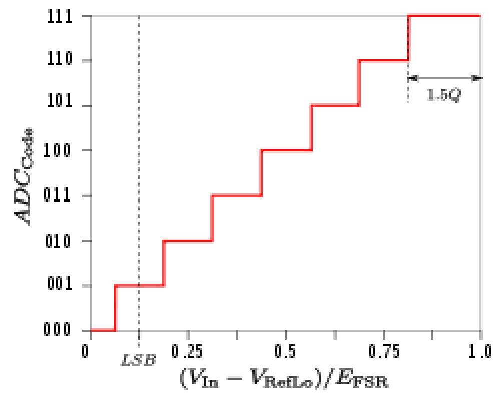
The digital output may use different coding schemes. Typically the digital output will be a two's complement binary number that is proportional to the input, but there are other possibilities. An encoder, for example, might output a Gray code.

An ADC might be used to make an isolated measurement. ADCs are also used to quantize time-varying signals by turning them into a sequence of digital samples. The result is quantized in both time and value.

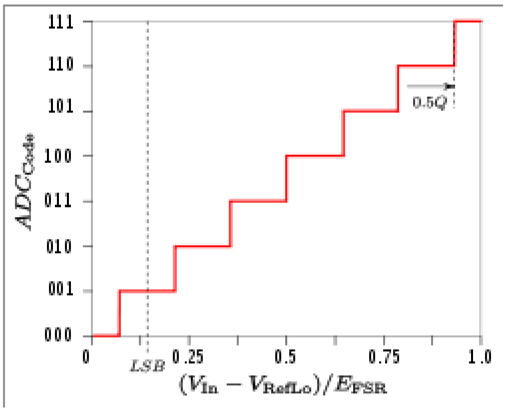
Resolution :



An 8-level ADC coding scheme.



An 8-level ADC coding scheme. As in figure 1 but with mid-tread coding.



An 8-level ADC mid-tread coding scheme. As in figure 2 but with equal half-LSB intervals at the highest and lowest codes. Note that LSB is now slightly larger than in figures 1 and 2.The resolution of the converter indicates the number of discrete values it can produce over the range of analog values. The values are usually stored electronically in binary form, so the resolution is usually expressed in bits. In consequence, the number of discrete values available, or "levels", is usually a power of two. For example, an ADC with a resolution of 8 bits can encode an analog input to one in 256 different levels, since 28 = 256. The values can represent the ranges from 0 to 255 (i.e. unsigned integer) or from -128 to 127 (i.e. signed integer), depending on the application.Resolution can also be defined electrically, and expressed in volts. The minimum change in voltage required to guarantee a change in the output code level is called the LSB (least significant bit, since this is the voltage represented by a change in the LSB). The resolution Q of the ADC is equal to the LSB voltage. The voltage resolution of an ADC is equal to its overall voltage measurement range divided by the number of discrete voltage intervals:

$$Q = \frac{E_{\mathrm{FSR}}}{N}$$

where:

N is the number of voltage intervals,

EFSR is the full scale voltage range, given by,

$$E_{\mathrm{FSR}} = V_{\mathrm{RefHi}} - V_{\mathrm{RefLow}}$$

the upper and lower extremes respectively of the voltages that can be coded.

Normally, the number of voltage intervals is given by,

$$N = 2^M$$

where

M is the ADC's resolution in bits.

That is, one voltage interval is assigned per code level. However, figure 3 shows a situation where

$$N = 2^M - 1$$

Some examples:

- Example 1
  - o Coding scheme as in figure 1
  - o Full scale measurement range = 0 to 10 volts
  - o ADC resolution is 12 bits: 212 = 4096 quantization levels (codes)
  - o ADC voltage resolution, Q = (10V - 0V) / 4096 = 10V / 4096 ≈0.00244 V ≈2.44 mV.
- Example 2
  - o Coding scheme as in figure 2
  - o Full scale measurement range = -10 to +10 volts
  - o ADC resolution is 14 bits: 214 = 16384 quantization levels (codes)
  - o ADC voltage resolution is, Q = (10V - (-10V)) / 16384 = 20V / 16384 ≈ 0.00122 V ≈1.22 mV.
- Example 3
  - o Coding scheme as in figure 3
  - o Full scale measurement range = 0 to 7 volts
  - o ADC resolution is 3 bits: 23 = 8 quantization levels (codes)
  - o ADC voltage resolution is, Q = (7 V − 0 V)/7 = 7 V/7 = 1 V = 1000 mV

In most ADCs, the smallest output code ("0" in an unsigned system) represents a voltage range which is 0.5Q, that is, half the ADC voltage resolution (Q). The largest code represents a range of 1.5Q as in figure 2 (if this were 0.5Q also, the result would be as figure 3). The other N − 2 codes are all equal in width and represent the ADC voltage resolution (Q) calculated above. Doing this centers the code on an input voltage that represents the M th division of the input voltage range. This practice is called "mid-tread" operation. This type of ADC can be modeled mathematically as:

$$ADC_{\text{Code}} = \text{round}\left(\left(\frac{2^M}{V_{\text{RefHi}} - V_{\text{RefLow}}}\right) \cdot (V_{\text{In}} - V_{\text{RefLow}})\right)$$

The exception to this convention seems to be the Microchip PIC processor, where all M steps are equal width, as shown in figure 1. This practice is called "Mid-Rise with Offset" operation.

$$ADC_{\text{Code}} = \text{floor}\left(\left(\frac{2^M}{V_{\text{RefHi}} - V_{\text{RefLow}}}\right) \cdot (V_{\text{In}} - V_{\text{RefLow}})\right)$$

In practice, the useful resolution of a converter is limited by the best signal-to-noise ratio (SNR) that can be achieved for a digitized signal. An ADC can resolve a signal to only a certain number of bits of resolution, called the effective number of bits (ENOB). One effective bit of resolution changes the signal-to-noise ratio of the digitized signal by 6 dB, if the resolution is limited by the ADC. If a preamplifier has been used prior to A/D conversion, the noise introduced by the amplifier can be an important contributing factor towards the overall SNR.

Linear ADCs

Most ADCs are of a type known as linear[1] The term linear as used here means that the range of the input values that map to each output value has a linear relationship with the output value, i.e., that the output value k is used for the range of input values from

m(k + b) to m(k + 1 + b)

where m and b are constants. Here b is typically 0 or −0.5. When b = 0, the ADC is referred to as mid-rise, and when b = −0.5 it is referred to as mid-tread.

Non-linear ADCs

If the probability density function of a signal being digitized is uniform, then the signal-to-noise ratio relative to the quantization noise is the best possible. Because this is often not the case, it is usual to pass the signal through its cumulative distribution function (CDF) before the quantization. This is good because the regions that are more important get quantized with a better resolution. In the dequantization process, the inverse CDF is needed.This is the same principle behind the companders used in some tape-recorders and other communication systems, and is related to entropy maximization.For example, a voice signal has a Laplacian

distribution. This means that the region around the lowest levels, near 0, carries more information than the regions with higher amplitudes. Because of this, logarithmic ADCs are very common in voice communication systems to increase the dynamic range of the representable values while retaining fine-granular fidelity in the low-amplitude region.
An eight-bit A-law or the μ-law logarithmic ADC covers the wide dynamic range and has a high resolution in the critical low-amplitude region, that would otherwise require a 12-bit linear ADC.

Accuracy

An ADC has several sources of errors. Quantization error and (assuming the ADC is intended to be linear) non-linearity is intrinsic to any analog-to-digital conversion. There is also a so-called aperture error which is due to a clock jitter and is revealed when digitizing a time-variant signal (not a constant value). These errors are measured in a unit called the LSB, which is an abbreviation for least significant bit. In the above example of an eight-bit ADC, an error of one LSB is 1/256 of the full signal range, or about 0.4%.

Quantization error

Quantization error is due to the finite resolution of the ADC, and is an unavoidable imperfection in all types of ADC. The magnitude of the quantization error at the sampling instant is between zero and half of one LSB.In the general case, the original signal is much larger than one LSB. When this happens, the quantization error is not correlated with the signal, and has a uniform distribution. Its RMS value is the standard deviation of this distribution, given by

$$\frac{1}{\sqrt{12}}\mathrm{LSB} \approx 0.289\,\mathrm{LSB}.$$

In the eight-bit ADC example, this represents 0.113% of the full signal range.

At lower levels the quantizing error becomes dependent of the input signal, resulting in distortion. This distortion is created after the anti-aliasing filter, and if these distortions are above 1/2 the sample rate they will alias back into the audio band. In order to make the quantizing error independent of the input signal, noise with an amplitude of 2 least significant bits is added to the signal. This slightly reduces signal to noise ratio, but, ideally, completely eliminates the distortion. It is known as dither.

Non-linearity :

All ADCs suffer from non-linearity errors caused by their physical imperfections, causing their output to deviate from a linear function (or some other function, in the case of a deliberately non-linear ADC) of their input. These errors can sometimes be mitigated by calibration, or prevented by testing.Important parameters for linearity are integral non-linearity (INL) and differential non- linearity (DNL). These non-linearities reduce the dynamic range of the signals that can be digitized by the ADC, also reducing the effective resolution of the ADC.

Aperture error :

Imagine that we are digitizing a sine wave x(t) = Asin(2πf0t). Provided that the actual sampling time uncertainty due to the clock jitter is Δt, the error caused by this phenomenon can be estimated as $E_{ap} \leq |x'(t)\Delta t| \leq 2A\pi f_0 \Delta t$.

The error is zero for DC, small at low frequencies, but significant when high frequencies have high amplitudes. This effect can be ignored if it is drowned out by the quantizing error.

Jitter requirements can be calculated using the following formula: $\Delta t < \frac{1}{2^q \pi f_0}$, where q is a number of ADC bits.

ADC resolution in bit	Input frequency						
	1 Hz	44.1 kHz	192 kHz	1 MHz	10 MHz	100 MHz	1 GHz
8	1243 µs	28.2 ns	6.48 ns	1.24 ns	124 ps	12.4 ps	1.24 ps
10	311 µs	7.05 ns	1.62 ns	311 ps	31.1 ps	3.11 ps	0.31 ps
12	77.7 µs	1.76 ns	405 ps	77.7 ps	7.77 ps	0.78 ps	0.08 ps

14	19.4 µs	441 ps	101 ps	19.4 ps	1.94 ps	0.19 ps	0.02 ps
16	4.86 µs	110 ps	25.3 ps	4.86 ps	0.49 ps	0.05 ps	–
18	1.21 µs	27.5 ps	6.32 ps	1.21 ps	0.12 ps	–	–
20	304 ns	6.88 ps	1.58 ps	0.16 ps	–	–	–
24	19.0 ns	0.43 ps	0.10 ps	–	–	–	–
32	74.1 ps	–	–	–	–	–	–

This table shows, for example, that it is not worth using a precise 24-bit ADC for sound recording if there is not an ultra low jitter clock. One should consider taking this phenomenon into account before choosing an ADC.Clock jitter is caused by phase noise.[2][3] The resolution of ADCs with a digitization bandwidth between 1 MHz and 1 GHz is limited by jitter.When sampling audio signals at 44.1 kHz, the anti-aliasing filter should have eliminated all frequencies above 22 kHz. The input frequency (in this case, 22 kHz), not the ADC clock frequency, is the determining factor with respect to jitter performance.

Sampling rate :

The analog signal is continuous in time and it is necessary to convert this to a flow of digital values. It is therefore required to define the rate at which new digital values are sampled from the analog signal. The rate of new values is called the sampling rate or sampling frequency of the converter. A continuously varying band limited signal can be sampled (that is, the signal values at intervals of time T, the sampling time, are measured and stored) and then the original signal can be exactly reproduced from the discrete-time values by an interpolation formula. The accuracy is limited by quantization error. However, this faithful reproduction is only possible if the sampling rate is higher than twice the highest frequency of the signal. This is essentially what is embodied in theorem. Since a practical ADC cannot make an instantaneous conversion, the input value must necessarily be held constant during the time that the converter performs a conversion (called the conversion time). An input circuit called a sample and hold performs this task—in most cases by using a capacitor to store the analog voltage at the input, and using an electronic switch or gate to disconnect the capacitor from the input. Many ADC integrated circuits include the sample and hold subsystem internally.

Aliasing :

All ADCs work by sampling their input at discrete intervals of time. Their output is therefore an incomplete picture of the behaviour of the input. There is no way of knowing, by looking at the output, what the input was doing between one sampling instant and the next. If the input is known to be changing slowly compared to the sampling rate, then it can be assumed that the value of the signal between two sample instants was somewhere between the two sampled values. If, however, the input signal is changing rapidly compared to the sample rate, then this assumption is not valid.If the digital values produced by the ADC are, at some later stage in the system, converted back to analog values by a digital to analog converter or DAC, it is desirable that the output of the DAC be a faithful representation of the original signal. If the input signal is changing much faster than the sample rate, then this will not be the case, and spurious signals called aliases will be produced at the output of the DAC. The frequency of the aliased signal is the difference between the signal frequency and the sampling rate. For example, a 2 kHz sine wave being sampled at 1.5 kHz would be reconstructed as a 500 Hz sine wave. This problem is called aliasing.To avoid aliasing, the input to an ADC must be low-pass filtered to remove frequencies above half the sampling rate. This filter is called an anti-aliasing filter, and is essential for a practical ADC system that is applied to analog signals with higher frequency content.Although aliasing in most systems is unwanted, it should also be noted that it can be exploited to provide simultaneous down-mixing of a band-limited high frequency signal (see undersampling and frequency mixer).

In A-to-D converters, performance can usually be improved using dither. This is a very small amount of random noise (white noise) which is added to the input before conversion. Its amplitude is set to be twice the value of the least significant bit. Its effect is to cause the state of the LSB to randomly oscillate between 0 and 1 in the presence of very low levels of input, rather than sticking at a fixed value. Rather than the signal simply getting cut off altogether at this low level (which is only being quantized to a resolution of 1 bit), it extends the effective range of signals that the A-to-D converter can convert, at the expense of a slight increase in noise - effectively the quantization error is diffused across a series of noise values which is far less objectionable than a hard cutoff. The result is an accurate representation of the signal over time. A suitable filter at the output of the system can thus recover this small signal variation.An audio signal of very low level (with respect to the bit depth of the ADC) sampled without dither sounds extremely distorted and unpleasant. Without dither the low level may cause the least significant bit to "stick" at 0 or 1. With dithering, the true level of the audio may be calculated by averaging the actual quantized sample with a series of other samples [the dither] that are recorded over time.

A virtually identical process, also called dither or dithering, is often used when quantizing photographic images to a fewer number of bits per pixel—the image becomes noisier but to the eye looks far more realistic than the quantized image, which otherwise becomes banded. This analogous process may help to visualize the effect of dither on an analogue audio signal that is converted to digital.Dithering is also used in integrating systems such as electricity meters. Since the values are added together, the dithering produces results that are more exact than the LSB of the analog-to-digital converter.
Note that dither can only increase the resolution of a sampler, it cannot improve the linearity, and thus accuracy does not necessarily improve.

Oversampling

Usually, signals are sampled at the minimum rate required, for economy, with the result that the quantization noise introduced is white noise spread over the whole pass band of the converter. If a signal is sampled at a rate much higher than the Nyquist frequency and then digitally filtered to limit it to the signal bandwidth then there are three main advantages:

- digital filters can have better properties (sharper rolloff, phase) than analogue filters, so a sharper anti-aliasing filter can be realized and then the signal can be down sampled giving a better result
- a 20-bit ADC can be made to act as a 24-bit ADC with 256× oversampling
- the signal-to-noise ratio due to quantization noise will be higher than if the whole available band had been used. With this technique, it is possible to obtain an effective resolution larger than that provided by the converter alone
- The improvement in SNR is 3 dB (equivalent to 0.5 bits) per octave of oversampling which is not sufficient for many applications. Therefore, oversampling is usually coupled with noise shaping (see sigma-delta modulators). With noise shaping, the improvement is 6L+3 dB per octave where L is the order of loop filter used for noise shaping. e.g. - a 2nd order loop filter will provide an improvement of 15 dB/octave.

Relative speed and precision

The speed of an ADC varies by type. The Wilkinson ADC is limited by the clock rate which is processable by current digital circuits. Currently, frequencies up to 300 MHz are possible. The conversion time is directly proportional to the number of channels. For a successive approximation ADC, the conversion time scales with the logarithm of the number of channels. Thus for a large number of channels, it is possible that the successive approximation ADC is faster than the Wilkinson. However, the time consuming steps in the Wilkinson are digital, while those in the successive approximation are analog. Since analog is inherently slower than digital, as the number of channels increases, the time required also increases. Thus there are competing processes at work. Flash ADCs are certainly the fastest type of the three. The conversion is basically performed in a single parallel step. For an 8-bit unit, conversion takes place in a few tens of nanoseconds.There is, as expected, somewhat of a trade off between speed and precision. Flash ADCs have drifts and uncertainties associated with the comparator levels, which lead to poor uniformity in channel width. Flash ADCs have a resulting poor linearity. For successive approximation ADCs, poor linearity is also apparent, but less so than for flash ADCs. Here, non-linearity arises from accumulating errors from the subtraction processes. Wilkinson ADCs are the best of the three. These have the best differential non-linearity. The other types require channel smoothing in order to achieve the level of the Wilkinson.

The sliding scale principle

The sliding scale or randomizing method can be employed to greatly improve the channel width uniformity and differential linearity of any type of ADC, but especially flash and successive approximation ADCs. Under normal conditions, a pulse of particular amplitude is always converted to a certain channel number. The problem lies in that channels are not always of uniform width, and the differential linearity decreases proportionally with the divergence from the average width. The sliding scale principle uses an averaging effect to overcome this phenomenon. A random, but known analog voltage is added to the input pulse. It is then converted to digital form, and the equivalent digital version is subtracted, thus restoring it to its original value. The advantage is that the conversion has taken place at a random point. The statistical distribution of the final channel numbers is decided by a weighted average over a region of the range of the ADC. This in turn desensitizes it to the width of any given channel.

ADC structures

These are the most common ways of implementing an electronic ADC:

- A direct conversion ADC or flash ADC has a bank of comparators sampling the input signal in parallel, each firing for their decoded voltage range. The comparator bank feeds a logic circuit that generates a code for each voltage range. Direct conversion is very fast, capable of gigahertz sampling rates, but usually has only 8 bits of resolution or fewer, since the number of comparators needed, 2N - 1, doubles with each additional bit, requiring a large expensive circuit. ADCs of this type have a large die size, a high input capacitance, high power dissipation, and are prone to produce glitches on the output (by outputting an out-of-sequence code). Scaling to newer submicrometre technologies does not help as the device mismatch is the dominant design limitation. They are often used for video, wideband communications or other fast signals in optical storage.

- A successive-approximation ADC uses a comparator to reject ranges of voltages, eventually settling on a final voltage range. Successive approximation works by constantly comparing the input voltage to the output of an internal digital to analog converter (DAC, fed by the current value of the approximation) until the best approximation is achieved. At each step in this process, a binary value of the approximation is stored in a successive approximation register (SAR). The SAR uses a reference voltage (which is the largest signal the ADC is to convert) for comparisons. For example if the input voltage is 60 V and the reference voltage is 100 V, in the 1st clock cycle, 60 V is compared to 50 V (the reference, divided by two. This is the voltage at the output of the internal DAC when the input is a '1' followed by zeros), and the voltage from the comparator is positive (or '1') (because 60 V is greater than 50 V). At this point the first binary digit (MSB) is set to a '1'. In the 2nd clock cycle the input voltage is compared to 75 V (being halfway between

100 and 50 V: This is the output of the internal DAC when its input is '11' followed by zeros) because 60 V is less than 75 V, the comparator output is now negative (or '0'). The second binary digit is therefore set to a '0'. In the 3rd clock cycle, the input voltage is compared with 62.5 V (halfway between 50 V and 75 V: This is the output of the internal DAC when its input is '101' followed by zeros). The output of the comparator is negative or '0' (because 60 V is less than 62.5 V) so the third binary digit is set to a 0. The fourth clock cycle similarly results in the fourth digit being a '1' (60 V is greater than 56.25 V, the DAC output for '1001' followed by zeros). The result of this would be in the binary form 1001. This is also called bit-weighting conversion, and is similar to a binary search. The analogue value is rounded to the nearest binary value below, meaning this converter type is mid-rise (see above). Because the approximations are successive (not simultaneous), the conversion takes one clock-cycle for each bit of resolution desired. The clock frequency must be equal to the sampling frequency multiplied by the number of bits of resolution desired. For example, to sample audio at 44.1 kHz with 32 bit resolution, a clock frequency of over 1.4 MHz would be required. ADCs of this type have good resolutions and quite wide ranges. They are more complex than some other designs. A ramp-compare ADC produces a saw-tooth signal that ramps up or down then quickly returns to zero. When the ramp starts, a timer starts counting. When the ramp voltage matches the input, a comparator fires, and the timer's value is recorded. Timed ramp converters require the least number of transistors.The ramp time is sensitive to temperature because the circuit generating the ramp is often just some simple oscillator. There are two solutions: use a clocked counter driving a DAC and then use the comparator to preserve the counter's value, or calibrate the timed ramp.

A special advantage of the ramp-compare system is that comparing a second signal just requires another comparator, and another register to store the voltage value. A very simple (non-linear) ramp-converter can be implemented with a microcontroller and one resistor and capacitor [10]. Vice versa, a filled capacitor can be taken from an integrator, time-to-amplitude converter, phase detector, sample and hold circuit, or peak and hold circuit and discharged. This has the advantage that a slow comparator cannot be disturbed by fast input changes.

- An integrating ADC (also dual-slope or multi-slope ADC) applies the unknown input voltage to the input of an integrator and allows the voltage to ramp for a fixed time period (the run-up period). 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 (the run-down period). 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. Likewise, the speed
- of the converter can be improved by sacrificing resolution. Converters of this type (or variations on the concept) are used in most digital voltmeters for their linearity and flexibility.

- A delta-encoded ADC or Counter-ramp has an up-down counter that feeds a digital to analog converter(DAC). The input signal and the DAC both go to a comparator. The comparator controls the counter. The circuit uses negative feedback from the comparator to adjust the counter until the DAC's output is close enough to the input signal. The number is read from the counter. Delta converters have very wide ranges, and high resolution, but the conversion time is dependent on the input signal level, though it will always have a guaranteed worst-case. Delta converters are often very good choices to read real-world signals. Most signals from physical systems do not change abruptly. Some converters combine the delta and successive approximation approaches; this works especially well when high frequencies are known to be small in magnitude.

- A pipeline ADC (also called subranging quantizer) uses two or more steps of subranging. First, a coarse conversion is done. In a second step, the difference to the input signal is determined with a digital to analog converter (DAC). This difference is then converted finer, and the results are combined in a last step. This can be considered a refinement of the successive approximation ADC wherein the feedback reference signal consists of the interim conversion of a whole range of bits (for example, four bits) rather than just the next-most-significant bit. By combining the merits of the successive approximation and flash ADCs this type is fast, has a high resolution, and only requires a small die size.

- A Sigma-Delta ADC (also known as a Delta-Sigma ADC) oversamples the desired signal by a large factor and filters the desired signal band. Generally, a smaller number of bits than required are converted using a Flash ADC after the filter. The resulting signal, along with the error generated by the discrete levels of the Flash, is fed back and subtracted from the input to the filter. This negative feedback has the effect of noise shaping the error due to the Flash so that it does not appear in the desired signal frequencies. A digital filter (decimation filter) follows the ADC which reduces the sampling rate, filters off unwanted noise signal and increases the resolution of the output (sigma-delta modulation,also called delta-sigma modulation).

- A Time-interleaved ADC uses M parallel ADCs where each ADC sample data every M:th cycle of the effective sample clock. The result is that the sample rate is increased M times compared to what each individual ADC can manage. In practice, the individual differences between the M ADCs degrade the overall performance reducing the SFDR. However, technologies exist to correct for these time-interleaving mismatch errors.

- An ADC with intermediate FM stage first uses a voltage-to-frequency converter to converts the desired signal into an oscillating signal with a frequency proportional to the voltage of the desired signal, and then uses a frequency counter to convert that frequency into a digital count proportional to the desired signal voltage. Longer integration times allow for higher resolutions. Likewise, the speed of the converter can be improved by sacrificing resolution. The two parts of the ADC may be widely separated, with the frequency signal passed through a opto-isolator or transmitted wirelessly. Some such ADCs use sine wave or square wave frequency modulation; others use pulse-frequency modulation. Such ADCs were once the most popular way to show a digital display of the status of a remote analog sensor.

There can be other ADCs that use a combination of electronics and other technologies:

- A Time-stretch analog-to-digital converter (TS-ADC) digitizes a very wide bandwidth analog signal, that cannot be digitized by a conventional electronic ADC, by time-stretching the signal prior to digitization. It commonly uses a photonic preprocessor frontend to time-stretch the signal, which effectively slows the signal down in time and compresses its bandwidth. As a result, an electronic backend ADC, that would have been too slow to capture the original signal, can now capture this slowed down signal. For continuous capture of the signal, the frontend also divides the signal into multiple segments in addition to time-stretching. Each segment is individually digitized by a separate electronic ADC. Finally, a digital signal processor rearranges the samples and removes any distortions added by the frontend to yield the binary data that is the digital representation of the original analog signal.

Commercial analog-to-digital converters

These are usually integrated circuits.Most converters sample with 6 to 24 bits of resolution, and produce fewer than 1 megasample per second. Thermal noise generated by passive components such as resistors masks the measurement when higher resolution is desired. For audio applications and in room temperatures, such noise is usually a little less than 1 μV (microvolt) of white noise. If the Most Significant Bit corresponds to a standard 2 volts of output signal, this translates to a noise-limited performance that is less than 20~21 bits, and obviates the need for any dithering. Mega- and gigasample per second converters are available, though (Feb 2002). Megasample converters are required in digital video cameras, video capture cards, and TV tuner cards to convert full-speed analog video to digital video files. Commercial converters usually have ±0.5 to ±1.5 LSB error in their output.In many cases the most expensive part of an integrated circuit is the pins, because they make the package larger, and each pin has to be connected to the integrated circuit's silicon. To save pins, it is common for slow ADCs to send their data one bit at a time over a serial interface to the computer, with the next bit coming out when a clock signal changes state, say from zero to 5V. This saves quite a few pins on the ADC package, and in many cases, does not make the overall design any more complex (even microprocessors which use memory-mapped I/O only need a few bits of a port to implement a serial bus to an ADC). Commercial ADCs often have several inputs that feed the same converter, usually through an analog multiplexer. Different models of ADC may include sample and hold circuits, instrumentation amplifiers or differential inputs, where the quantity measured is the difference between two voltages.

Applications

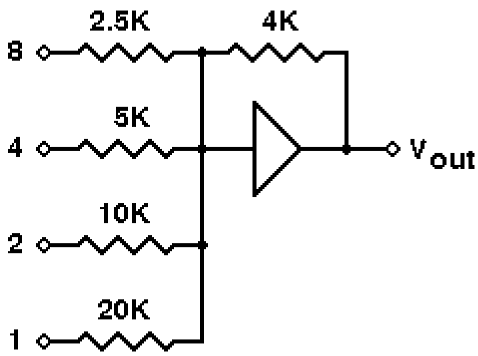
Application to music recording

ADCs are integral to current music reproduction technology. Since much music production is done on computers, when an analog recording is used, an ADC is needed to create the PCM data stream that goes onto a compact disc or digital music file.The current crop of AD converters utilized in music can sample at rates up to 192 kilohertz.

High bandwidth headroom allows the use of cheaper or faster anti-aliasing filters of less severe filtering slopes. The proponents of oversampling assert that such shallower anti-aliasing filters produce less deleterious effects on sound quality, exactly because of their gentler slopes. Others prefer entirely filterless AD conversion, arguing that aliasing is less detrimental to sound perception than pre-conversion brickwall filtering. Considerable literature exists on these matters, but commercial considerations often play a significant role. Most[citation needed] high-profile recording studios record in 24-bit/192-176.4 kHz PCM or in DSD formats, and then downsample or decimate the signal for Red-Book CD production (44.1 kHz or at 48 kHz for

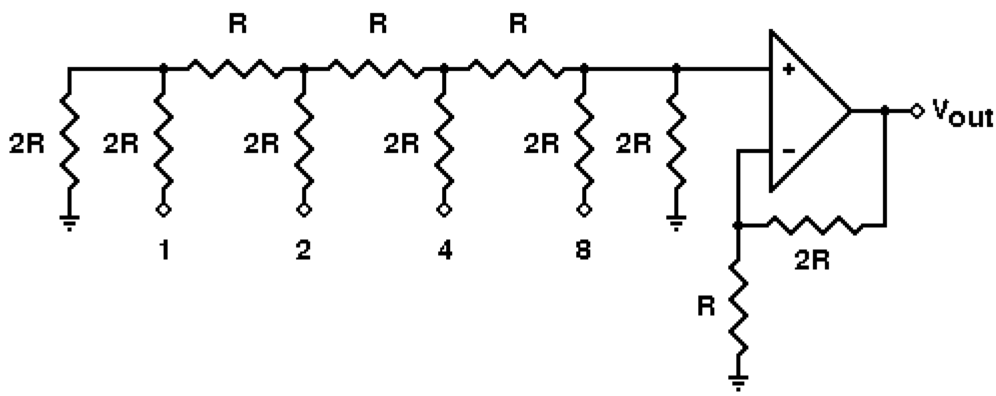
Digital to Analog Conversion :

One common requirement in electronics is to convert signals back and forth between analog and digital forms. Most such conversions are ultimately based on a digital-to-analog converter circuit. Therefore, it is worth exploring just how we can convert a digital number that represents a voltage value into an actual analog voltage.



The circuit to the right is a basic digital-to-analog (D to A) converter. It assumes a 4-bit binary number in Binary-Coded Decimal (BCD) format, using +5 volts as a logic 1 and 0 volts as a logic 0. It will convert the applied BCD number to a matching (inverted) output voltage. The digits 1, 2, 4, and 8 refer to the relative weights assigned to each input. Thus, 1 is the Least Significant Bit (LSB) of the input binary number, and 8 is the Most Significant Bit (MSB).

If the input voltages are accurately 0 and +5 volts, then the "1" input will cause an output voltage of -5 × (4k/20k) = -5 × (1/5) = -1 volt whenever it is a logic 1. Similarly, the "2," "4," and "8" inputs will control output voltages of -2, -4, and -8 volts, respectively. As a result, the output voltage will take on one of 10 specific voltages, in accordance with the input BCD code.Unfortunately, there are several practical problems with this circuit. First, most digital logic gates do not accurately produce 0 and +5 volts at their outputs. Therefore, the resulting analog voltages will be close, but not really accurate. In addition, the different input resistors will load the digital circuit outputs differently, which will almost certainly result in different voltages being applied to the summer inputs.



The circuit above performs D to A conversion a little differently. Typically the inputs are driven by CMOS gates, which have low but equal resistance for both logic 0 and logic 1. Also, if we use the same logic levels, CMOS gates really do provide +5 and 0 volts for their logic levels.The input circuit is a remarkable design, known as an R-2R ladder network. It has several advantages over the basic summer circuit we saw first:

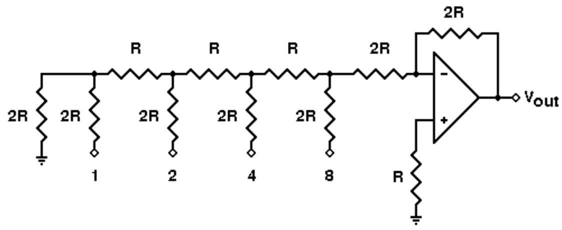
1. Only two resistance values are used anywhere in the entire circuit. This means that only two values of precision resistance are needed, in a resistance ratio of 2:1. This requirement is easy to meet, and not especially expensive.
2. The input resistance seen by each digital input is the same as for every other input. The actual impedance seen by each digital source gate is 3R. With a CMOS gate resistance of 200 ohms, we can use the very standard values of 10k and 20k for our resistors.
3. The circuit is indefinitely extensible for binary numbers. Thus, if we use binary inputs instead of BCD, we can simply double the length of the ladder network for an 8-bit number (0 to 255) or double it again for a 16-bit

number(0 to 65535). We only need to add two resistors for each additional binary input.

4. The circuit lends itself to a non-inverting circuit configuration. Therefore we need not be concerned about intermediate inverters along the way. However, an inverting version can easily be configured if that is appropriate.

One detail about this circuit: Even if the input ladder is extended, the output will remain within the same output voltage limits. Additional input bits will simply allow the output to be subdivided into smaller increments for finer resolution. This is equivalent to adding inputs with ever-larger resistance values (doubling the resistance value for each bit), but still using the same two resistance values in the extended ladder.The basic theory of the R-2R ladder network is actually quite simple. Current flowing through any input resistor (2R) encounters two possible paths at the far end. The effective resistances of both paths are the same (also 2R), so the incoming current splits equally along both paths. The half-current that flows back towards lower orders of magnitude does not reach the op amp, and therefore has no effect on the output voltage. The half that takes the path towards the op amp along the ladder can affect the output.

The most significant bit (marked "8" in the figure) sends half of its current toward the op amp, so that half of the input current flows through that final 2R resistance and generates a voltage drop across it. This voltage drop (from bit "8" only) will be one-third of the logic 1 voltage level, or 5/3 = 1.667 volts. This is amplified by the op amp, as controlled by the feedback and input resistors connected to the "-" input. For the components shown, this gain will be 3 (see the page on non-inverting amplifiers). With a gain of 3, the amplifier output voltage for the "8" input will be 5/3 × 3 = 5 volts.The current from the "4" input will split in half in the same way. Then, the half going towards the op amp will encounter the junction from the "8" input. Again, this current "sees" two equal-resistance paths of 2R each, so it will split in half again. Thus, only a quarter of the current from the "4" will reach the op amp. Similarly, only 1/8 of the current from the "2" input will reach the op amp and be counted. This continues backwards for as many inputs as there are on the R-2R ladder structure.The maximum output voltage from this circuit will be one step of the least significant bit below 10 volts. Thus, an 8-bit ladder can produce output voltages up to 9.961 volts (255/256 × 10 volts). This is fine for many applications. If you have an application that requires a 0-9 volt output from a BCD input, you can easily scale the output upwards using an amplifier with a gain of 1.6 (8/5).



If you want an inverting D to A converter, the circuit shown above will work well. You may need to scale the output voltage, depending on your requirements. Also, it is possible to have a bipolar D to A converter. If you apply the most significant bit to an analog inverter and use that output for the MSB position of the R-2R ladder, the binary number applied to the ladder will be handled as a two's-complement number, going both positive and negative. A frequency counter is an electronic instrument, or component of one, that is used for measuring frequency. Frequency is defined as the number of events of a particular sort occurring in a set period of time. Frequency counters usually measure the number of oscillations or pulses per second in a repetitive electronic signal.

Operating principle :

Most frequency counters work by using a counter which accumulates the number of events occurring within a specific period of time. After a preset period (1 second, for example), the value in the counter is transferred to a display and the counter is reset to zero. If the event being measured repeats itself with sufficient stability and the frequency is considerably lower than that of the clock oscillator being used, the resolution of the measurement can be greatly improved by measuring the time required for an entire number of cycles, rather than counting the number of entire cycles observed for a pre-set duration (often referred to as the reciprocal technique). The internal oscillator which provides the time signals is called the timebase, and must be calibrated very accurately.If the thing to be counted is already in electronic form, simple interfacing to the instrument is all that is required. More complex signals may need some conditioning to make them suitable for counting. Most general purpose frequency counters will include some form of amplifier, filtering and shaping circuitry at the input. DSP technology, sensitivity control and hysteresis are other techniques to improve performance. Other types of periodic events that are not inherently electronic in nature will need to be converted using some form of transducer. For example, a mechanical event could be arranged to interrupt a light beam, and the counter made to count the resulting pulses.

Frequency counters designed for radio frequencies (RF) are also common and operate on the same principles as lower frequency counters. Often, they have more range before they overflow. For very high (microwave)frequencies, many designs use a high-speed prescaler to bring the signal frequency down to a point where normal digital circuitry can operate. The displays on such instruments take this into account so they still display the correct value. Microwave frequency counters can currently measure frequencies up to almost 100 GHz. Above these frequencies the signal to be measured is combined in a mixer with the signal from a local oscillator, producing a signal at the difference frequency, which is low enough to be measured directly.

Accuracy :

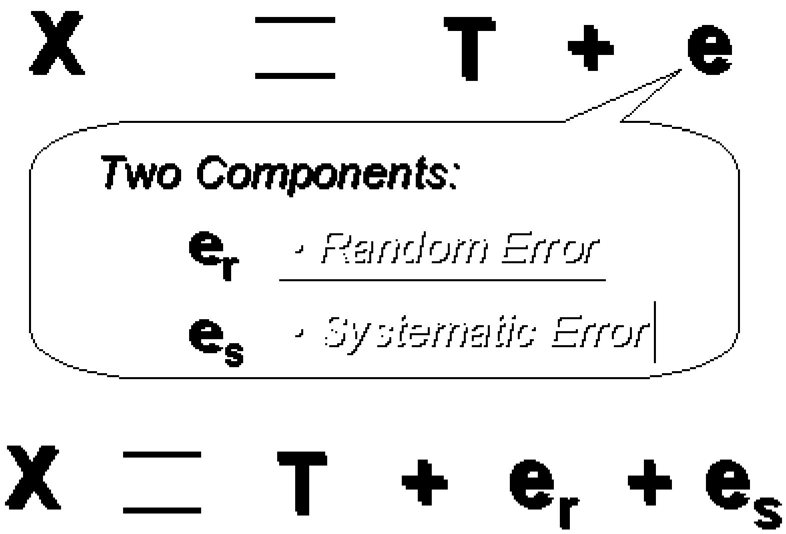
The accuracy of a frequency counter is strongly dependent on the stability of its timebase. Highly accurate circuits are used to generate this for instrumentation purposes, usually using a quartz crystal oscillator within a sealed temperature-controlled chamber known as a crystal oven or OCXO (oven controlled crystal oscillator). For higher accuracy measurements, an external frequency reference tied to a very high stability oscillator such as a GPS disciplined rubidium oscillator may be used. Where the frequency does not need to be known to such a high degree of accuracy, simpler oscillators can be used. It is also possible to measure frequency using the same techniques in software in an embedded system. A CPU for example, can be arranged to measure its own frequency of operation provided it has some reference timebase to compare with.

I/O Interfaces

I/O interfaces allow the user to send information to the frequency counter and receive information from the frequency counter. Commonly-used interfaces include RS232, USB, GPIB and Ethernet. Besides sending measurement results, a counter can notify the user when user-defined measurement limits are exceeded. Common to many counters are the SCPI commands used to control them. A new development is built-in LAN-based control via Ethernet complete with GUI's. This allows one computer to control one or several instruments and eliminates the need to write SCPI commands.

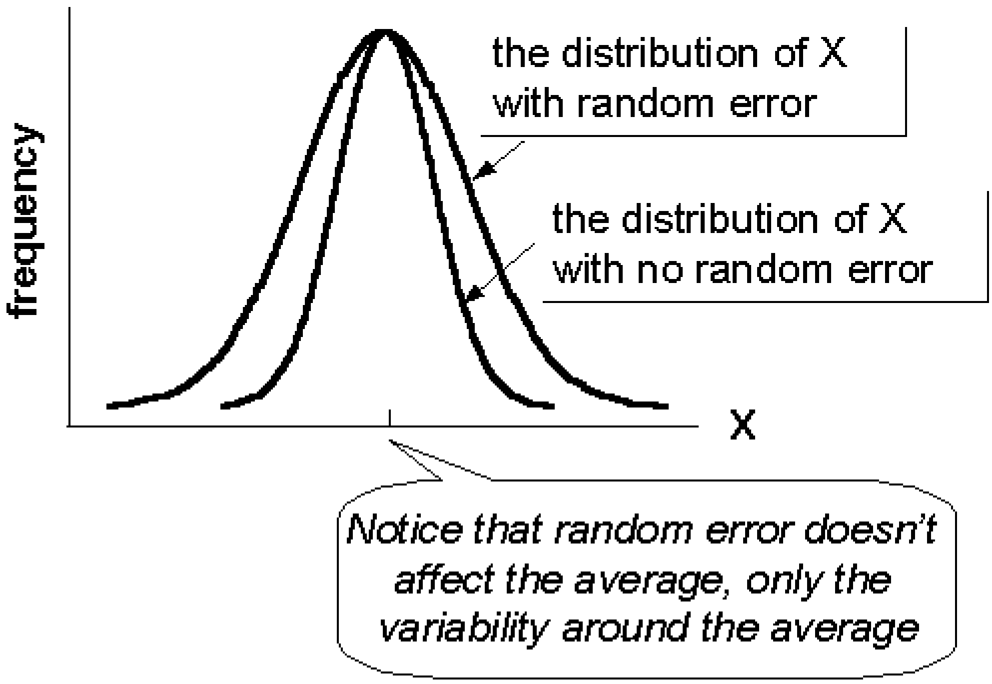
Measurement Error :

The true score theory is a good simple model for measurement, but it may not always be an accurate reflection of reality. In particular, it assumes that any observation is composed of the true value plus some random error value. But is that reasonable? What if all error is not random? Isn't it possible that some errors are systematic, that they hold across most or all of the members of a group? One way to deal with this notion is to revise the simple true score model by dividing the error component into two subcomponents, random error and systematic error. here, we'll look at the differences between these two types of errors and try to diagnose their effects on our research.



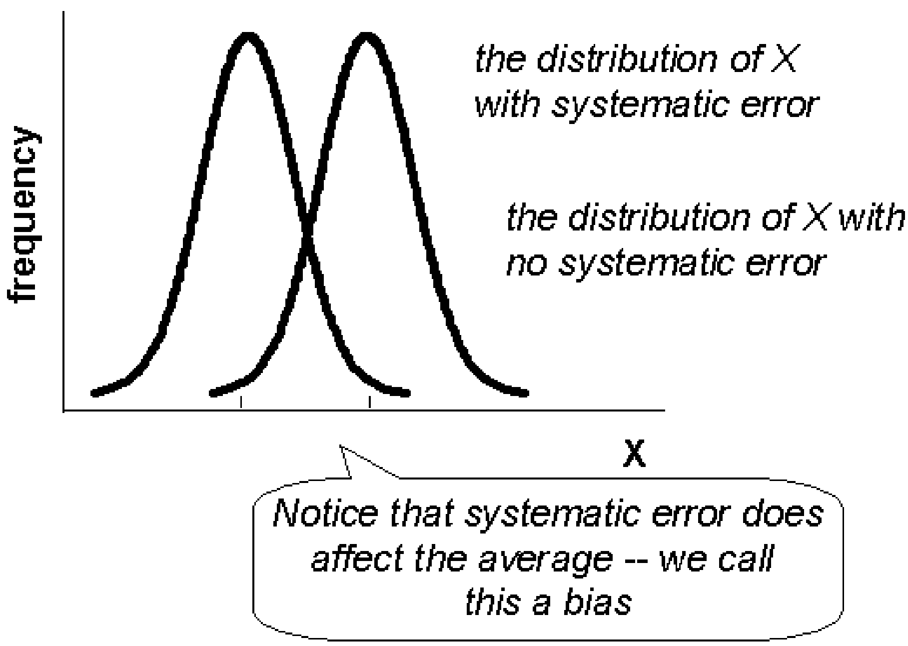
What is Random Error?

Random error is caused by any factors that randomly affect measurement of the variable across the sample. For instance, each person's mood can inflate or deflate their performance on any occasion. In a particular testing, some children may be feeling in a good mood and others may be depressed. If mood affects their performance on the measure, it may artificially inflate the observed scores for some children and artificially deflate them for others. The important thing about random error is that it does not have any consistent effects across the entire sample. Instead, it pushes observed scores up or down randomly. This means that if we could see all of the random errors in a distribution they would have to sum to 0 -- there would be as many negative errors as positive ones. The important property of random error is that it adds variability to the data but does not affect average performance for the group. Because of this, random error is sometimes considered noise.



What is Systematic Error?

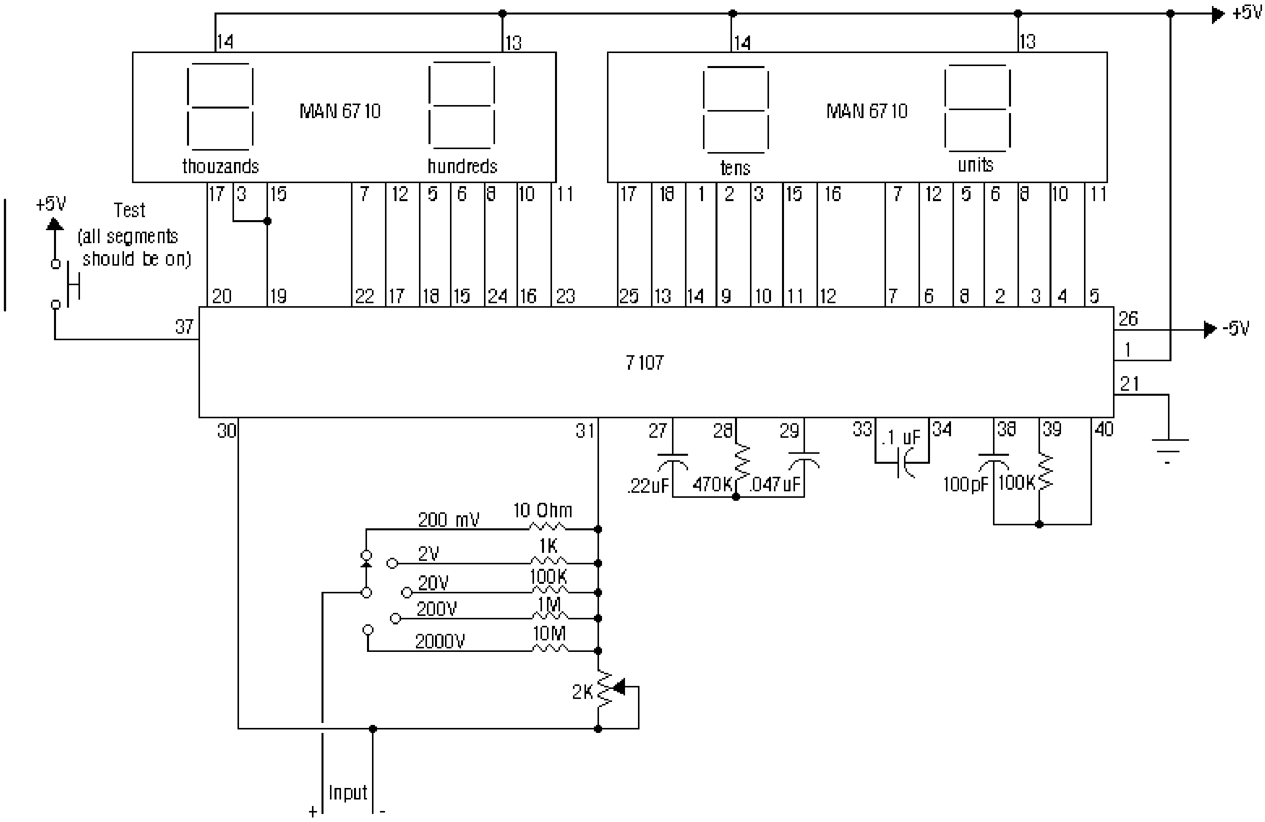
Systematic error is caused by any factors that systematically affect measurement of the variable across the sample. For instance, if there is loud traffic going by just outside of a classroom where students are taking a test, this noise is liable to affect all of the children's scores -- in this case, systematically lowering them. Unlike random error, systematic errors tend to be consistently either positive or negative -- because of this, systematic error is sometimes considered to be bias in measurement.



Reducing Measurement Error :

So, how can we reduce measurement errors, random or systematic? One thing you can do is to pilot test your instruments, getting feedback from your respondents regarding how easy or hard the measure was and information about how the testing environment affected their performance. Second, if you are gathering measures using people to collect the data (as interviewers or observers) you should make sure you train them thoroughly so that they aren't inadvertently introducing error. Third, when you collect the data for your study you should double-check the data thoroughly. All data entry for computer analysis should be "double-punched" and verified. This means that you enter the data twice, the second time having your data entry machine check that you are typing the exact same data you did the first time. Fourth, you can use statistical procedures to adjust for measurement error. These range from rather simple formulas you can apply directly to your data to very complex modeling procedures for modeling the error and its effects. Finally, one of the best things you can do to deal with measurement errors, especially systematic errors, is to use multiple measures of the same construct. Especially if the different measures don't share the same systematic errors, you will be able to triangulate across the multiple measures and get a more accurate sense of what's going on.

A digital voltmeter, or DVM, is used to take highly accurate voltage measurements. These instruments measure the electrical potential difference between two conductors in a circuit. DVMs are electric voltmeters, and the preferred standard, as they offer several benefits over their analog counterparts. Voltmeters are used to measure the gain or loss of voltage between two points in a circuit. The leads are connected in parallel on each side of the circuit being tested. The positive terminal of the meter should be connected closest to the power supply. In turn, the negative terminal would be connected after the circuit being tested. The analog dial or digital display will exhibit the voltage measurement.



A digital voltmeter typically consists of an analog to digital converter (A/D) with a digital display. The analog signal is converted into a digital code proportionate to the magnitude of the signal. Voltages from picovolts to megavolts are measurable, though the scale usually graduates in millivolts, volts, or kilovolts. Frequencies between zero and several megahertz may also be measured.DVMs measure both alternating current (AC) and direct current (DC) in electronics. Common laboratory and commercial applications involve electromechanical machinery with a current flowing through wires and circuits. Often, a digital voltmeter is used to monitor a unit, such as a generator. Portable or handheld devices, such as the digital multimeter (DMM), for example, may combine several functions into one instrument measuring voltage, current, and resistance. This is the preferred tool of an electrician.Many DVMs integrate outputs for monitoring, controlling, transmitting, and printing of data. Advanced systems are often connected to computers, allowing for automation, optimization of processes, and prevention of malfunctions and critical failure safeties. Chemical plants can convert measurements to voltage, and control and monitor temperature, pressure, level, or flow. Medical equipment, such as x-ray machines, may use a digital voltmeter to make sure the voltage of the equipment is in the proper range.

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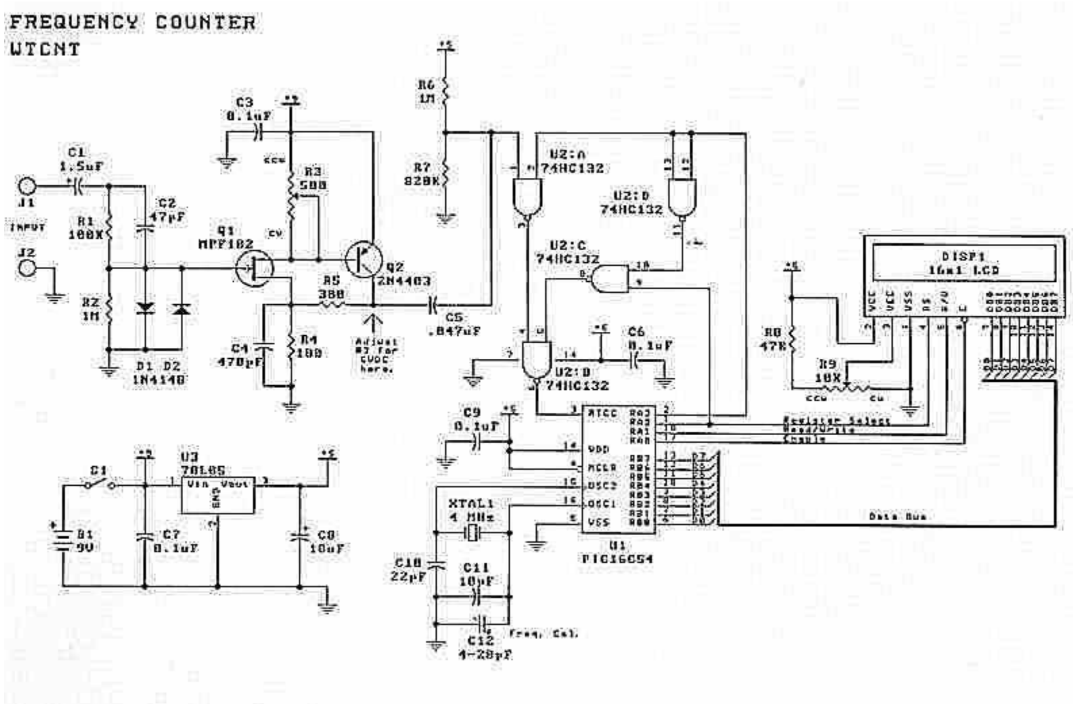

Figure 2

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the SCPI commands used to control them. A new development is built-in LAN-based control via Ethernet complete with GUI's. This allows one computer to control one or several instruments and eliminates the need to write SCPI commands any time

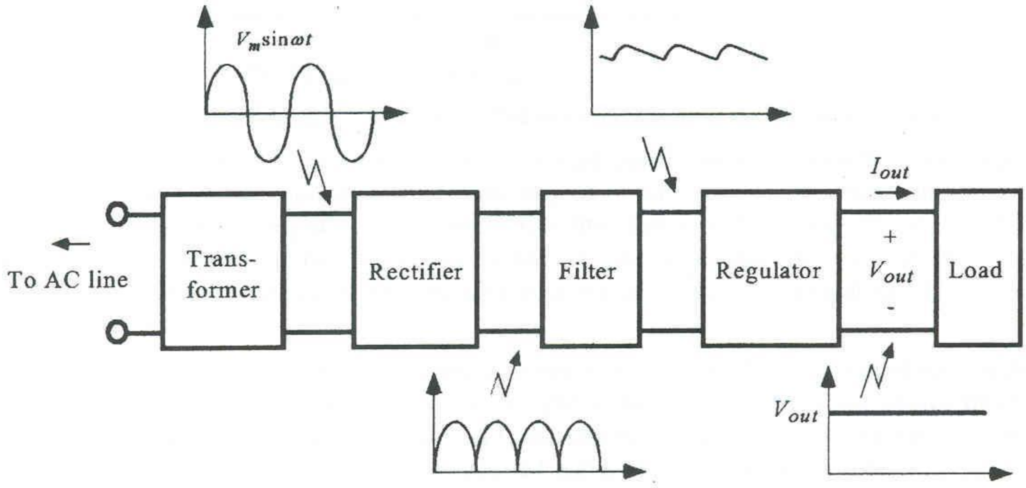
Multimeter :

Multimeter or a multitester, also known as a volt/ohm meter or VOM, is an electronic measuring instrument that combines several measurement functions in one unit. A typical multimeter may include features such as the ability to measure voltage, current and resistance. Multimeters may use analog or digital circuits—analog multimeters and digital multimeters (often abbreviated DMM or DVOM.) Analog instruments are usually based on a microammeter whose pointer moves over a scale calibration for all the different measurements that can be made; digital instruments usually display digits, but may display a bar of a length

BASIC

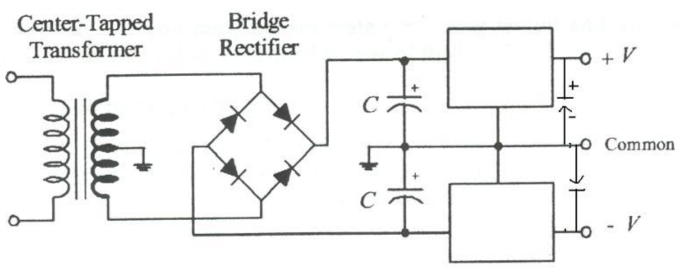
CIRCUIT: (i)

Block Diagram:



Components of a typical linear power supply

(ii) Circuit:



A multimeter can be a hand-held device useful for basic fault finding and field service work or a bench instrument which can measure to a very high degree of accuracy. They can be used to troubleshoot electrical problems in a wide array of industrial and household devices such as electronic equipment, motor controls, domestic appliances, power supplies, and wiring systems.

Quantities measured :

Contemporary multimeters can measure many quantities. The common ones are:

- Voltage, alternating and direct, in volts.
- Current,alternating and direct, in amperes.
  The frequency range for which AC measurements are accurate must be

specified. • Resistance in ohms.

Additionally, some multimeters measure:

- Capacitance in farads.
- Conductance in siemens.
- Decibels.
- Duty cycle as a percentage.
- Frequency in hertz.
- Inductance in henrys.
- Temperature in degrees Celsius or Fahrenheit, with an appropriate temperature test probe, often a thermocouple.

Digital multimeters may also include circuits for:

- Continuity; beeps when a circuit conducts.
- Diodes (measuring forward drop of diode junctions, i.e., diodes and transistor junctions) and transistors (measuring current gain and other parameters).
- Battery checking for simple 1.5 volt and 9 volt batteries. This is a current loaded voltage scale. Battery checking (ignoring internal resistance, which increases as the battery is depleted), is less accurate when using a DC voltage scale.

Various sensors can be attached to multimeters to take measurements such as:

- Light level
- Acidity/Alkalinity(pH)
- Wind speed
- Relative humidity

Resolution

Digital

The resolution of a multimeter is often specified in "digits" of resolution. For example, the term 5½ digits refers to the number of digits displayed on the display of a multimeter.By convention, a half digit can display either a zero or a one, while a three-quarters digit can display a numeral higher than a one but not nine. Commonly, a three-quarters digit refers to a maximum value of 3 or 5. The fractional digit is always the most significant digit in the displayed value. A 5½ digit multimeter would have five full digits that display values from 0 to 9 and one half digit that could only display 0 or 1.[3] Such a meter could show positive or negative values from 0 to 199,999. A 3¾ digit meter can display a quantity from 0 to 3,999 or 5,999, depending on the manufacturer.While a digital display can easily be extended in precision, the extra digits are of no value if not accompanied by care in the design and calibration of the analog portions of the multimeter. Meaningful high-resolution measurements require a good understanding of the instrument specifications, good control of the measurement conditions, and traceability of the calibration of the instrument.

Specifying "display counts" is another way to specify the resolution. Display counts give the largest number, or the largest number plus one (so the count number looks nicer) the multimeter's display can show, ignoring a decimal separator. For example, a 5½ digitmultimeter can also be specified as a 199999 display count or 200000 display count multimeter. Often the display count is just called the count in multimeter specifications.

Analog :

Resolution of analog multimeters is limited by the width of the scale pointer, vibration of the pointer, the accuracy of printing of scales, zero calibration, number of ranges, and errors due to non-horizontal use of the mechanical display. Accuracy of readings obtained is also often compromised by miscounting division markings, errors in mental arithmetic, parallax observation errors, and less than perfect eyesight. Mirrored scales and larger meter movements are used to improve resolution; two and a half to three digits equivalent resolution is usual (and is usually adequate for the limited precision needed for most measurements).Resistance measurements, in particular, are of low precision due to the typical resistance measurement circuit which compresses the scale heavily at the higher resistance values. Inexpensive analog meters may have only a single resistance scale, seriously restricting the range of precise measurements. Typically an analog meter will have a panel adjustment to set the zero-ohms calibration of the meter, to compensate for the varying voltage of the meter battery.

Accuracy

Digital multimeters generally take measurements with accuracy superior to their analog counterparts. Standard analog multimeters measure with typically three percent accuracy,[4] though instruments of higher accuracy are made. Standard portable digital multimeters are specified to have an accuracy of typically 0.5% on the DC voltage ranges. Mainstream bench-top multimeters are available with specified accuracy of better than ±0.01%. Laboratory grade instruments can have accuracies of a few parts per million.Accuracy figures need to be interpreted with care. The accuracy of an analog instrument usually refers to full-scale deflection; a measurement of 10V on the 100V scale of a 3% meter is subject to an error of 3V, 30% of the reading. Digital meters usually specify accuracy as a percentage of reading plus a percentage of full-scale value, sometimes expressed in counts rather than percentage terms.Quoted accuracy is specified as being that of the lower millivolt (mV) DC range, and is known as the "basic DC volts accuracy" figure. Higher DC voltage ranges, current, resistance, AC and other ranges will usually have a lower accuracy than the basic DC volts figure. AC measurements only meet specified accuracy within a specified range of frequencies.

Test equipment tends to drift out of calibration over time, and the specified accuracy cannot be relied upon indefinitely. For more expensive equipment, manufacturers and third parties provide calibration services so that older equipment may be recalibrated and recertified. The cost of such services is disproportionate for inexpensive equipment; however extreme accuracy is not required for most routine testing. Multimeters used for critical measurements may be part of a metrology program to assure calibration.

Sensitivity and input impedance

When used for measuring voltage, the input impedance of the multimeter must be very high compared to the impedance of the circuit being measured; otherwise circuit operation may be changed, and the reading will also be inaccurate.Meters with electronic amplifiers (all digital multimeters and some analog meters) have a fixed input impedance that is high enough not to disturb most circuits. This is often either one or ten megohms;the standardization of the input resistance allows the use of external high-resistance probes which form a voltage divider with the input resistance to extend voltage range up to tens of thousands of volts.

Most analog multimeters of the moving-pointer type are unbuffered, and draw current from the circuit under test to deflect the meter pointer. The impedance of the meter varies depending on the basic sensitivity of the meter movement and the range which is selected. For example, a meter with a typical 20,000 ohms/volt sensitivity will have an input resistance of two million ohms on the 100 volt range (100 V * 20,000 ohms/volt = 2,000,000 ohms). On every range, at full scale voltage of the range, the full current required to deflect the meter movement is taken from the circuit under test. Lower sensitivity meter movements are acceptable for testing in circuits where source impedances are low compared to the meter impedance, for example, power circuits; these meters are more rugged mechanically. Some measurements in signal circuits require higher sensitivity movements so as not to load the circuit under test with the meter impedance. Sometimes sensitivity is confused with resolution of a meter, which is defined as the lowest voltage, current or resistance change that can change the observed reading.For general-purpose digital multimeters, the lowest voltage range is typically several hundred millivolts AC or DC, but the lowest current range may be several hundred milliamperes, although instruments with greater current sensitivity are available. Measurement of low resistance requires lead resistance (measured by touching the test probes together) to be subtracted for best accuracy.The upper end of multimeter measurement ranges varies considerably; measurements over perhaps 600 volts, 10 amperes, or 100 megohms may require a specialized test instrument.

Burden voltage

Any ammeter, including a multimeter in a current range, has a certain resistance. Most multimeters inherently measure voltage, and pass a current to be measured through a shunt resistance, measuring the voltage developed across it. The voltage drop is known as the burden voltage, specified in volts per ampere. The value can change depending on the range the meter selects, since different ranges usually use different shunt resistors.The burden voltage can be significant in low-voltage circuits. To check for its effect on accuracy and on external circuit operation the meter can be switched to different ranges; the current reading should be the same and circuit operation should not be affected if burden voltage is not a problem. If this voltage is significant it can be reduced (also reducing the inherent accuracy and precision of the measurement) by using a higher current range.

Alternating current sensing

Since the basic indicator system in either an analog or digital meter responds to DC only, a multimeter includes an AC to DC conversion circuit for making alternating current measurements. Basic meters utilize a rectifier circuit to measure the average or peak absolute value of the voltage, but are calibrated to show the calculated root mean square (RMS) value for a sinusoidal waveform;this will give correct readings for alternating current as used in power distribution. User guides for some such meters give correction factors for some simple non-sinusoidal waveforms, to allow the correct root mean square (RMS) equivalent value to be calculated. More expensive multimeters include an AC to DC converter that measures the true RMS value of the waveform within certain limits; the user manual for the meter may indicate the limits of the crest factor and frequency for which the meter calibration is valid. RMS sensing is necessary for measurements on non-sinusoidal periodic waveforms, such as found in audio signals and variable-frequency drives.

Question Bank

UNIT-IV

DIGITAL INSTRUMENTS

PART-A (2 Marks)

1. **What is digital voltmeter?**

   The digital voltmeters generally referred as DVM, convert the analog signals into digital and display the voltages to be measured as discrete numerical instead of pointer deflection, on the digital displays.

2. **Give classification of digital voltmeters? (AU APRIL 2004)**

   DVM mainly classified into two types.

   I. Non integrating type.
   II. Integrating type.

Non-integrating type DVM.

- i. Potentiometric type
  - a) Servo potentiometric type
  - b) Successive approximation type
  - c) Null balance type
- ii. Ramp type
  - a) Linear type
  - b) Staircase type

Integrating type DVM

- i. Voltage to frequency converter type
- ii. Potentiometric type
- iii. Dual slope integrating type

3. **What is the principle of ramp type digital voltmeter? (AU APRIL 2005)**

The basic principle of such measurement is based on the measurement of the time taken by a linear ramp to rise 0 v to the level of the input voltage or to decrease the electronic time interval counter and the count is displayed in the numeric form with the help of a digital display.

4. **What are the essential parts of the ramp type DVM? (AU NOV 2004)**

   - Comparator
   - Oscillator
   - Attenuator
   - Gate
   - Counter

5. **What are the advantages of digital instruments? (AU APRIL 2006)**
   - Very high accuracy
   - No loading effect.
   - No moving parts
   - Auto range and polarity
   - Very high input impedance
   - Reading speed is high
   - Computerized control
6. **Why period mode is preferred for measurement of very low frequency in a frequency counter? (AU NOV 2006)**

The time period T = 1/f. So if the frequency to be measured low, then the accuracy of the frequency counter decreases as less number of pulses are connected to the gating circuit. Thus in low frequency region it is better to measure period rather than the frequency

7. **What is the importance of gate time in frequency counter? (AU APRIL 2007)**

   The gate time and gating circuit is used to adjust the train pulses in the frequency counter. This will done by using the selector switch S in the time base generator.

8. **How trigger time error reduced? (AU APRIL 2007)**

To limit the trigger level error, we can use the large signal amplitudes and fast rise times in the signal.

9. **What is the difference between analog and digital instruments? (AU APRIL 2008)**

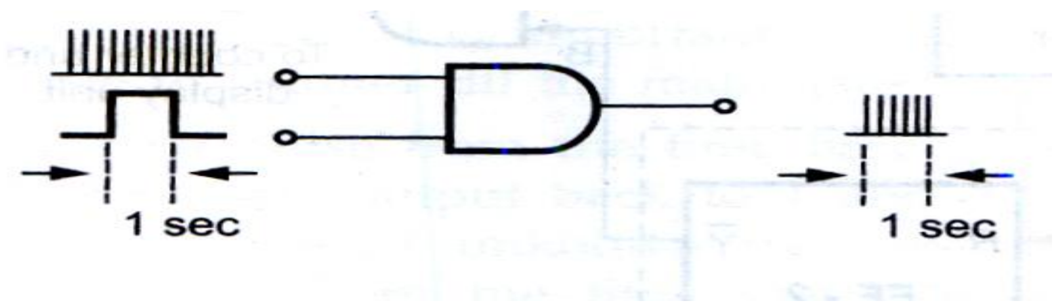
SL.NO	Parameter	Analog	Digital
1.	Accuracy	Less upto $\pm$ 0..1 %	Very high upto $\pm$ 0.005
2.	Resolution	Limited	High
3.	Power	Power required is High	Power required is Low
4.	Cost	Low	High
5.	Frictional errors	Errors due to movimg parts	No movong parts so no errors
6.	Range and polarity	No facility of auto ranging and polarity	Has the facility of auto ranging and polarity
7.	Speed	Reading speed is low	Reading speed is very high.

PART-B (16 Marks)

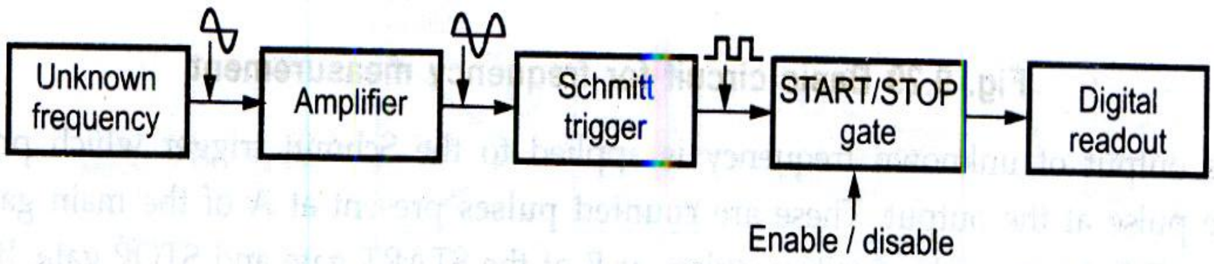
1. **Draw and explain the circuit of a digital frequency meter. (AU APRIL 2007,2010, NOV 2006,2008) (16)**

The frequency is the measure of repeatation of any signal. The frequency is nothing but the number of cycles of the signal per unit time.

Working principle

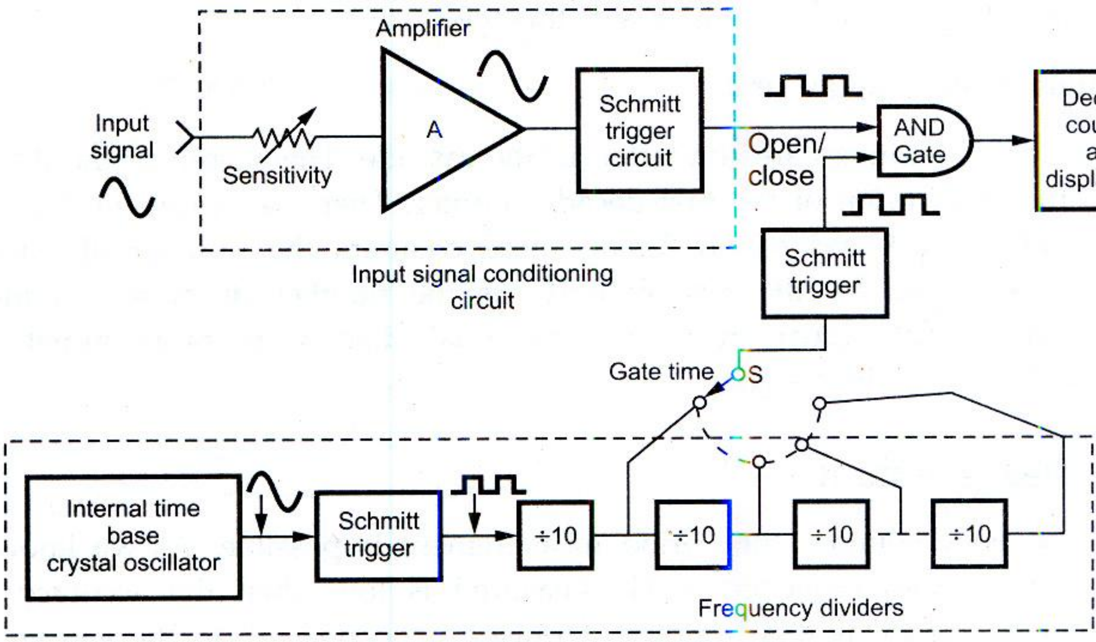


The signal waveform whose frequency is to be measured is converted into trigger pulses and applied continuously to one terminal of an AND gate. To the other terminal of the gate, a pulse of 1 sec is applied. The number of pulses counted at the output terminal during period of 1 sec indicates the frequency.



Digital Frequency Counter

For the unknown frequency measurements the digital frequency counter is the most accurate and reliable instrument available.



The major components of the digital frequency counter are as given below

1) Input signal conditioning circuit
2) Time base generator
3) Gating circuit
4) Decimal counter and display unit.

Input signal conditioning circuit.

In this circuit, an amplifier and Schmitt trigger are included. The threshold voltage of the Schmitt trigger can be controlled by sensitivity control on the control panel. First of all the input signal of unknown frequency is fed into the input signal conditioning circuit. There the signal is amplified and then it is converted into square wave by Schmitt trigger circuit.

Time base generator

The crystal oscillator produces a signal of 1 MHz or 100 MHz depending upon the requirement. In general, the accuracy of the digital frequency counter depends on the accuracy of the time base signals produced, thus the temperature compensated crystal oscillator is used. Then the output of the oscillator passed to the other Schmitt trigger circuit producing square wave output. Then it is fed to frequency dividers connected in cascade. Thus the train pulses are obtained after each frequency divider section. Using time base selector switch S the gate time can be adjusted.

Gating circuit

The gating circuit consists of AND gate. When the enable signal is provided to the AND gate, it allows a train pulses to pass through the gate for the time period selected by the time base circuit. The pulses are counted and then the second pulse generated from the time base generator disables AND gate and thus closes it.

Decimal counter and display unit

In this unit, decade counters are connected in the cascade. The output of the AND gate is connected to the clock input of the first decade counter. Then the output of this counter to the clock input of next and so on. Using these counters the number of pulses are counted and are displayed by the display unit. As the number of pulses are counted are proportional to the input signal frequency, the final display is proportional to the unknown frequency of the input signal.

2. Explain with a neat block diagram, the operation of ramp type digital voltmeter. (AU APRIL 2007,2010, NOV 2007,2008,2009) (16)

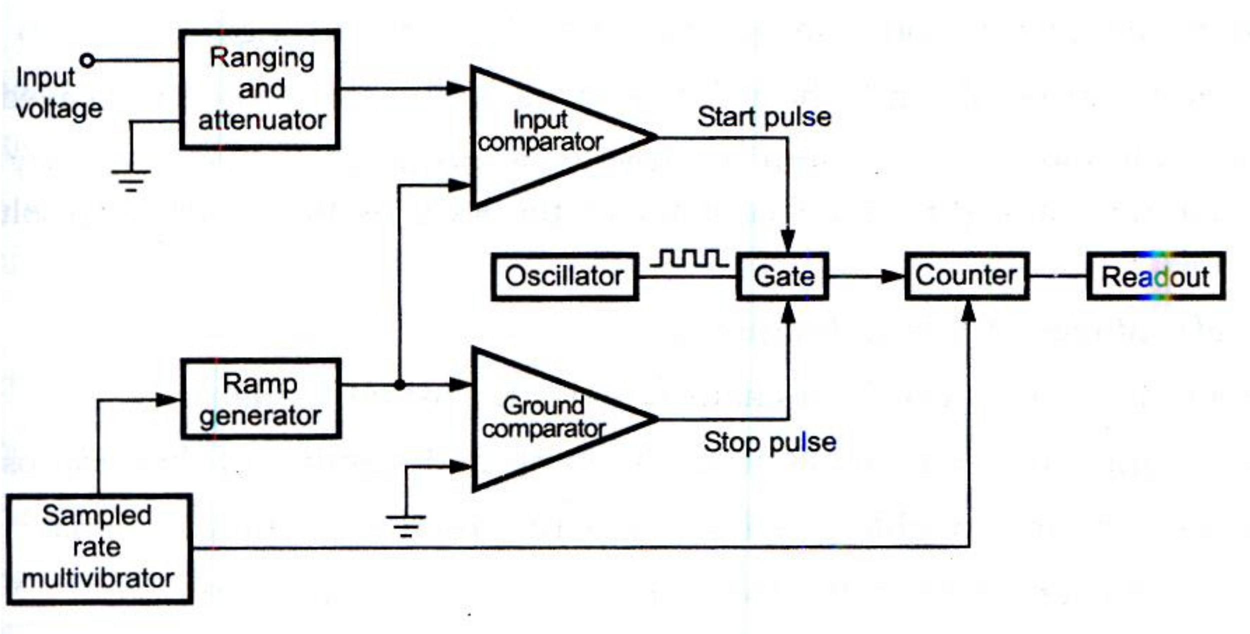
Ramp type DVM

It uses a linear ramp technique or staircase ramp technique. The stair case ramp technique is simpler than the linear ramp technique.

Linear ramp technique

The basic principle of such measurement is based on the measurement of the time taken by linear ramp to rise from 0 V to the level of the input voltage or to decrease from the level of the input voltage to zero. The time measured with the help of electronic time interval counter and the count is displayed in the numeric form with the help of digital display.

Block Diagram

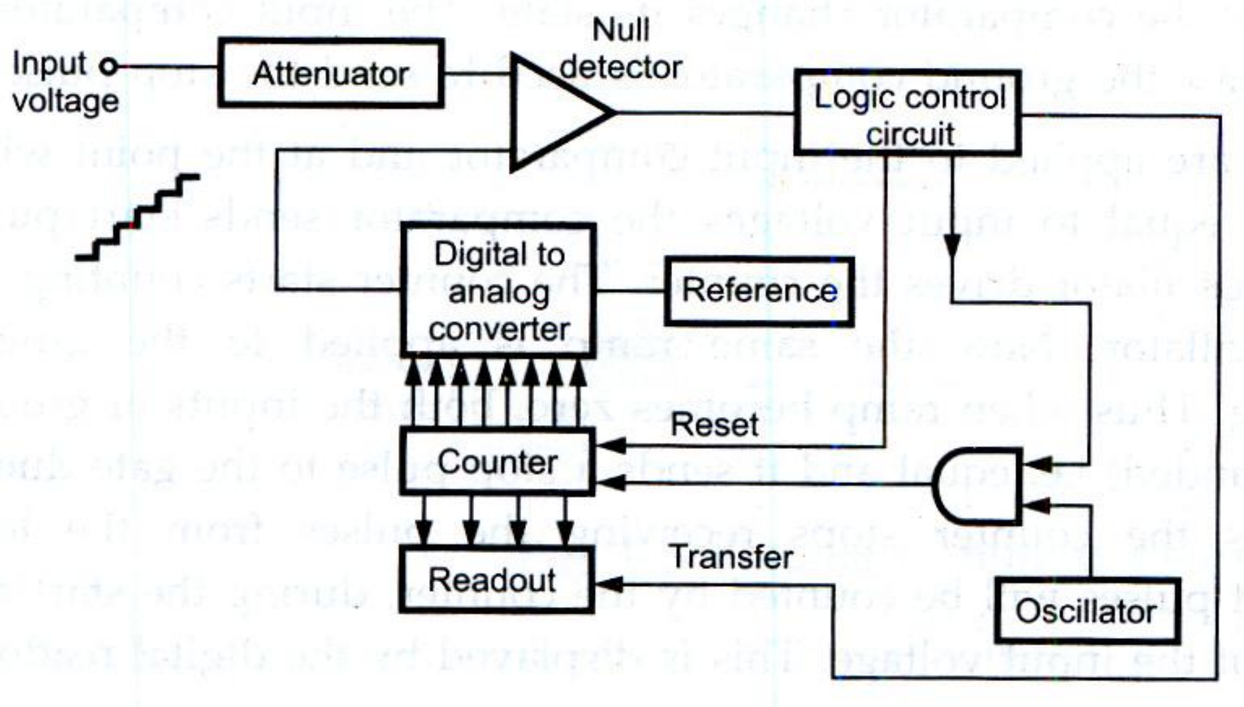


- Properly attenuated input signal is applied as one input to the input comparator
- The ramp generator generates the proper linear ramp signal which is applied to both the comparators.
- The input comparator is used to send the start pulse while the ground comparator is used to send the stop pulse.
- When the input ramp are applied to the input comparator and at the point when negative going ramp becomes equal to input voltages the comparator sends the start pulse due to which gate opens.
- The oscillator drives the counter. The counter starts counting the pulses received from the oscillator.
- Now the input ramp is applied to the ground comparator and it is decreasing. Thus when ramp becomes zero, both the inputs of ground comparator becomes zero and send it the stop pulse due to which gate closed.
- The sample rate multivibartor determines the rate at which the measurement cycles are initiated.

Staircase Ramp Technique

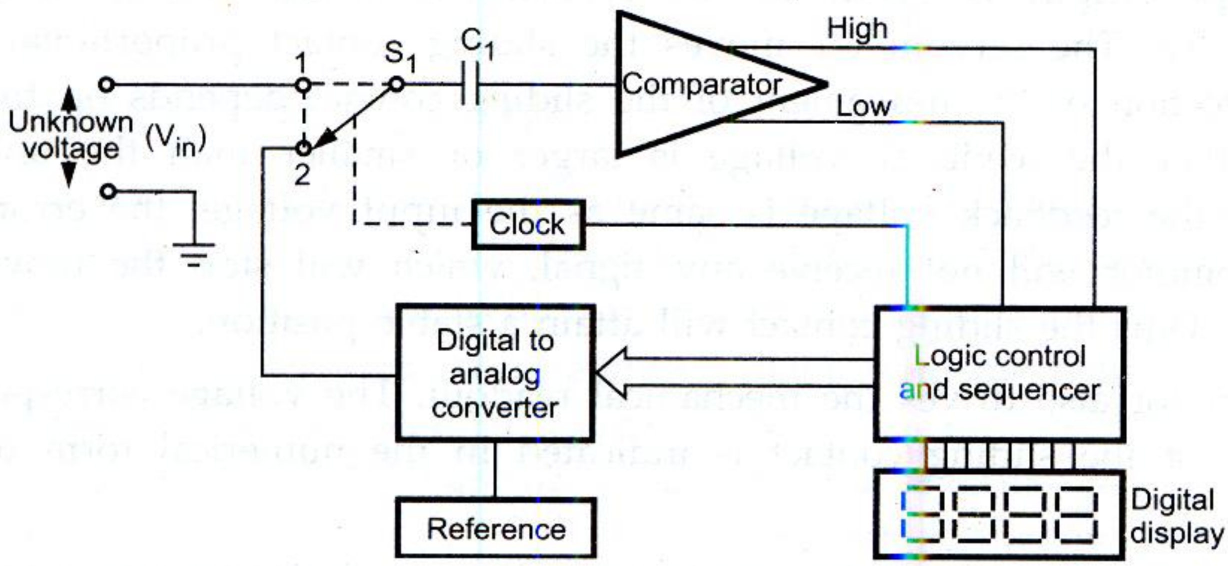
In this type of DVM, instead of linear ramp, the staircase ramp is used. The staircase ramp is generated by the digital to analog converter. The technique of using staircase ramp is also called null balance technique.

Block Diagram



- The input voltage is properly attenuated and is applied to a null detector. The input to null detector is the staircase ramp generated by the digital to analog converter.
- The ramp is continuously compared with the input signal.
- The logical control circuit sends a rest signal. This signal resets the counter. The digital to analog converter is also resetted by same signal.
- The output counter is given to the digital to analog converter which generates the ramp signal.
- At every count there is an incremental change in the ramp generated. Thus the staircase ramp is generated at the output of the digital to analog converter.
- This is given as the second input of the null detector.
- The increase in ramp continues till it achieves the voltage equal to input voltage.
- When the two voltages are equal, the null detector generates a signal which inturn initiates the logic control circuit.

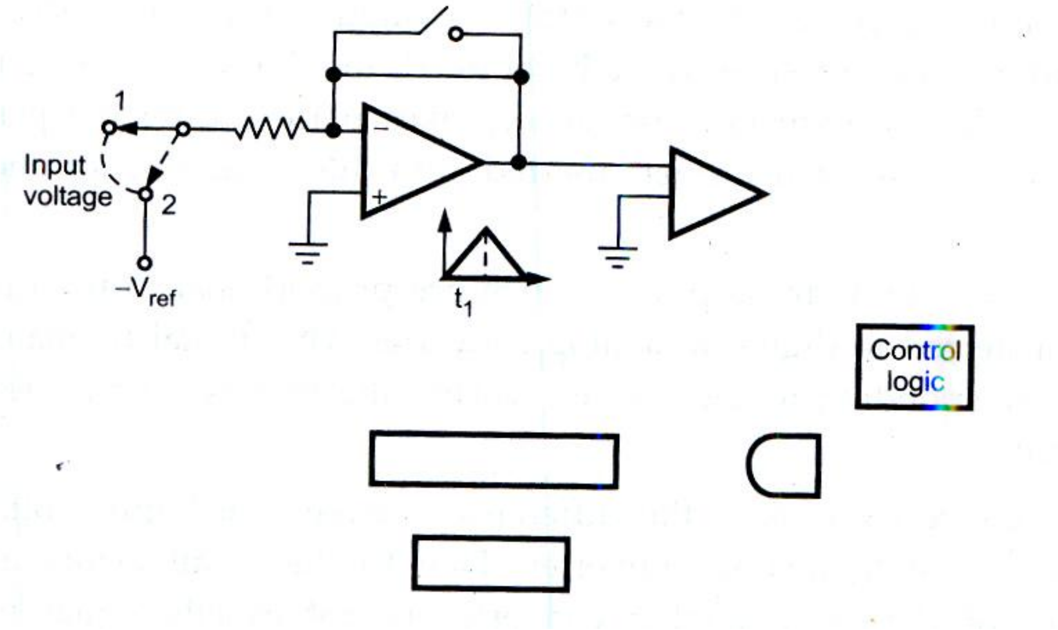
3. Explain with a neat block diagram, the operation of successive approximation type digital voltmeter. (AU NOV2008) (16)



- The potentiometric used in the servo balancing type DVM is a linear divider but in successive approximation type a digital divider is used.
- The digital divider is a Digital to analog (D/A) converter.
- The servo motor replaced by an electronic logic.
- The basic principle of measurement by this method is similar to the simple example of determination of weight of the object.
- The object is placed on one side of the balance and the approximate weight is placed on the other side.
- If this weight is smaller than the object, another small weight is added weight is removed and smaller weight is added.
- Thus by such successive procedure of adding and removing, the weight of the object is determined.
- The successive approximation type DVM works exactly on the same principle.
- In successive approximation type DVM, the comparator compares the output of digital to analog converter with the unknown voltage.
- Accordingly, the comparator provides logic high or low signals.
- The digital to analog converter successively generates the set pattern of signals.
- The procedure continues till the output of the digital to analog converter becomes equal to the unknown voltage.

4. Explain with a neat block diagram, the operation of dual slope integrating type digital voltmeter. (AU APRIL 2008, NOV 2009) (16)

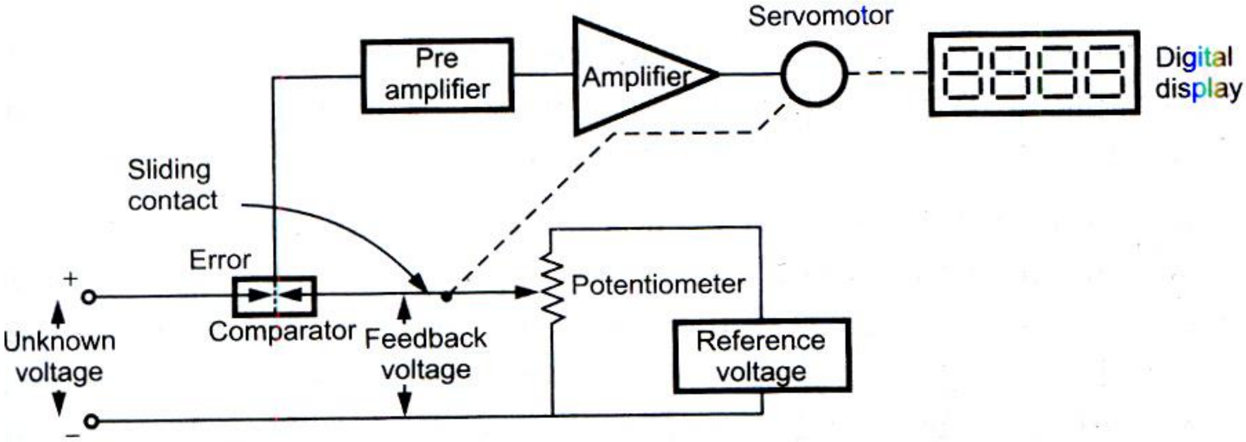
Block Diagram



- This is the most popular method of analog to digital conversion.
- In the ramp techniques, the noise can cause large errors but in dual slope method the noise is averaged out by the positive and negative ramps using the process of integration.
- The basic principle of this method is that the input signal is integrated for a fixed interval of time.
- And then the same integrator is used to integrate the reference voltage with reverse slope.
- Hence the name given to the technique is dual slope integration technique.
- When the switch S_1 is in the position 1, the capacitor C starts charging from zero level.
- The rate of charging is proportional to the input voltage level.
- After the interval t_1, the input voltage is disconnected and a negative voltage – V ref is connected by throwing the switch S_1 in position 2.
- Thus the input voltage is dependent on the time periods t_1 and t_2 and not on the values of R_1 and C.

5. Explain with a neat block diagram, the operation of servo potentiometric type digital voltmeter. . (AU APRIL 2006) (16)

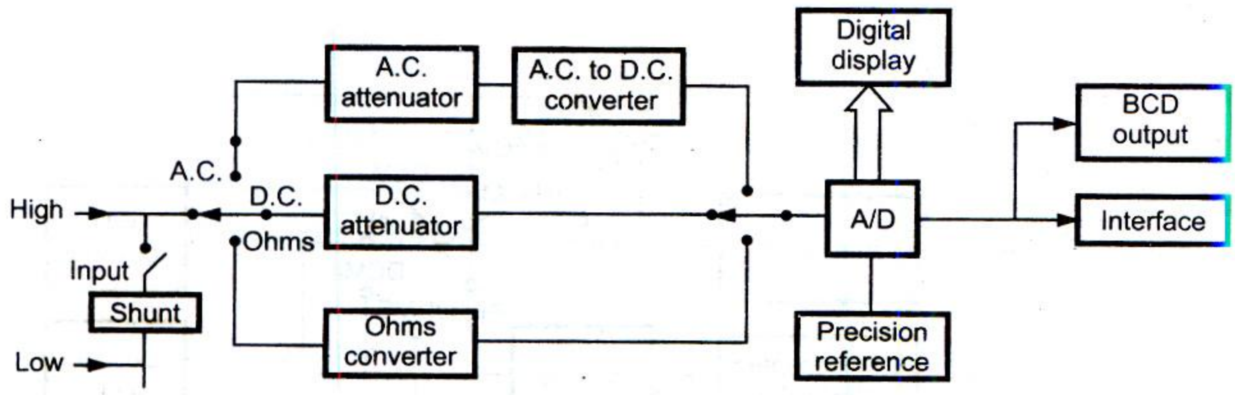
Block diagram



- In this potentiometric type voltmeters internal reference voltage is provided.
- A voltage comparison technique is used to measure the input voltage.
- The unknown voltage is compared with the reference voltage with the help of the setting of the calibrated potentiometer i.e. potential divider.
- The arm of the potentiometer is varied to obtain the null condition i.e. balance condition.
- The internal reference voltage is present at the two terminals of the potentiometer.
- When the null condition is obtained, the value of the unknown voltage is indicated by the dial setting of the potentiometer.
- Practically, the null balancing is not obtained manually but is obtained automatically.
- Such a voltmeter is called self balancing potentiometric type DVM.
- The servomotor is used to vary the arm of the potentiometer hence it is also called servo balancing potentiometer type DVM.

6. Describe a "digital multimeter" with a help of a block diagram explain its working. (AU NOV 2009) (16)

Block diagram



- In the digital multimeter the quantity measured by the meter is displayed by using 7 segment LED displays, alphanumeric displays or liquid crystal displays (LCDs), in the digital converters and other digital processing circuits.
- The digital multimeter is an instrument which is capable of measuring AC voltages DC voltages, AC and DC currents and resistance over several ranges.
- The current is converted to voltage by passing it through low shunt resistance.
- The AC quantities are converted to Dc by employing various rectifier and filtering circuits.
- While for the resistance measurements the meter consists of a precision low current source that is applied across the unknown resistance while gives DC voltages.
- All the quantities are digitalized using analog to digital converter and displayed in the digital form on the display.

- The analog multimeters require mo power supply and they suffer less from electric noise and isolation problems but still the digital multimeters have the following advantages over analog multimeters.
    - Accuracy is very high.
    - The input impedance is very high hence there is no loading effect.
    - An unambiguous reading at greater viewing distances is obtained.
    - The output available is electrical which can be used for interfacing with external equipment.
    - The prices are going down.
    - Small in size.

UNIT V- DATA ACQUISITION SYSTEMS AND FIBER OPTIC MEASUREMENTS

Data acquisition systems:

Data acquisition is the process of real world physical conditions and conversion of the resulting samples into digital numeric values that can be manipulated by a computer. Data acquisition and data acquisition systems (abbreviated with the acronym **DAS**) typically involves the conversion of analog waveforms into digital values for processing. The components of data acquisition systems include:

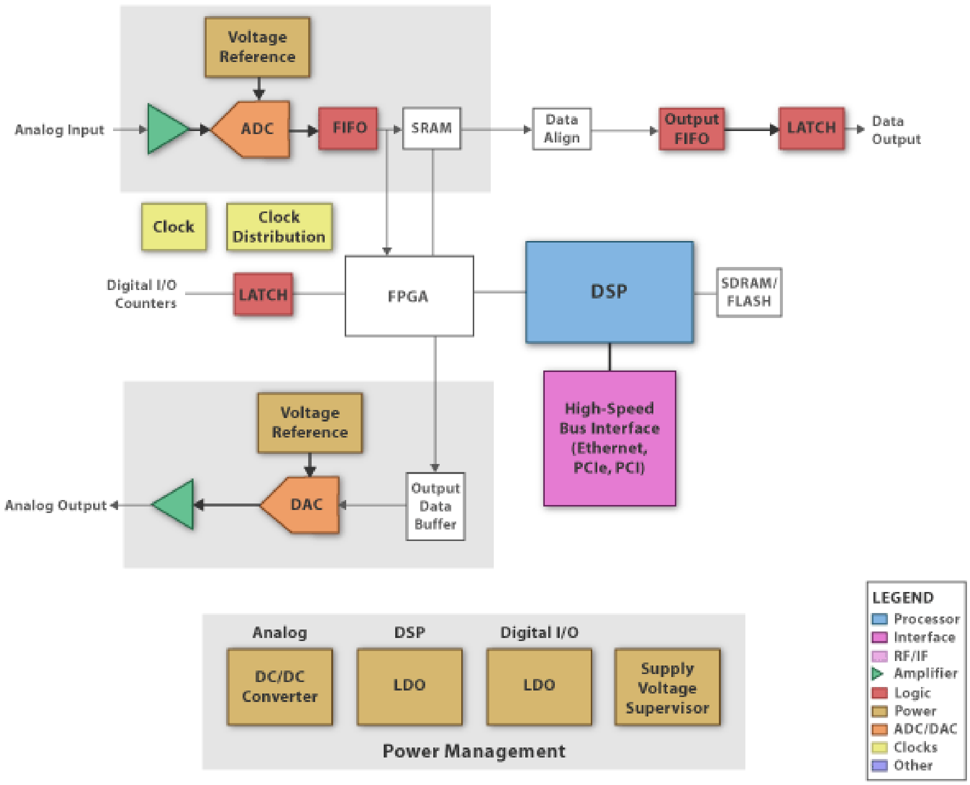
- Sensors that convert physical parameters to electrical signals.
- Signal conditioning circuitry to convert sensor signals into a form that can be converted to digital values.
- Analog-to-digital converters, which convert conditioned sensor signals to digital values.

Data acquisition is the process of extracting, transforming, and transporting data from the source systems and external data sources to the data processing system to be displayed, analyzed, and stored. A data acquisition system (DAQ) typically consist of transducers for asserting and measuring electrical signals, signal conditioning logic to perform amplification, isolation, and filtering, and other hardware for receiving analog signals and providing them to a processing system, such as a personal computer. Data acquisition systems are used to perform a variety of functions, including laboratory research, process monitoring and control, data logging, analytical chemistry, tests and analysis of physical phenomena, and control of mechanical or electrical machinery. Data recorders are used in a wide variety of applications for imprinting various types of forms, and documents. Data collection systems or data loggers generally include memory chips or strip charts for electronic recording, probes or sensors which measure product environmental parameters and are connected to the data logger. Hand-held portable data collection systems permit in field data collection for up-to-date information processing.

Source

Data acquisition begins with the physical phenomenon or physical property to be measured. Examples of this include temperature, light intensity, gas pressure, fluid flow, and force. Regardless of the type of physical property to be measured, the physical state that is to be measured must first be transformed into a unified form that can be sampled by a data acquisition system. The task of performing such transformations falls on devices called sensors.

A sensor, which is a type of transducer, is a device that converts a physical property into a corresponding electrical signal (e.g., a voltage or current) or, in many cases, into a corresponding electrical characteristic (e.g., resistance or capacitance) that can easily be converted to electrical signal.The ability of a data acquisition system to measure differing properties depends on having sensors that are suited to detect the various properties to be measured. There are specific sensors for many different applications. DAQ systems also employ various signal conditioning techniques to adequately modify various different electrical signals into voltage that can then be digitized using an Analog-to-digital converter (ADC).



Signals

Signals may be digital (also called logic signals sometimes) or analog depending on the transducer used.Signal conditioning may be necessary if the signal from the transducer is not suitable for the DAQ hardware being used. The signal may need to be amplified, filtered or demodulated. Various other examples of signal conditioning might be bridge completion, providing current or voltage excitation to the sensor, isolation, linearization. For transmission purposes, single ended analog signals, which are more susceptible to noise can be converted to differential signals. Once digitized, the signal can be encoded to reduce and correct transmission errors.

DAQ hardware

DAQ hardware is what usually interfaces between the signal and a PC. It could be in the form of modules that can be connected to the computer's ports (parallel, serial, USB, etc.) or cards connected to slots (S-100 bus, Apple Bus, ISA, MCA, PCI, PCI-E, etc.) in the mother board. Usually the space on the back of a PCI card is too small for all the connections needed, so an external breakout box is required. The cable between this box and the PC can be expensive due to the many wires, and the required shielding.

DAQ cards often contain multiple components (multiplexer, ADC, DAC, TTL-IO, high speed timers, RAM). These are accessible via a bus by a microcontroller, which can run small programs. A controller is more flexible than a hard wired logic, yet cheaper than a CPU so that it is alright to block it with simple polling loops. For example: Waiting for a trigger, starting the ADC, looking up the time, waiting for the ADC to finish, move value to RAM, switch multiplexer, get TTL input, let DAC proceed with voltage ramp. Many times reconfigurable logic is used to achieve high speed for specific tasks and Digital signal processors are used after the data has been acquired to obtain some results. The fixed connection with the PC allows for comfortable compilation and debugging. Using an external housing a modular design with slots in a bus can grow with the needs of the user.Not all DAQ hardware has to run permanently connected to a PC, for example intelligent stand-alone loggers and oscilloscopes, which can be operated from a PC, yet they can operate completely independent of the PC.

DAQ software

DAQ software is needed in order for the DAQ hardware to work with a PC. The device driver performs low-level register writes and reads on the hardware, while exposing a standard API for developing user applications. A standard API such as COMEDI allows the same user applications to run on different operating systems, e.g. a user application that runs on Windows will also run on Linux and BSD.

Multiplexing

In telecommunications and computer networks, multiplexing (also known as muxing) is a process where multiple analog message signals or digital data streams are combined into one signal over a shared medium. The aim is to share an expensive resource. For example, in telecommunications, several phone calls may be transferred using one wire. It originated in telegraphy, and is now widely applied in communications.The multiplexed signal is transmitted over a communication channel, which may be a physical transmission medium. The multiplexing divides the capacity of the low-level communication channel into several higher-level logical channels, one for each message signal or data stream to be transferred. A reverse process, known as demultiplexing, can extract the original channels on the receiver side.A device that performs the multiplexing is called a multiplexer (MUX), and a device that performs the reverse process is called a demultiplexer (DEMUX).Inverse multiplexing (IMUX) has the opposite aim as multiplexing, namely to break one data stream into several streams, transfer them simultaneously over several communication channels, and recreate the original data stream.

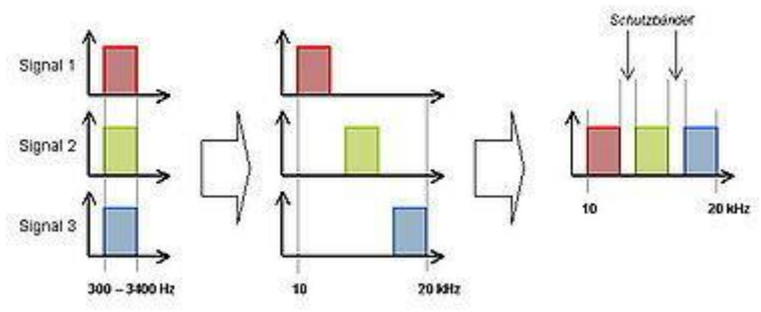
Types of multiplexing

Multiplexing technologies may be divided into several types, all of which have significant variations: space-division multiplexing (SDM), frequency-division multiplexing (FDM), time-division multiplexing (TDM), and code division multiplexing (CDM). Variable bit rate digital bit streams may be transferred efficiently over a fixed bandwidth channel by means of statistical multiplexing, for example packet mode communication. Packet mode communication is an asynchronous mode time-domain multiplexing which resembles time-division multiplexing.
Digital bit streams can be transferred over an analog channel by means of code-division multiplexing (CDM) techniques such as frequency-hopping spread spectrum (FHSS) and direct-sequence spread spectrum (DSSS).In wireless communications, multiplexing can also be accomplished through alternating polarization (horizontal/vertical or clockwise/counterclockwise) on each adjacent channel and satellite, or through phased multi-antenna array combined with a Multiple-input multiple-output communications (MIMO) scheme.

Space-division multiplexing

In wired communication, space-division multiplexing simply implies different point-to-point wires for different channels. Examples include an analogue stereo audio cable, with one pair of wires for the left channel and another for the right channel, and a multipair telephone cable. Another example is a switched star network such as the analog telephone access network (although inside the telephone exchange or between the exchanges, other multiplexing techniques are typically employed) or a switched Ethernet network. A third example is a mesh network. Wired space-division multiplexing is typically not considered as multiplexing.In wireless communication, space-division multiplexing is achieved by multiple antenna elements forming a phased array antenna. Examples are multiple-input and multiple-output (MIMO), single-input and multiple-output (SIMO) and multiple-input and single-output (MISO) multiplexing. For example, a IEEE 802.11n wireless router with N antennas makes it possible to communicate with N multiplexed channels, each with a peak bit rate of 54 Mbit/s, thus increasing the total peak bit rate with a factor N. Different antennas would give different multi-path propagation (echo) signatures, making it possible for digital signal processing techniques to separate different signals from each other. These techniques may also be utilized for space diversity (improved robustness to fading) or beamforming (improved selectivity) rather than multiplexing.

Frequency-division multiplexing



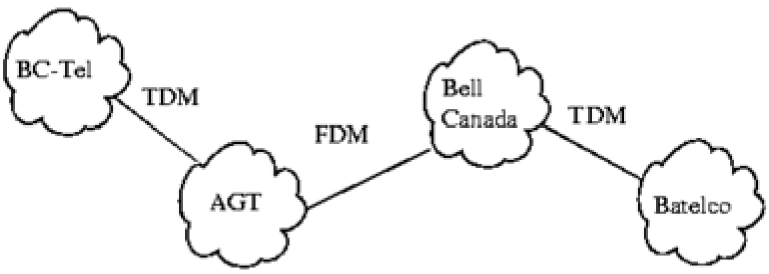
Frequency-division multiplexing (FDM): The spectrums of each input signal are swifted in several distinct frequency ranges.Frequency-division multiplexing (FDM) is inherently an analog technology. FDM achieves the combining of several digital signals into one medium by sending signals in several distinct frequency ranges over that medium.One of FDM's most common applications is cable television. Only one cable reaches a customer's home but the service provider can send multiple television channels or signals simultaneously over that cable to all subscribers. Receivers must tune to the appropriate frequency (channel) to access the desired signal.A variant technology, called wavelength-division multiplexing (WDM) is used in optical communications.

Time-division multiplexing

Time-division multiplexing (TDM) is a digital technology. TDM involves sequencing groups of a few bits or bytes from each individual input stream, one after the other, and in such a way that they can be associated with the appropriate receiver. If done sufficiently and quickly, the receiving devices will not detect that some of the circuit time was used to serve another logical communication path.Consider an application requiring four terminals at an airport to reach a central computer. Each terminal communicated at 2400 bps, so rather than acquire four individual circuits to carry such a low-speed transmission, the airline has installed a pair of multiplexers.

Code-division multiplexing

Code division multiplexing (CDM) is a technique in which each channel transmits its bits as a coded channel-specific sequence of pulses. This coded transmission typically is accomplished by transmitting a unique time-dependent series of short pulses, which are placed within chip times within the larger bit time. All channels, each with a different code, can be transmitted on the same fiber and asynchronously demultiplexed. Other widely used multiple access techniques are Time Division Multiple Access (TDMA) and Frequency Division Multiple Access (FDMA).Code Division Multiplex techniques are used as an access technology, namely Code Division Multiple Access (CDMA), in Universal Mobile Telecommunications System (UMTS) standard for the third generation (3G) mobile communication identified by the ITU. Another important application of the CDMA is the Global Positioning System (GPS).However, the term Code Division Multiple access (CDMA) is also widely used to refer to a group of specific implementations of CDMA defined by Qualcomm for use in digital cellular telephony, which include IS-95 and IS-2000. The two different uses of this term can be confusing. Actually, CDMA (the Qualcomm standard) and UMTS have been competing for adoption in many markets.



Relation to multiple access.A multiplexing technique may be further extended into a multiple access method or channel access method, for example TDM into Time-division multiple access (TDMA) and statistical multiplexing into carrier sense multiple access (CSMA). A multiple access method makes it possible for several transmitters connected to the same physical medium to share its capacity.Multiplexing is provided by the Physical Layer of the OSI model, while multiple access also involves a media access control protocol, which is part of the Data Link Layer.The Transport layer in the OSI model as well as TCP/IP model provides statistical multiplexing of several application layer data flows to/from the same computer.

Application areas

Telegraphy

The earliest communication technology using electrical wires, and therefore sharing an interest in the economies afforded by multiplexing, was the electric telegraph. Early experiments allowed two separate messages to travel in opposite directions simultaneously, first using an electric battery at both ends, then at only one end.

- Émile Baudot developed a time-multiplexing system of multiple Hughes machines in the 1870s.
- In 1874, the quadruplex telegraph developed by Thomas Edison transmitted two messages in each direction simultaneously, for a total of four messages transiting the same wire at the same time.
- Several workers were investigating acoustic telegraphy, a frequency-division multiplexing technique, which led to the invention of the telephone.

Telephony

In telephony, a customer's telephone line now typically ends at the remote concentrator box down the street, where it is multiplexed along with other telephone lines for that neighborhood or other similar area. The multiplexed signal is then carried to the central switching office on significantly fewer wires and for much further distances than a customer's line can practically go. This is likewise also true for digital subscriber lines (DSL).Fiber in the loop (FITL) is a common method of multiplexing, which uses optical fiber as the backbone. It not only connects POTS phone lines with the rest of the PSTN, but also replaces DSL by connecting directly to Ethernet wired into the home. Asynchronous Transfer Mode is often the communications protocol used.Because all of the phone (and data) lines have been clumped together, none of them can be accessed except through a demultiplexer. This provides for more-secure communications, though they are not typically encrypted.The concept is also now used in cable TV, which is increasingly offering the same services as telephone companies. IPTV also depends on multiplexing.

Video processing

In video editing and processing systems, multiplexing refers to the process of interleaving audio and video into one coherent MPEG transport stream (time-division multiplexing).In digital video, such a transport stream is normally a feature of a container format which may include metadata and other information, such as subtitles. The audio and video streams may have variable bit rate. Software that produces such a transport stream and/or container is commonly called a statistical multiplexor or muxer. A demuxer is software that extracts or otherwise makes available for separate processing the components of such a stream or container.

Digital broadcasting

In digital television and digital radio systems, several variable bit-rate data streams are multiplexed together to a fixed bitrate transport stream by means of statistical multiplexing. This makes it possible to transfer several video and audio channels simultaneously over the same frequency channel, together with various services.In the digital television systems, this may involve several standard definition television (SDTV) programmes (particularly on DVB-T, DVB-S2, ISDB and ATSC-C), or one HDTV, possibly with a single SDTV companion channel over one 6 to 8 MHz-wide TV channel. The device that accomplishes this is called a statistical multiplexer. In several of these systems, the multiplexing results in an MPEG transport stream. The newer DVB standards DVB-S2 and DVB-T2 has the capacity to carry several HDTV channels in one multiplex. Even the original DVB standards can carry more HDTV channels in a multiplex if the most advanced MPEG-4 compressions hardware is used.On communications satellites which carry broadcast television networks and radio networks, this is known as multiple channel per carrier or MCPC. Where multiplexing is not practical (such as where there are different sources using a single transponder), single channel per carrier mode is used.Signal multiplexing of satellite TV and radio channels is typically carried out in a central signal playout and uplinkcentre, such as ASTRA Platform Services in Germany, which provides playout, digital archiving, encryption, and satellite uplinks, as well as multiplexing, for hundreds of digital TV and radio channels.In digital radio, both the Eureka 147 system of digital audio broadcasting and the in-band on- channel HD Radio, FMeXtra, and Digital Radio Mondiale systems can multiplex channels. This is essentially required with DAB-type transmissions (where a multiplex is called an ensemble), but is entirely optional with IBOC systems.

Analog broadcasting

In FM broadcasting and other analog radio media, multiplexing is a term commonly given to the process of adding subcarriers to the audio signal before it enters the transmitter, where modulation occurs. Multiplexing in this sense is sometimes known as MPX, which in turn is also an old term for stereophonic FM, seen on stereo systems since the 1960s.

IEEE-488 Bus:

IEEE-488 is a short-range digital communications bus specification. It was created for use with automated test equipment in the late 1960s, and is still in use for that purpose. IEEE-488 was created as HP-IB (Hewlett-Packard Interface Bus), and is commonly called GPIB (General Purpose Interface Bus). It has been the subject of several standards.

Characteristics

IEEE-488 is an 8-bit, electrically parallel bus. The bus employs sixteen signal lines — eight used for bi-directional data transfer, three for handshake, and five for bus management — plus eight ground return lines.Every device on the bus has a unique 5-bit primary address, in the range from 0 to 30 (31 total possible addresses).The standard allows up to 15 devices to share a single physical bus of up to 20 meters total cable length. The physical topology can be linear or star (forked). Active extenders allow longer buses, with up to 31 devices theoretically possible on a logical bus.Control and data transfer functions are logically separate; a controller can address one device as a –talker|| and one or more devices as –listeners|| without having to participate in the data transfer. It is possible for multiple controllers to share the same bus; but only one can be the "Controller In Charge" at a time In the original protocol, transfers use an interlocked, three-wire ready–valid–accepted handshake. The maximum data rate is about one Mbyte/s. The later HS-488 extension relaxes the handshake requirements, allowing up to 8 Mbyte/s. The slowest participating device determines the speed of the bus.

Use as a computer interface

HP's designers did not specifically plan for IEEE-488 to be a peripheral interface for general-purpose computers; the focus was on instrumentation. But when HP's early microcomputers needed an interface for peripherals (disk drives, tape drives, printers, plotters, etc.), HP-IB was readily available and easily adapted to the purpose.HP computer products which used HP-IB included the HP series 80, HP 9800 series, the HP 2100 series, and the HP 3000 series. Some of HP's advanced pocket calculators of the 1980s, such asthe HP-41 and HP-71B series, also had IEEE-488 capabilities, via an optional HP-IL/HP-IB interface module.Other manufacturers adopted GPIB for their computers as well, such as with the Tektronix 405x line.The Commodore PET (introduced 1977) range of personal computers connected their peripherals using the IEEE-488 bus, but with a non-standard card edge connector. Commodore's following 8- bit machines, including the VIC-20, C-64, and C-128, utilized an unrelated, proprietary serial interface, using a round DIN connector, for which they retained the IEEE-488 programming interface and terminology, however.

Advantages and disadvantages

Advantages

- Simple hardware interface
- Ease of connecting multiple device to a single host
- Allows mixing of slow and fast devices
- Well-established and mature, widely supported

Disadvantages

- Mechanically bulky connectors and cables
- Limited speed and expansion
- Lack of command protocol standards (before SCPI)
- Implementation options (e.g. end of transmission handling) can complicate interoperability in pre-IEEE-488.2 devices
- No mandatory galvanic isolation between bus and devices
- High cost (compared to RS-232/USB/Firewire/Ethernet)
- Limited availability (again compared to RS-232/USB/Firewire/Ethernet)

Optical time-domain reflectometer

An optical time-domain reflectometer (OTDR) is an optoelectronic instrument used to characterize an optical fiber. An OTDR injects a series of optical pulses into the fiber under test. It also extracts, from the same end of the fiber, light that is scattered (Rayleigh Backscatter) or reflected back from points along the fiber. (This is equivalent to the way that an electronic time-domain reflectometer measures reflections caused by changes in the impedance of the cable under test.) The strength of the return pulses is measured and integrated as a function of time, and is plotted as a function of fiber length.

An OTDR may be used for estimating the fiber's length and overall attenuation, including splice and mated-connector losses. It may also be used to locate faults, such as breaks, and to measure optical return loss. To measure the attenuation of multiple fibers, it is advisable to test from each end and then average the results, however this considerable extra work is contrary to the common claim that testing can be perfomed from only one end of the fiber.

In addition to required specialized optics and electronics, OTDRs have significant computing ability and a graphical display, so they may provide significant test automation. However, proper instrument operation and interpretation of an OTDR trace still requires special technical training and experience.

OTDRs are commonly used to characterize the loss and length of fibers as they go from initial manufacture, through to cabling, warehousing while wound on a drum, installation and then splicing. The last application of installation testing, is more challenging, since this can be over extremely long distances, or multiple splices spaced at short distances, or fibers with different optical characteristics joined together. OTDR test results are often carefully stored in case of later fiber failure or warranty claims. Fiber failures can be very expensive, both in terms of the direct cost of repair, and consequential loss of service.

OTDRs are also commonly used for fault finding on installed systems. In this case, reference to the installation OTDR trace is very useful, to determine where changes have occurred. Use of an OTDR for fault finding may require an experienced operator who is able to correctly judge the appropriate instrument settings to locate a problem accurately. This is particularly so in cases involving long distance, closely spaced splices or connectors, or PONs.

OTDRs are available with a variety of fiber types and wavelengths, to match common applications. In general, OTDR testing at longer wavelengths such as 1550 nm or 1625 nm, can be used to identify fiber attenuation caused by fiber problems, as opposed to the more common splice or connector losses.

The optical dynamic range of an OTDR is limited by a combination of optical pulse output power, optical pulse width, input sensitivity, and signal integration time. Higher optical pulse output power, and better input sensitivity, combine directly to improve measuring range, and are usually fixed features of a particular instrument. However optical pulse width and signal integration time are user adjustable, and require trade-offs which make them application specific.

A longer laser pulse improves dynamic range and attenuation measurement resolution at the expense of distance resolution. For example, using a long pulse length, it may possible to measure attenuation over a distance of more than 100 km, however in this case an optical event may appear to be over 1 km long. This scenario is useful for overall characterisation of a link, but would be of much less use when trying to locate faults. A short pulse length will improve distance resolution of optical events, but will also reduce measuring range and attenuation measurement resolution

The OTDR "dead zone" is a topic of much interest to users. Dead zone is classified in two ways.

Firstly, an "Event Dead Zone" is related to a reflective discrete optical event. In this situation, the measured dead zone will depend on a combination of the pulse length (see table), and the size of the reflection. Secondly, an "Attenuation Dead Zone" is related to a non-reflective event. In this situation, the measured dead zone will depend on a combination of the pulse length (see table).

A long signal integration time effectively increases OTDR sensitivity by averaging the receiver output. The sensitivity increases with the square root of the integration time. So if the integration time is increased by 16 times, the sensitivity increases by a factor of 4. This imposes a sensitivity practical limit, with integration times of seconds to a few minutes.

The dynamic range of an OTDR is usually specified as the attenuation level where the measured signal gets lost in the detection noise level, for a particular combination of pulse length and signal integration time. This number is easy to deduce by inspection of the output trace, and is useful for comparison, but is not very useful in practice, since at this point the measured values are random. So the practical measuring range is smaller, depending on required attenuation measurement resolution.

When an OTDR is used to measure the attenuation of multiple joined fiber lengths, the output trace can incorrectly show a joint as having gain, instead of loss. The reason for this is that adjacent fibers may have different backscatter coefficients, so the second fiber reflects more light than the first fiber, with the same amount of light travelling through it. If the OTDR is placed at the other end of this same fiber pair, it will measure an abnormally high loss at that joint. However if the two signals are then combined, the correct loss will be obtained. For this reason, it is common OTDR practice to measure and combine the loss from both ends of a link, so that the loss of cable joints, and end to end loss, can be more accurately measured.

The theoretical distance measuring accuracy of an OTDR is extremely good, since it is based on software and a crystal clock with an inherent accuracy of better than 0.01%. This aspect does not need subsequent calibration since practical cable length measuring accuracy is typically limited to about 1% due to: The cable length is not the same as the fiber length, the speed of light in the fiber is known with limited accuracy (the refractive index is only specified to 3 significant figures such as e.g. 1.45 etc.), and cable length markers have limited accuracy (0.5% - 1%).

An OTDR excels at identifying the existence of unacceptable point loss or return loss in cables. It's ability to accurately measure absolute end-to-end cable loss or return loss can be quite poor, so cable acceptance ususally includes an end to end test with a light source and power meter, and optical return loss meter. It's ability to exactly locate a hidden cable fault is also limited, so for fault finding it may be augmented with other localised tools such as a red laser fault locator, clip-on identifier, or "Cold Clamp" optical cable marker.

Question Bank

UNIT-V

DATA ACQUISITION SYSTEMS AND FIBER OPTIC MEASUREMENTS

PART-A (2 Marks)

1. **What is meant by data acquisition? (AU NOV 2008,2009)**

The system used for data processing, data conversion, data transmission, data storage is called data acquisition system.

2. **What are the objectives of Data Acquisition system?**
   - The data acquisition system must acquire the necessary data at the correct time.
   - It must use all the data efficiently to inform the operator about the state of the plant.
   - It must monitor the operation of complete plant so that optimum online safe operations are maintained.
   - It must provide effective human communication system.
   - It must be able to collect, summarise and store data properly for diagnosis and record purpose of any operation.
3. **What is Multiplexing? (AU APRIL 2006, 2009)**

   Multiplexing means combining different signals. In data processing and handling it is the frequently required to combine number of analog signals into a single digital channel. Both digital signals and analog voltages can be multiplexed.
4. **What are the three basic requirements of a computer controlled systems? (AU APRIL 2003)**
5. **What is an IEE 488 bus system? (AU APRIL 2004)**
6. **What is meant by IEE 488 standard and GPIB? (AU NOV 2006)**
7. **What is Optical Time Domain Reflectometer? (AU NOV 2007)**

PART-B (16 Marks)

1. Explain the generalized block schematic of a Digital Data Acquisition system.
2. (16 marks)) (AU NOV 2008)
3. Explain the various techniques of multiplexing? (16 marks) (AU NOV 2006,2009)
4. Explain the block diagram of optical time domain reflectometer. (16 marks) (AU APRIL 2007, NOV 2009)
5. Write short notes on IEEE 488 bus. (16 marks) (AU APRIL 2010, NOV 2007)
6. Explain microprocessor based measurement. (8 marks) (AU APRIL 2005, NOV 2005)
7. Write short notes on VI (Virtual Instrumentation)

Anna University sample Question Paper

B.E./B.Tech. DEGREE EXAMINATION, APRIL/MAY 2011

Sixth Semester

Electronics and Communication Engineering

EC 2351 — MEASUREMENTS AND INSTRUMENTATION

(Regulation 2008)

Time : Three hours Maximum : 100 marks

Answer ALL questions

PART A — (10 × 2 = 20 marks)

1. Mention the significance of measurements.
2. Compare Moving coil with Moving iron instruments.
3. Draw the internal structure of CRT and list its functions.
4. What are the two significant problems with diodes when used for RF rectification?
5. What is Barkhausen Criteria for sustained oscillation?
6. Draw the block diagram of spectrum analyzer.
7. What are the advantages of digital instruments over analog instruments?
8. What are the different types of Digital Voltmeter?
9. Draw the block diagram of Digital Data Acquisition System.
10. What are the key features of fully automatic digital instruments?

PART B — (5 × 16 = 80 marks)

11. (a) (i) What is the need for standards of measurements? How they are classified? Explain (8)

(ii) How the unknown frequency is measured using Wein bridge method? (8)

Or

(b) (i) What are the different types of errors in measurement? Explain. (8)

(ii) How do you measure the unknown inductance using Hay Bridge? (8)

12. (a) (i) Draw the block diagram of sampling oscilloscope and explain the principle. (8)

(ii) Explain the measurement of quality factor of a coil. (8)

Or

(b) (i) Discuss the measurement of DC and AC voltages and currentsusing an Electronic Multimeter. (8)

(ii) Draw the block diagram of True RMS reading voltmeter and explain its operation. (8)

13. (a) (i) Explain how function generator generates sine wave, triangular wave and square wave. (8)

(ii) Draw the block diagram of sweep-frequency generator and explain.(8)

Or

(b) (i) What is wave analyzer? How it analyzes the harmonics? Explain. (8)
(ii) Explain the vector network analyzer and list its application. (8)

14. (a) (i) How computer controlled measurement system is used for testing radio receiver? (8)
(ii) What is virtual instrument? List the advantages of virtual instrument over conventional instrument (8)

Or

(b) (i) With necessary diagrams explain Ramp type digital voltmeter. (8)
(ii) Draw the block diagram of digital frequency meter and explain. (8)

15. (a) (i) What are the factors to be considered while interfacing transducers to electronic control and measuring systems? (8)
(ii) Draw the block schematic representation of the IEEE 488 instrumentation bus and explain. (8)

Or

(b) (i) Explain the optical time domain reflectometer with a neat diagram.(8)
(ii) Write a detailed note on data loggers. (8)

B.E/B.Tech DEGREE EXAMINATION NOVEMBER/DECEMBER 2007

Sixth Semester
Electrical and Electronics Engineering

EI 1361 --MEASUREMENTS AND INSTRUMENTATION

(Regulation 2004)

Time:Three hours Maximum marks:100

Answer ALL questions

PART A (10 x 2=20 marks)

1. Distinguish between the direct and indirect methods of measurements.

2. With one example explain "Instrumental Errors"

3. Explain the principle of analog type electrical instruments.

4. How a PMMC meter can be used as voltmeter and ammeter?

5. Draw Maxwell's AC Bridge and give the balance equation interms of resistance.

6. Explain any two technical parameters to be consider in grounding.

7. Explain the characteristics of Time domain output device using in measurements.

8. Explain the following term as applied to digital displays.
3½ digit and 4½ digit displays.
9. Define inverse transducer with example.
10. Explain the principle of piezoelectric transducers and name any two piezoelectric materials.

PART B (5 x 16 =80)

11 (i) What are the basic blocks of a generalized instrumentation system. Draw the various blocks and explain their functions. (10)

(ii) Explain in detail calibration technique and draw the calibration curve in general. (6) (or)
(b) (i) Discuss in detail various types errors associated in measurement and how these errors can be minimized? (10)

(ii)Define the following terms in the context of normal frequency distribution of data (6)
(1)Mean value (2)Standard Deviation (3)Average(4) Variance

12. (a) Describe the construction, principle of working and applications of synchro transformers. (or)

(b) Discuss why is it necessary to carry out frequency domain analysis of measurement systems? What are the two plots obtained when the frequency response of a system is carried out? (16)

13 (a) Explain voltage sensitive self balancing bridge, and derive the bridge sensitivity of voltage sensitive bridge with fundamentals. (16)

(or)

(b) (i) With fundamentals distinguish between DC and AC potentiometers, and give any two specific applications for each.(8)

(ii) Discuss the advantages and limitations of electromagnetic interference in measurements. (8)

14. (a) Describe the construction and working of LCDs, mention the difference between light scattering and field effect types of LCDs, also explain the advantages of LCDs. (16)

(or)

(b) (i) What is an XY recorders? How do you distinguish is from X - t and Y - t recorders? (8)

(ii) Describe the pulse duration modulation (PDM) as used in magnetic tape recording and explain its merits and demerits. (8)

15. (a) (i) Describe the different criteria for selection transducers for a particular application. (8) (ii) Explain the different principles of working of capacitive transducers. (8)

O
r

(b) (i) How is a differential output taken from an inductive transducer?Explain the advantages when inductive transducers are used in push - pull configuration. (8)

(ii) Describe in detail the successive approximation method of analog to digital conversion. (8)

B.E./B.Tech. DEGREE EXAMINATION, MAY/JUNE 2010.

Sixth Semester
Electronics and Communication engineering
EC1255-MEASUREMENTS ANDINSTRUMENTAION

(Regulation 2004)
Time:3hrs Maximum:100marks

Answer ALL questions.

PART A - (10X2=20 marks)

1. What are the different types of standard of measurement?
2. What is a transfer instrument?

3. How is the electron beam focused to a fine spot on the face of the cathode ray tube?

4. List the disadvantages of storage cathode ray tube.

5. Give the functions of an attenuator in a signal generator.

6. What are the draw backs of tuned circuit analyzers?

7. What is the importance of gate time in frequency counter?

8. How is trigger time error reduced?

9. Distinguish between analog and digital data acquisition systems.

10. How much elapsed time would occur to a reflection from a break in an optical fibre of 1.4km if the index of refraction of the core was 1.55?

PART B - (16X5=80 marks)

11.(a)(i)With a neat diagram explain in detail the construction of PMMC instrument.(8) (ii)How do you measure large currents in PMMC instruments?(4)

(iii)What is Ayrton shunt?(4)

Or

(b)(i)Discuss in detail about Kelvin double bridge.(8) (ii)With a neat diagram explain in detail about Hay bridge.(8)

12.(a)With a neat diagram explain in detail about

(i)Ramp type DVM(8)

(ii)Successive approximation DVM.(8)

Or

(b)With a neat block diagram explain in detail about vector impedance meter.(16)

13.(a)With a neat block diagram explain in detail about the frequency divider typer of signal generator with frequency modulation.(16)

Or

(b)Explain:

(i)General purpose spectrum analyzer.(8)

(ii)Phase locked circuit for the first local oscillator of spectrum analyzer.(8)

14.(a)(i)What method can be used to increase the frequency range of frequency counter.(8) (ii)How can this be achieved without degrading the accuracy of the counter.(8)

Or

(b)Discuss in detail about

(i)Gating error

(ii)Time-base error and trigger level error.

(iii)Maximum accuracy achieved for frequency measurements.

15.(a)(i)What are the requirements of an automatic test system? (ii)Explain in detail about IEEE 488 system.

Or

(b)With a neat block diagram explain

(i)Optical power meter (ii) Auto ranging power meter

(iii) Optical time-domain reflectometer.

B.E./B.Tech. DEGREE EXAMINATION,
NOVEMBER/DECEMBER 2010
Third emester
Electrical and Electronics Engineering
EE 2201 — MEASUREMENTS AND
INSTRUMENTATION (Regulation 2008)

Time : Three hours Maximum :
100 Marks

Answer ALL
questions

PART A — (10 × 2 = 20
Marks)

1. Define static characteristics of an instrument.
2. What is meant by absolute error of measurement?
3. Why are the ordinary watt-meters not suitable for low power factor circuits?
4. What is a phase sequence indicator?
5. List the application of D.C. potentiometers.
6. What are parasitic voltages and how are they eliminated?
7. What is the purpose of a Post Deflection Acceleration (PDA) in a CRT?
8. Differentiate between LED and LCD.
9. What are the classifications of encoder?
10. What is the need of sample and hold circuit in A/D converter?

PART B — (5 × 16 = 80 Marks)

11. (a) (i) Draw the block diagram of functional elements of measuring system and explain the function of each block. (Marks 8)

(ii) Explain the different types of errors in measurements. (8)

Or

(b) (i) The probable values of two resistors and their S.D are specified as R1 – 18.62 , S.D = 0.025 , R2 = 74.48 , S.D. = 0.05 . Find the probable value and S.D for the two resistors when they are connected in
(1) Series and
(2) Parallel. (Marks 8)

(ii) Discuss the different types of standards of measurements. (Marks 8)

12. (a) (i) What are the various types of digital voltmeters? With a neat sketch explain the working principle of any one type of a digital voltmeter. (Marks 8)

(ii) With a neat diagram explain the construction and its working principle of elector dynamo-meter type watt-meter. Also derive its torque equation. (Marks 8)

Or

(b) (i) Explain the method of measurements of B.H curve of a ring specimen with a neat diagram. (Marks 8)

(ii) Describe the construction and working principle of digital frequency meter. (Marks 8)

13. (a) (i) Draw a neat sketch of a modern slide-wire D.C potentiometer and discuss how the potentiometer is standardized. (Marks 8)

(ii) Describe how co-ordinate type potentiometer can be used for calibration of a voltmeter and A.C energy meters. (Marks 8)

Or

(b) (i) Explain the theory and working principle of Kelvin's double bridge method for measurement of low resistance. Derive the relation for finding unknown resistance. (Marks 8)

(ii) Discuss briefly how Hay's Bridge can be used for the measurement of inductance. (Marks 8)

14. (a) (i) Explain the construction and its working principle of X-Y Recorder. (6)

(ii) Briefly discuss the features of digital plotters and printers. (Marks 10) Or

(b) (i) Explain the working principle of electrostatic deflection system in a CRT. (10)

(ii) Explain the working principle of digital storage oscilloscope. (Marks 6)

15. (a) (i) Explain the construction and working of unbounded and bonded type strain gauges. (8)

(ii) Explain the construction and working of optical encoders with a neat diagram. (Marks 8)

Or

(b) (i) Draw the generalized block diagram of a digital data acquisition system and explain. (8)

(ii) Explain the successive approximation method of A/D converter. (8)

Reg. No. :

Question Paper Code : 21370

B.E./B.Tech. DEGREE EXAMINATION, MAY/JUNE 2013.

Sixth Semester

Electronics and Communication engineering

EC 2351/EC 61/10144/EC 602 – MEASUREMENTS AND INSTRUMENTATION

(Regulation 2008/2010)

Time : Three hours | Maximum : 100 marks

Answer ALL questions

PART A – (10 X 2 = 20 marks)

1. List out the various standards of measurements.
2. Mention the errors in moving coil meters.
3. Prepare the comparison table between analog and digital storage oscilloscope.
4. Write a short note on true RMS meters.
5. How do you measure the resistance values in digital RLC meters?
6. What is meant by network analyzer?
7. Define automatic ranging.
8. Write a short note on digital voltmeter.
9. List out the drawbacks of reflectometer.
10. Define the term transducer.

PART B – (5 X 16 = 80 marks)

11. (a) With neat circuit diagrams describe in detail about the following bridge measurement system.

(i) Maxwell bridge (8)

(ii) Wien bridge. (8)

Or

(b) (i) Explain in detail about the various error measurement systems with statistical analysis. (8)

(ii) Describe in detail about the moving iron meters with suitable example. (8)

12. (a) Discuss in detail about the function of delay time base oscilloscopes with neat diagram. (16)

Or

(b) With neat diagram explain in detail about the function of following measurement system.

(i) Vector meter (8)

(ii) Q meter (8)

13. (a) Explain the operations of RF signal and sweep generators.

Or

(b) Explain with neat diagrams , the working of the following :

(i) Spectrum analyzer (8)

(ii) Frequency synthesizer. (8)

14. (a) Discuss in detail about the computer controlled fully automatic digital instruments with test systems.

Or

(b) (i) Enumerate the measurement system of frequency and time intervals in a particular range of signals. (8)

(ii) Discuss in detail about the digital multimeter. (8)

15. (a) With the neat diagram , explain the working of IEEE 488 bus , operations and characteristics.

Or

(b) Draw and explain the block diagram of analog and digital data acquisition system.

Reg. No. : | | | | | | | | | | | |

Question Paper Code : 11340

B.E./B.Tech. DEGREE EXAMINATION, NOVEMBER/DECEMBER 2012.

Sixth Semester

Electronics and Communication Engineering

EC 2351/EC 61/EI 1306 — MEASUREMENTS AND INSTRUMENTATION

(Regulation 2008)

Time : Three hours Maximum : 100 marks

Answer ALL questions.

PART A — ($10 \times 2 = 20$ marks)

1. A set of independent current measurements were recorded as 10.03, 10.10, 10.11 and 10.08 A. Calculate the range of an error.
2. How is the international standard of length defined?
3. Compare and contrast analog and digital storage oscilloscopes.
4. Distributed capacitance of a coil is measured by changing the capacitance of the tuning capacitor. The value of the tuning capacitor are C_1 and C_2 for the resonant frequencies f_1 and $2f_1$. What is the value of the distributed capacitance?
5. In a sweep frequency generator, two oscillators one with frequency range of 3 GHz to 5 GHz is hetrodyned with a second oscillator having a fixed frequency output of 3 GHz. How the output frequency varies?
6. What is intermodulation distortion?
7. Why Schmitt trigger is used in digital frequency meter?
8. Draw the block diagram of integrating type DVM.
9. List the elements of digital data acquisition system.
10. What is the need for data loggers?

PART B — ($5 \times 16 = 80$ marks)

11. (a) (i) How to convert the PMMC meter into a voltmeter and ammeter? How to extend the range of these meters? (8)

(ii) Explain the types of error with an example. (8)

Or

(b) (i) What are the conditions for bridge balance? (8)

(ii) How to measure the unknown inductance using Maxwell's LC Bridge? Draw the phasor diagram also. (8)

12. (a) (i) Draw the block diagram of the sampling oscilloscope. How does the sampling oscilloscope increase the apparent frequency response of an oscilloscope? (8)

(ii) How to measure large capacitors and small coils using Q-meters. (8)

Or

(b) (i) Explain the vector impedance meter with a neat block diagram. (8)

(ii) How to measure the RF voltage and power using RF millivoltmeter? (8)

13. (a) (i) Draw the block diagram of the frequency divider type of signal generator with frequency modulation and explain. (8)

(ii) What are the basic elements of function generator? Explain how to generate the square wave, triangular wave and sine wave using function generator. (8)

Or

(b) (i) Explain the working of frequency selective wave analyzer with neat block diagram. (8)

(ii) How the fundamental frequency is suppressed using the fundamental suppression distortion analyzer? (8)

14. (a) (i) Draw the block diagram of a multiplexed display used in a frequency counter and explain. (8)

(ii) Explain how to extend the frequency range of the counter. (8)

Or

(b) (i) How to make automatic polarity indication and automatic ranging in a digital instrument? (8)

(ii) Explain the need for virtual instrument with an example. (8)

15. (a) (i) Draw the schematic of an isolation amplifier and explain the need for isolation amplifier in interfacing transducers. (8)

(ii) With neat diagrams explain the digital to analog multiplexing. (8)

Or

(b) (i) Explain the IEEE 488 electrical interface system. (8)

(ii) How to measure the power using optical instrument? Draw the auto ranging power meter and explain. (8)

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