模拟电子技术 (Analogue Electronics)

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- Delivery: 90 minutes, 08:00 – 09:50, 10 minute break, 08:55 – 09:05 Beijing time, Monday – Friday 27/05/2024 – 14:06/2024
- Learning outcomes:

On successful completion of this module a student will be able to:

1 Design analogue electronic circuits using a variety of techniques.

2 Understand the theory of operation of the main components used in analogue electronic systems.

3 Analyse and design amplifiers, op amps circuits and filters.

4 Implement the principles of feedback theory and the operation of oscillators.

How to obtain these notes...

- You can download from

http://dfdn.info/teaching/analouge-electronics/

Indicative content

- Problem-based learning and research informed teaching approaches will be used throughout the Module to engage and develop students' abilities. The following main topics will be covered:

- 1.Theory of operation of the main components used in analogue electronic circuits and systems including diodes, transistors and op amps.
- 2. Amplification and the design of amplifiers and their different classes.
- 3. Analysis and design of analogue filters, feedback theory and oscillators.
- 4. Introduction to analogue communication systems.

About me

- I am a British man and native English Speaker.
- I have a first degree in Electronics (Bsc Electronics) and a masters' degree in Computer Science (Msc Computer Science)
- I currently live in London, UK but I will be relocating to Ouagadougu, Burkina Faso in 2023
- I currently teach English as a foreign language, as well as electronics, mathematics and physics
- I have taught another group similar to this one
- Any questions? Feel free to ask!

Contact Details

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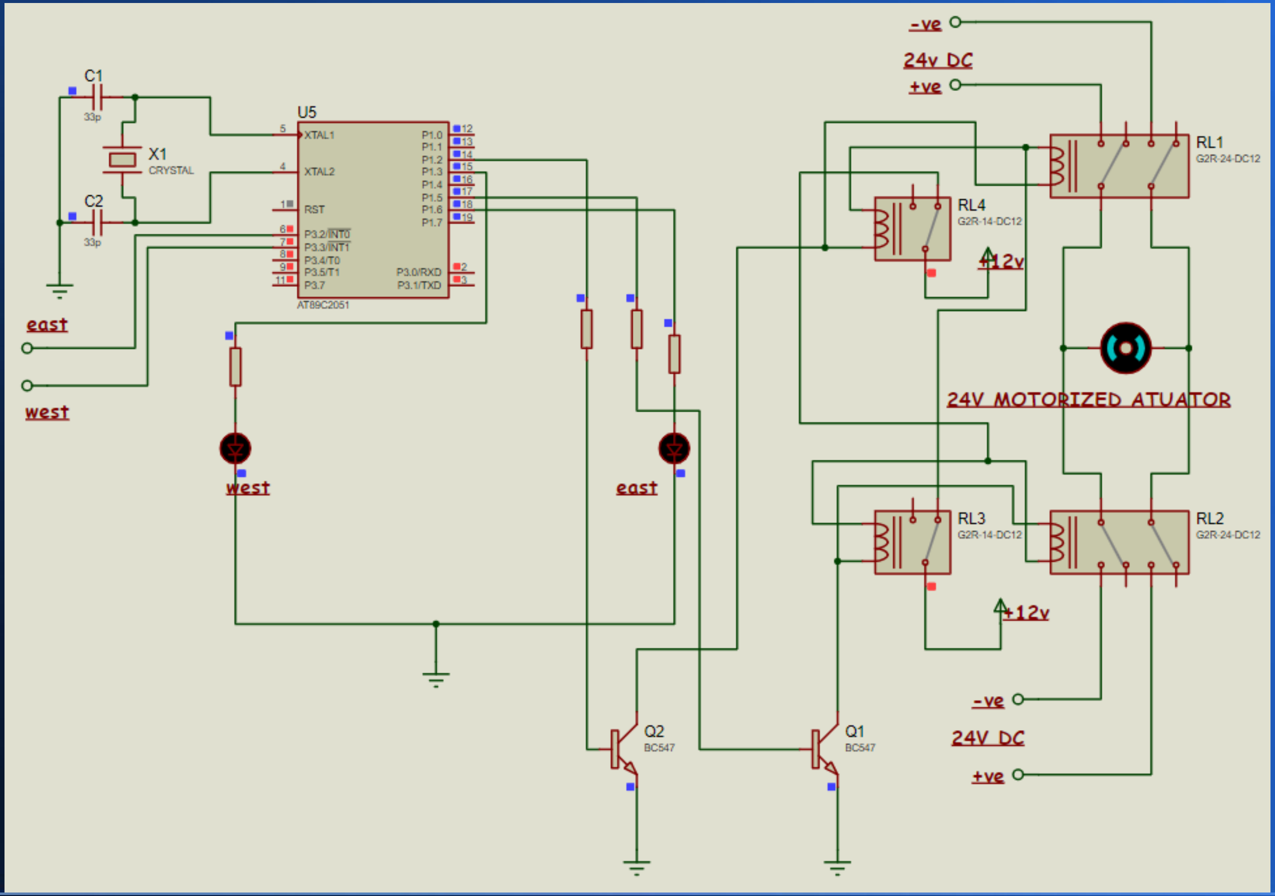
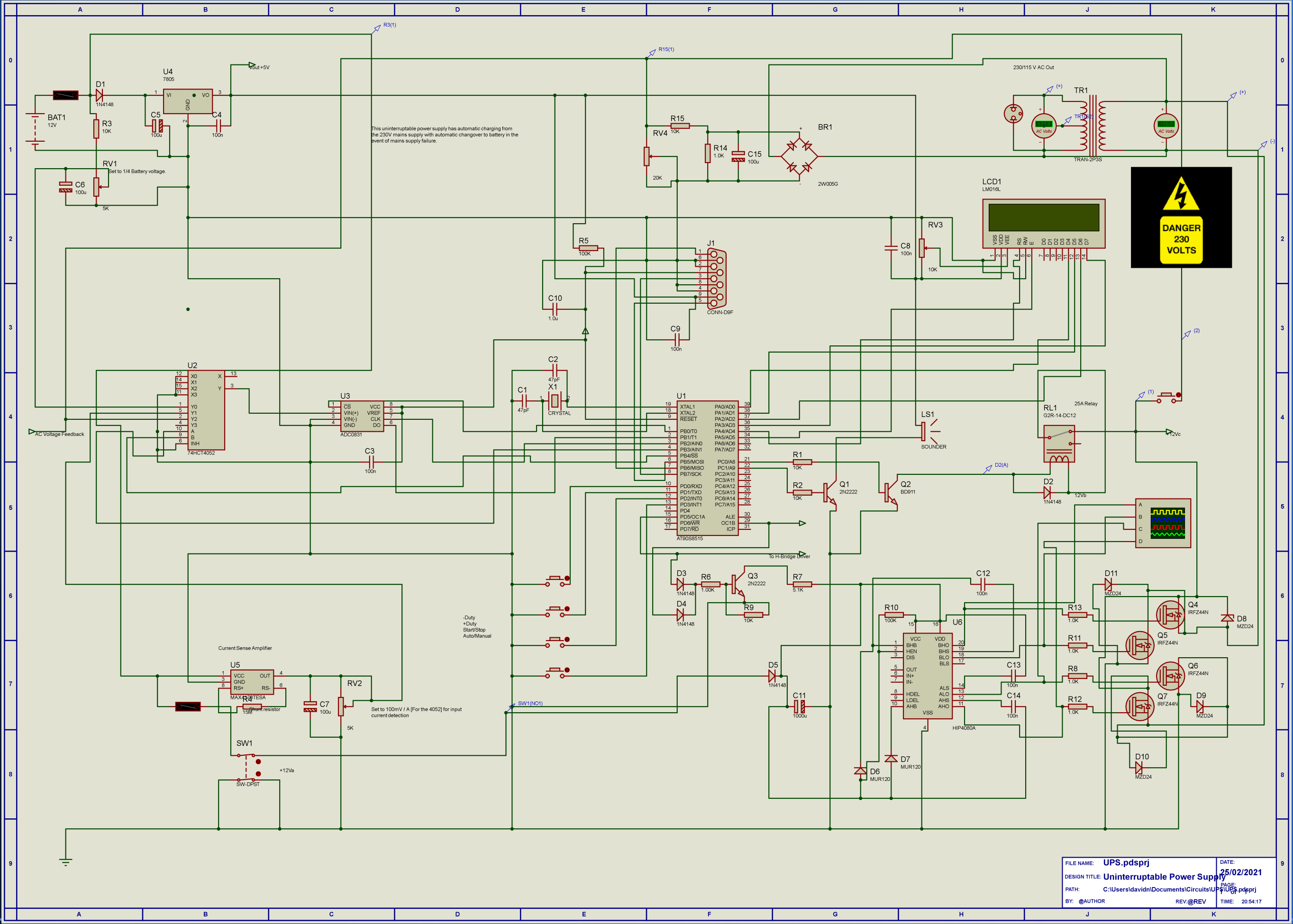
Analouge Electronics

- A simplified version of a device for keeping solar pannels aimed at the sun. I intend to set up a solar powered electronic business in Western Africa in the medium term. Used a mictocontroller and stepper motors. Firmware is written in 'C' and can use a time based tracking approach or a maximum power point tracking approach. The power yield of a solar panel is proportional to the cosine of the incident angle.



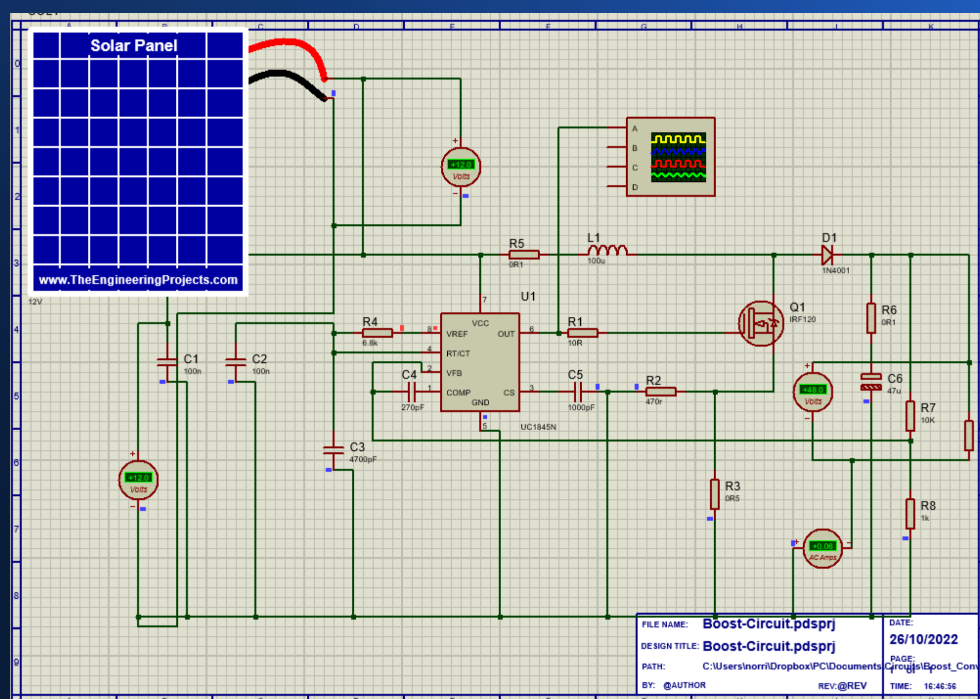
Analouge Electronics

- A low cost Uninteruptable power supply design for use in Western Africa, where I intend to establish a Business. Uses a microcontroller, firmware written in 'C'. Copyright © David Norris, 2021



Analouge Electronics

- A boost converter. Before semiconductors, the only way to change DC voltages was to use an inefficient motor/generator. Transformers operate only on AC supplies, and long distance distribution needs high voltage to reduce ohmic cable losses.



Teaching and learning activity

Use of software tools to analyse, design and build circuits and system blocks as well as practical testing.

• Conducting different laboratory experiments covering the main topics and components.

• Design and implementation of analogue electronic circuits and systems.

Assessment

Lab report - 50% LO - 1, 2, 3, 4. Pass mark - 40% 1500 words.

Exam - 50% LO - 1-4 2 hours. Exam will be based on topics covered during the lecture programme.

All elements of summative assessment must be passed to pass the module.

Nature of FORMATIVE assessment supporting student learning:

In-class assignments can be offered to help students grasp concepts introduced during the lectures programme. The feedback from these assignments will allow the teaching team to gauge students' understanding and focus support where required.The teaching team can also offer formative mock exams which will be carried out in-class and/or offered over a timed session via a Module Moodle page. These should mimic exam conditions and difficulties and will help prepare students for the summative exam.

Introduction to electrical circuits

- Units of measurement

Quantity	Quantity Symbol	Unit	Unit Symbol
Length	l	metre	m
Mass	m	kilogram	kg
Time	t	second	s
Velocity	v	metres per second	m/s or $m\,s^{-1}$
Acceleration	a	metres per second squared	m/s^2 or $m\,s^{-2}$
Force	F	newton	N
Electrical charge or quantity	Q	coulomb	C
Electric current	I	ampere	A

Quantity	Quantity Symbol	Unit	Unit Symbol
Resistance	R	ohm	Ω
Conductance	G	siemen	S
Electromotive force	E	volt	V
Potential difference	V	volt	V
Work	W	joule	J
Energy	E (or W)	joule	J
Power	P	watt	W

These are the units we need in our discussion of electrical machines. Please use these symbols to prevent confusion.

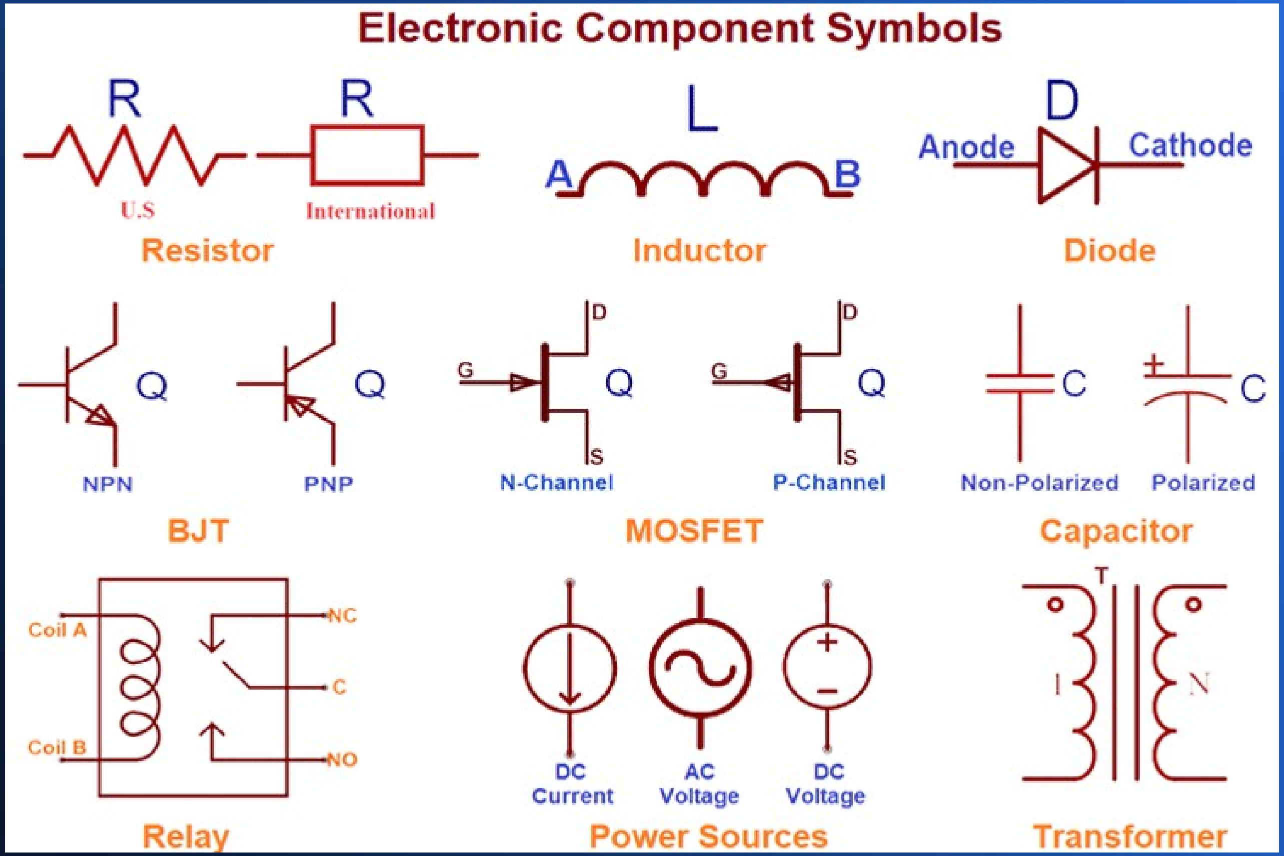
All of these are S.I units.

Electrical charatoristics

- Electrical charactoristics often have physical equivilents in the machanical world…

Presure	G/m²	Electrical presure	Voltage (V)
Current	Litres/ second	Electric current	Amps (I)
Friction	Newtons	Reistance	Ohms (Ω)
Capacity	Litres	Charge	Coulombs (C)
-	-	Power	Watts (W)
Thermal conductance	W/M².K	Conductanc e	Siemens (s)
-	-	Inductance	Hendries (L)

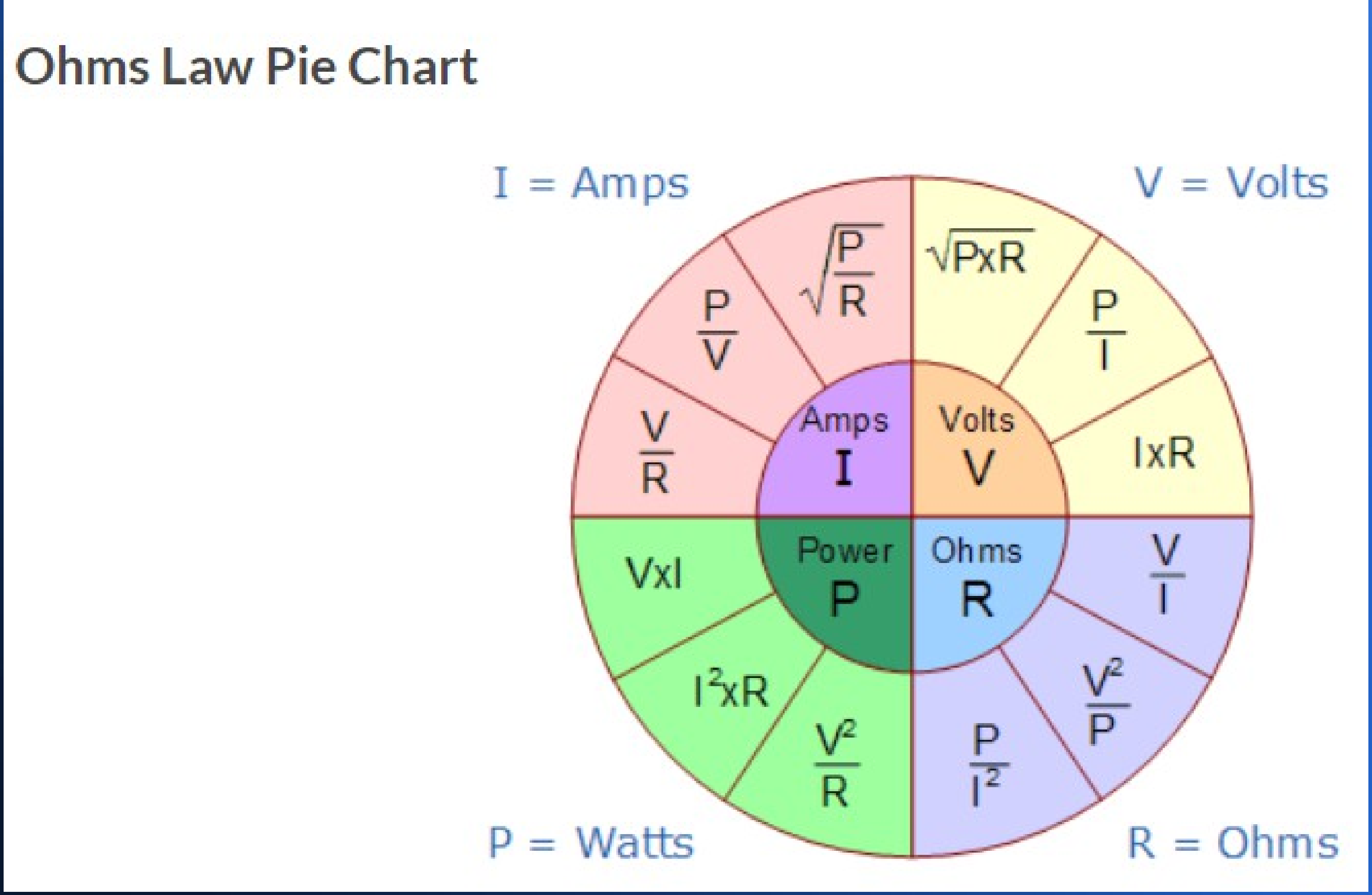
Standard Electronic Component Symbols



Ohms Law

- Ohm’s law
- Ohm’s law states that the current I flowing in a circuit:
- is directly proportional to the applied voltage V and
- inversely proportional to the resistance R, provided the
- temperature remains constant. Thus,
- $I = V/R$ or $V = IR$ or $R = V/I$
- This is analogous to current flow in a water pipe. Current flow is proportional to the pressure, and inversely proportional to the force of friction.
- Conductance is the reciprocal of resistance. It is measured in Seimens (S) as 1/R.

An easy way to remember Ohm's law...



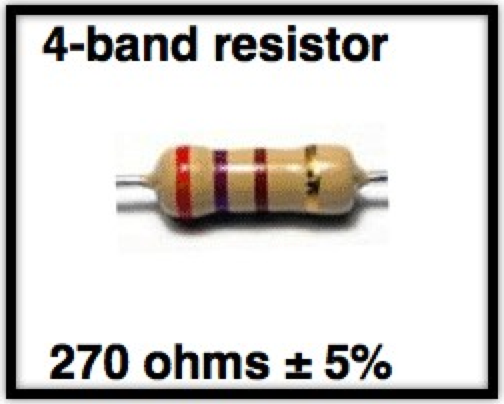
Example of a calculation

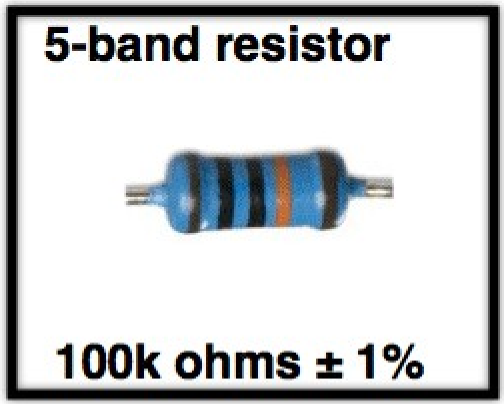
- A light emitting diode, l.e.d needs a current limiting resister to ensure that no more than 20mA flows through the l.e.d (or it will burn out). And the voltage is 9V from a PP3 battery. What is the lowest resistor value we can use? (Note that the l.e.d does not have an ohmic resistance, it introduces a voltage drop in a circuit. It is an example of a non-ohmic device. With no limiting resister the current flow would be: 9V /0Ω = ∞!. In practice, the maximum current the battery can deliver. This will burn the l.e.d out).
- Referring to ohms law, R =V/I = 9 X 0.002 = 4500Ω or 4.5kΩ. As 4.5kΩ is not a prefered value, in practice I would use a 4.7kΩ or 4k7 resistor.
- How can you tell the value of a resistor?

Resistor Colour code

- Our 4k7 resistor will have bands: yellow, purple, brown (the tolerence is not too critical here).

Color	Value	Multiplier	Tolerance
Black	0	$\times 10^0$	± 20%
Brown	1	$\times 10^1$	± 1%
Red	2	$\times 10^2$	± 2%
Orange	3	$\times 10^3$	± 3%
Yellow	4	$\times 10^4$	- 0, + 100%
Green	5	$\times 10^5$	± 0.5%
Blue	6	$\times 10^6$	± 0.25%
Violet	7	$\times 10^7$	± 0.10%
Gray	8	$\times 10^8$	± 0.05%
White	9	$\times 10^9$	± 10%
Gold	–	$\times 10^{-1}$	± 5%
Silver	–	$\times 10^{-2}$	± 10%



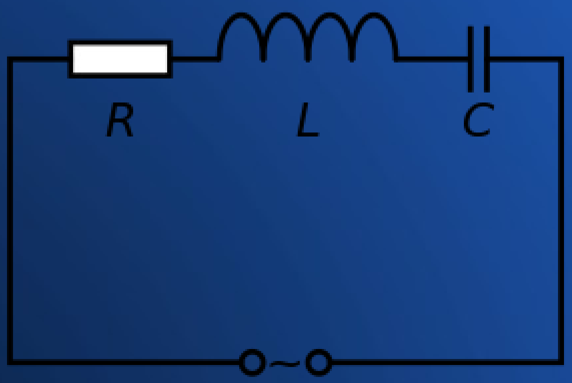


Ohms law excercise

- A kettle element is designed to draw 10A of current from the 230V mains supply. Find: the resistance of the element and the power rating of the kettle?
- $R(\Omega) = V/I = 230/10 = 23\Omega$
- Power (W) = VI = 230V x 10A = 2300W or 2.3KW.

Inductance and Capacitance as a second order system

- Inductance and capacitance together can form a second order system (as inductance and capacitance are both forms of energy storage) - in which a resonance can occour. This is an electrical equivilent of simple harmonic motion.
- The series resistor will provide 'electrical friction' and hence damping.



- The resonant frequency can be calculated using the formula: $f = 1 / (2 * \pi * \sqrt{(L * C)})$.

Capacitance

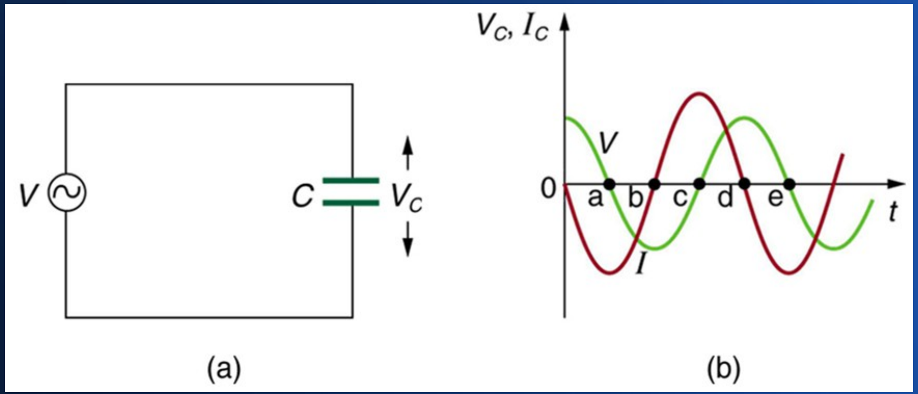
- Capacitance is the ratio of the amount of electric charge stored on a conductor to a difference in electric potential.
- Typically, two conductors are used to separate electric charge, with one conductor being positively charged and the other negatively charged, but the system having a total charge of zero. The ratio in this case is the magnitude of the electric charge on either conductor and the potential difference measured between the two conductors.
- The capacitance is a function only of the geometry of the design (the area of the plates and the distance between them) and the permittivity of the dielectric material between the plates of the capacitor. For many dielectric materials, the permittivity and thus the capacitance, is independent of the potential difference between the conductors and the total charge on them.
- The unit of capacitance is the Farad (F). A 1 farad capacitor, when charged with 1 coulomb of electrical charge, has a potential difference of 1 volt between its plates. The reciprocal of capacitance is called elastance. The formula for calculating capacitive reactance is: $X_c = 1 / (2\Pi FC)$ where F is in hertz, C in farads.
- The Farad is, in practice, a very large unit, so most values are in μF, nF and even pF.

Inductance

- Inductance is: the tendency of an electrical conductor to oppose a change in the electric current flowing through it. The flow of electric current creates a magnetic field around the conductor. The field strength depends on the magnitude of the current, and follows any changes in current. From Faraday's law of induction, any change in magnetic field through a circuit induces an electromotive force (EMF) (voltage) in the conductors, a process known as electromagnetic induction. This induced voltage created by the changing current has the effect of opposing the change in current. This is stated by Lenz's law, and the voltage is called back EMF.
- In the SI system, the unit of inductance is the henry (H), which is the amount of inductance that causes a voltage of one volt, when the current is changing at a rate of one ampere per second. The unit of inductance is the Hendry (L). Inductive reactance (which opposes current flow) is calcutated as $XL = 2\Pi FL$ where F is frequency in hertz, L is in hendries.
- In practice, the Hendry is a very large unit, and practical values are expressed in mH μH, nH and even pH.

Capacitive Reactance Example

- Capacitive Reactance Xc = 1/ (2ΠFC)
- For DC, F is = 0 and Xc = ∞Ω. (Any divisor of 0 leads to a ∞ result). So on DC a capacitor simply charges to the supply voltage with a time constant of T = RC.
- For 50Hz AC, a 1uF capacitor has a reactance of 1/ (2Π(50)(10^-6)) = 3.183kΩ. This is due to voltage and current being out of phase.



Inductive Reactance Example

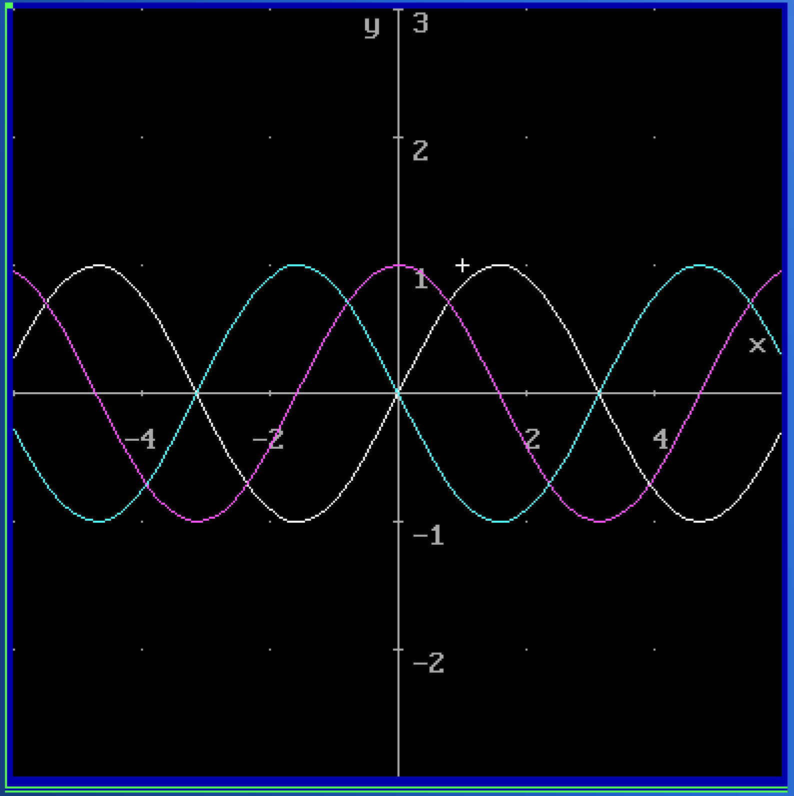
- Intuctive reactance, $X_L = 2\Pi FL$
- For DC, F is = 0 and $X_L = 0\Omega$. (Any multiplier of 0 leads to a 0 result). So for DC, an inductor has a reactance of 0Ω and only resistance determines the current – all wire has a little resistance!
- For 50Hz AC,a 1H inductor has an inductive reactance, $Xl = 2\Pi(50)(1) = 314.16\Omega$. Note that in a circuit, values in mH or uH are more common; the Hendry, like the Farad, is a very large unit indeed!
- For DC, be aware that a momentary transient voltage is induced at the time voltage is applied.

Why resistance, capacitive and inductive reactance are measured in Ohms, Ω

- In this way, once resistance, capacitive reactance and inductive reactance values are known, their effects of limiting current are equivilent and hence, a simple calculation can be carried out; simplifying calculations. Just be aware that resistance is not frequency dependent, but inductive reactance increases in proportion to frequency and capacitive reactance decreases in proportion to to frequency.
- These differences can lead to useful functions, such as in the design of filters. These can seperate different signals. And resonance is a very useful characteristic of combining inductance and capacitance values, but this is beyond the scope of this course.
- Be aware however, that these effects are direct equivilents of those which can be found in the mechanical world; a RLC circuit has equivilent charactoristics of a damped second order SHM system.

CIVIL – a mnemonic to help you remember...

- In a capacitor, current I leads voltage, V
- In an Inductor, voltage V leads current, I
- This is known as phase difference
- Difference is ±90°
- A RC combination has +0-90° phase shift
- A RL combination has -0-90° phase shift
- Reactance (Z) is frequency dependent...
- Resistance ® is not – a useful property
- This property is useful in a filter circuit.



Kirchhoff’s Laws | KCL & KVL

- Kirchoff's Voltage and Current laws:
- Many of the electrical circuits are complex in nature and the computations required to find the unknown quantities in such circuits, using simple ohm’s law and series/parallel combination simplifying methods is not possible. Therefore, in order to simplify these circuit calculations, Kirchhoff’s laws are used.
- Kirchhoff's Current Law (KCL)
- Kirchhoff's Current Law (KCL) states that the sum of all currents entering and leaving a node in any electrical network is always equal to zero. It is based on the principle of conservation of electric charge. The law is also referred to as Kirchhoff’s first law. In formula form this is given by:
- $I = 1\sum n \ Ii = 0$.

Kirchhoff’s Laws | KCL & KVL

- Kirchhoff's Voltage Law (KVL)
- The second law is also called Kirchhoff’s voltage law (KVL). It states that the sum of the voltage rises and voltage drops over all elements in a closed loop is equal to zero. In formula form:
- i=1∑n Vi=0

Kirchov's Voltage Law:

$$\sum_k V_k = 0 \quad \text{where } \mathbf{k} = 1,2,3,4,\ldots$$

$$V_1 + V_2 + V_3 + V_4 + \cdots = 0$$

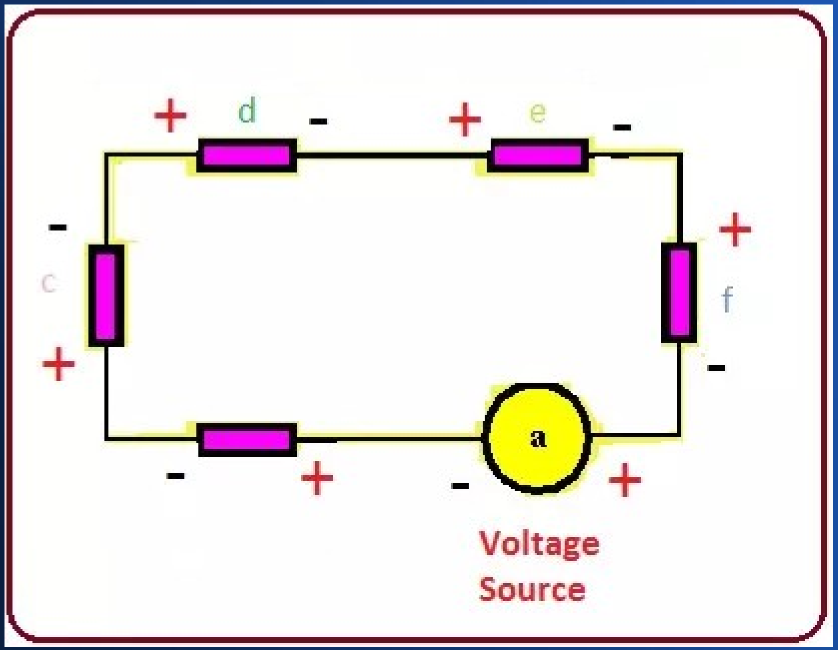
Kirchov's Current Law:

$$\sum_k i_k = 0 \quad \text{where } \mathbf{k} = 1,2,3,4,\ldots$$

$$i_1 + i_2 + i_3 + i_4 + \cdots = 0$$

What is KVL? (Kirchhoff's Voltage Law)

- Kirchhoff’s laws are not only applicable to DC circuitry, but also works for the AC circuits, when the electromagnetic radiation has large frequency values. In simple words, KVL says that the sum of voltages in an enclosed loop circuit is always equal to zero. By using this law we can easily find different parameters of a circuit like resistance, current or voltage quite easily.

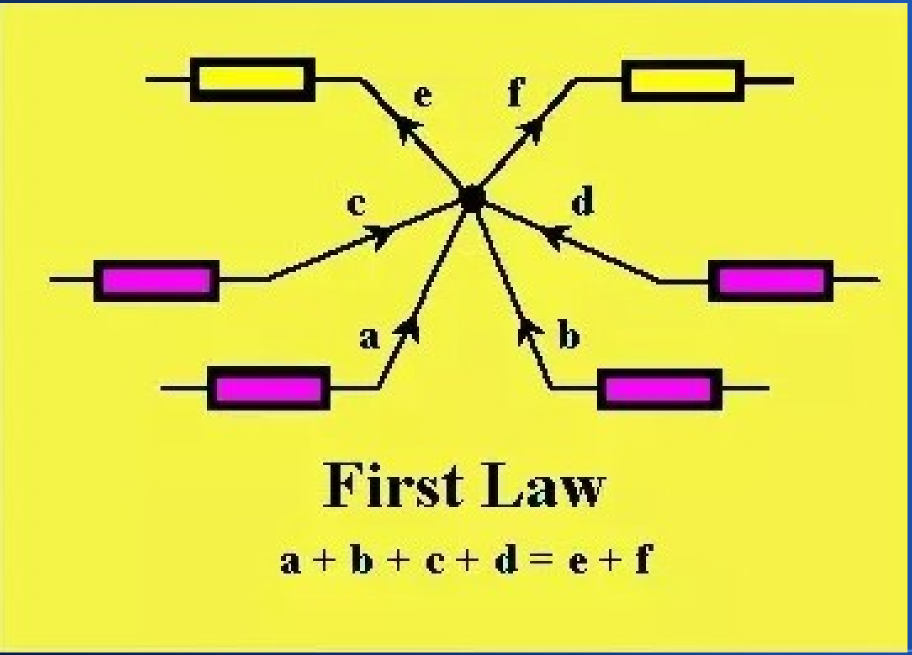


Application of KVL Law

- Mesh Analysis is the method to helps us to find the current and voltage in any close-loop working with the KVL, by this analysis we can find values of current and voltage across any component of the loop on the circuit.
- There are three steps to apply this mesh analysis. Which described here.
- Allocate discrete current values to every enclosed circle of the network.
- After that Apply Kirchoff voltage law about every enclosed circle of the system.
- And resolve the resultant concurrent linear equations to find the value of current in the ring.

Application of KCL Law

- These are some applications of this law.
- KCL is used to find the different electrical parameter like current, voltage and resistance in different circuits but it mostly used in complex circuits to find electrical parameters.

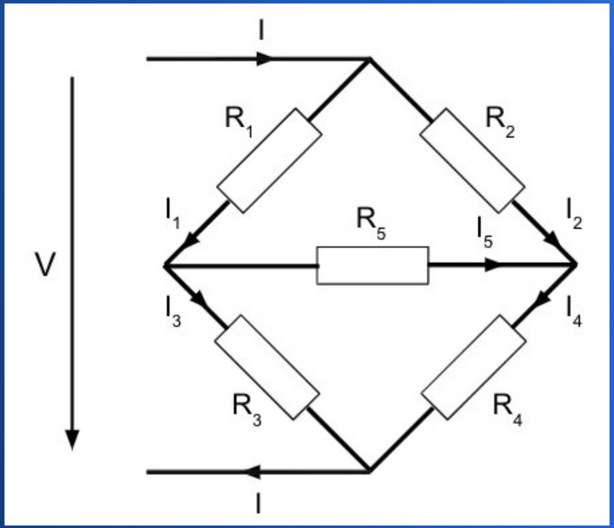


Nodal Analysis

- This method uses KCL to find the value of the voltage at the node and then calculate the values of current and voltage at any component of the circuit.
- There are some steps you should follow to apply this rule which described below.
- To apply this rule, first of all, you should find the no of nodes in a circuit and reference node.
- Then allocate current and its path to every discrete division (branch) of the nodes in the circuit.
- Apply KCL to every node of the circuit.
- Then make equations and resolve them to find the values of current (I) and voltage (V).
- Then find the values of current (I) and voltage (V) at every component of circuitry.

Kirchoff's Law Example: the Wheatstone Bridge

- Bridge circuits are a very common tool in electronics. They are used in measurements, transducers and switching circuits. I had an assignment involving one as an undergraduate. In this example, we will show how to use Kirchhoff's laws to determine the current I5. The circuit has four bridge sections with resistors R1 – R4. There is one cross bridge connection with resistor R5. The bridge is subject to a constant voltage V and current I.
- The first Kirchhoff law (KCL) states that the sum of all currents in one node is zero. So the total current entering must equal the total current leaving – electrons and energy cannot be made or destroyed.



Kirchoff's Law Example: the Wheatstone Bridge

- The first Kirchhoff law (KCL) states that the sum of all currents in one node is zero. In this example:
- I = I1 + I2, I = I3 + I4, I = I3 + I5
- The second Kirchhoff law (KVL) states the sum of all voltages across all elements in a loop is zero. For this example:
- R1 I1 + R3 I3 - V=0;
- R1 I1 + R5 I5 – V – R2 I2 = 0;
- R3 I3 – R4 I4 – R5 I5 = 0.
- The six sets of equations above can be rewritten to find the expression for I5 (the current in the cross branch):
- The equation shows that for the bridge

$$I_5 = \frac{V(R_2R_3 - R_1R_4)}{R_5(R_1+R_3)(R_2+R_4) + R_1R_3(R_2+R_4) + R_2R_4(R_1+R_3)}$$

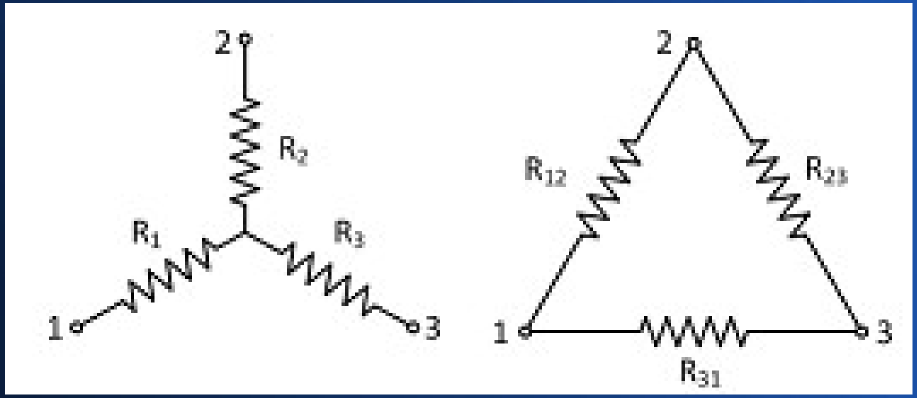
to be balanced with a bridge current

equal to zero:

R2 R3 = R1 R4.

Kirchoff's Law Example: the star-delta (or Y-Δ) conversion

- Kirchhoff's laws can be used to convert a star (also known as a Y) connection to a delta connection. For example, this connection is often seen in three phase AC systems, (for example the 400/230V mains supply in the European Union). A widely used application for star delta connections other than three phase transformers, is to limit the starting current of electric motors. The high starting current causes high voltage drops in the power system. As a solution, the motor windings are connected in the star configuration during starting and then change to the delta connection.



Kirchoff's Law Example: the star-delta (or Y-Δ) conversion

- The star connection as shown in the figure above, has the same voltage drops and currents as the delta connection shown on the right side, only when the following equations are valid:

$$R1 = \frac{R_{31}R_{12}}{R_{12} + R_{23} + R_{31}}$$

$$R_{12} = R_1 + R_2 + \frac{R_1 R_2}{R3}$$

$$R2 = \frac{R_{12}R_{23}}{R_{12} + R_{23} + R_{31}}$$

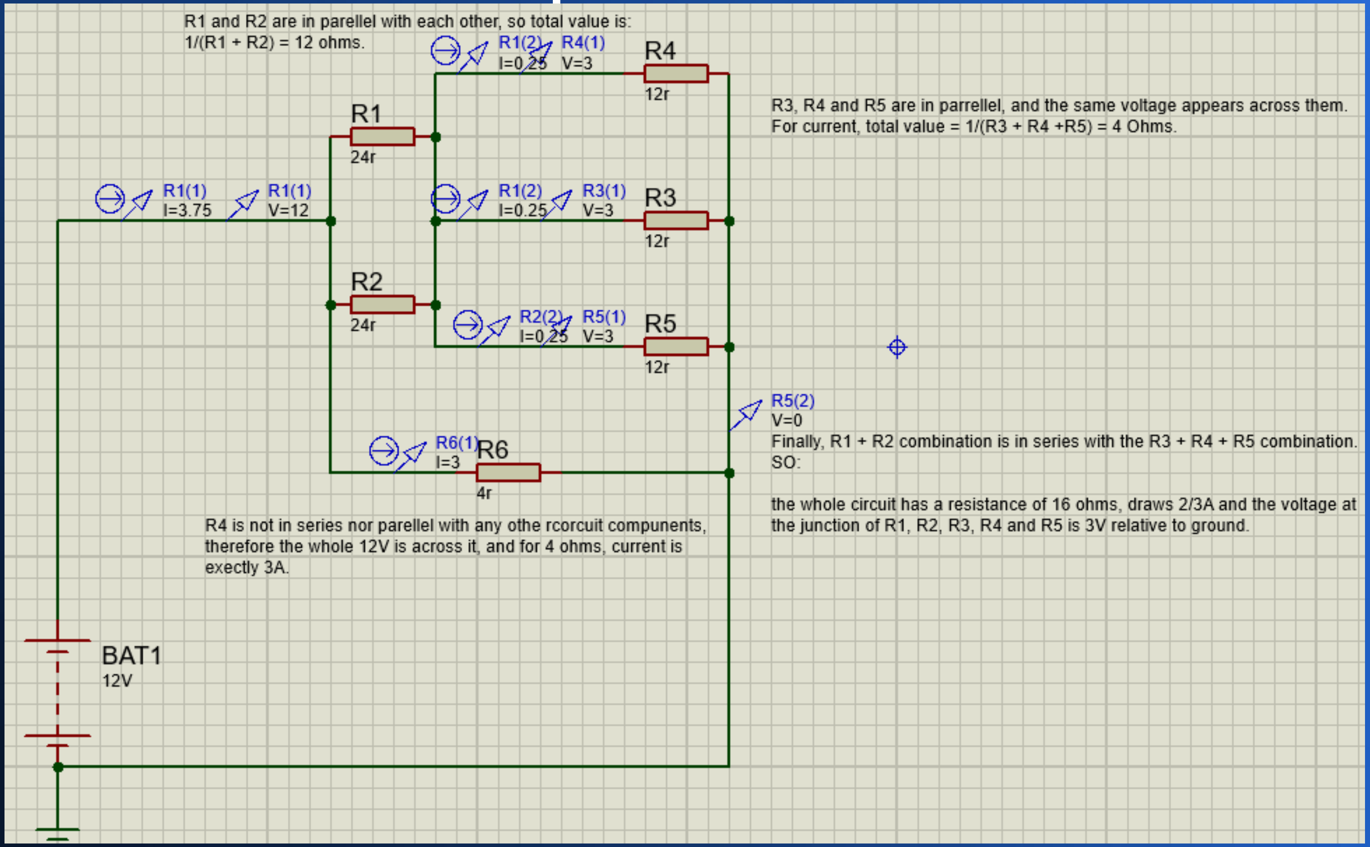
$$R_{23} = R_2 + R_3 + \frac{R_2 R_3}{R1}$$

$$R3 = \frac{R_{23}R_{31}}{R_{12} + R_{23} + R_{31}}$$

$$R_{31} = R_3 + R_1 + \frac{R_3 R_1}{R2}$$

Series and Parellel resistors

- An illustrated example in Proteus 8.6...



Series and Parellel Rules

- Resistors, series: R = R1 + R2 +R3....n
- Resistors, parellel: R = 1/(R1 + R2, + R3....n)
- Capacitors, series: C = 1/(C1 + C2 + C3....n)
- Capacitors, parellel: C = C1 + C2 + C3....n
- Inductors, series: L = L1 + L2 + L3....n
- Inductors, parellel: L = 1/(L1 + L2 + L3...n)

Impedance and Ohms' Law:

$Z = \sqrt{(X^2 + R^2)}$

- Impedance (Z) is a combined effect of resistance (not frequency dependant) and reactance (frequency dependent).
- Remember, inductive reactance increases with frequency, capacitive reactance decreases with frequency
- At DC, (0Hz) $X_L = 0\Omega$, $X_C = \infty\Omega$
- As current is out of phase by 90 degrees in a capacitor, and -90 degrees in an inductor, pythagoras' theorm can be used to calculate overall impedance.
- Formula for impedance: $Z = \sqrt{(X^2 + R^2)}$
- Example: if $X_C = 4\Omega$ and $R = 3\Omega$, Then: $Z = \sqrt{(4^2 + 3^2)} = 5\Omega$
- Sounds familiar? - it is Pythagoras Theorm. We met it in 'resultant forces' earlier!

Impedance Example

- In the example in the last slide, the required value of capacitance to obtain a capacitive reactance of 4Ω at 50Hz (the mains frequency in China – and all of Asia, Europe, Africa and Australasia) would require a capacitance of 795.8μF.
- North/South American continents use 60Hz please note.
- Be aware that is the reason why inductors and capacitors do not have their reactance printed on them – it depends on the AC frequency!!!
- In principle, equivilent circuits such as filters could use either an inductor or a capacitor. However, as inductors tend to be larger and heavier than a capacitor equivilent in reactance, it is rare to encounter RL circuits in practice, and analouge IC's can contain resistors, capacitors, diodes and transistors, but never include inductors; inductance values at chip scales are so small as to be useless.

Impedance Example

- In our example (capacitive reactance of 4Ω at 50Hz; capacitance of $795.8\mu F$), rearranging for C gives: $795.8\ \mu F = 1/2\Pi(50Hz)(4\Omega)$, remember $C = 1/(2\Pi FXc)$.

Remember, for our impedance calculation $Z= \sqrt{(X^2 + R^2)}$ can be rearranged as:

$X = -\sqrt{(Z^2 – R^2)}$. (remember Z is always ≥ 0; and of course square roots of any negative value can never exist);

$R = \sqrt{(Z^2 - X^2)}$.

Bipolar Transistors

- The transistor is our most important exam ple o f an “active” component, a device that can amplify, producing an output signal with more power in it than the input signal. The additional power comes from an external source of power (the pow er supply, of course).
- Note that voltage amplification isn’t what matters, -a step-up transformer, a “passive” component just like a resistor or capacitor, has voltage gain but no power gain.
- The property of power amplification seem ed very im portant to the inventors o f the transistor. Almost the first thing they did to convince themselves that they had really invented something was to power a loudspeaker from a transistor, observing that the output signal sounded louder than the input signal.
- The extra power comes from an external source.
- Power gain does NOT mean energy gain!

Bipolar Transistors

- The transistor is the essential ingredient of every electronic circuit, from the simplest amplifier or oscillator to the most elaborate digital computer. Integrated circuits (ICs), w hich have largely replaced circuits constructed from discrete transistors, are them selves merely arrays of transistors and other components built from a single chip o f semiconductor material.
- A good understanding of transistors is very important, even if most of your circuits are made from ICs, because you need to understand the input and output properties of the IC in order to connect it to the rest of your circuit and to the outside world.

Bipolar Transistors

- There are two major types of transistors: bipolar junction transistors
- (BJTs), historically came first with their Nobel Prize-winning invention in 1947 at Bell Laboratories. The next chapter deals with “field-effect” transistors (FETs), the now-dominant type in digital electronics. To give the coarsest com parison, BJTs excel in accuracy and low noise, whereas FETs excel in low power, high impedance, and high-current switching.
- For those learning electronics for the first time, this introduction will be slightly difficult. All the following are two-terminal devices, whether linear (resistors, capacitors, inductors) or nonlinear (diodes). So there w as only one voltage (the voltage betw een the term inals) and only one current (the current flowing through the device) to think about. Transistors, by contrast, are threeterm inal devices, which means there are two voltages and two currents to get used to.

Bipolar Transistors

- A bipolar transistor is a three-terminal device in which a small current applied to the base controls a m uch larger current flowing between the collector and emitter. It is available in two types (npn and pnp), with properties that meet the following rules for npn transistors (for pnp simply reverse all polarities)...
- These are some examples of transistors.
- B = base, C = collector; E = Emitter.
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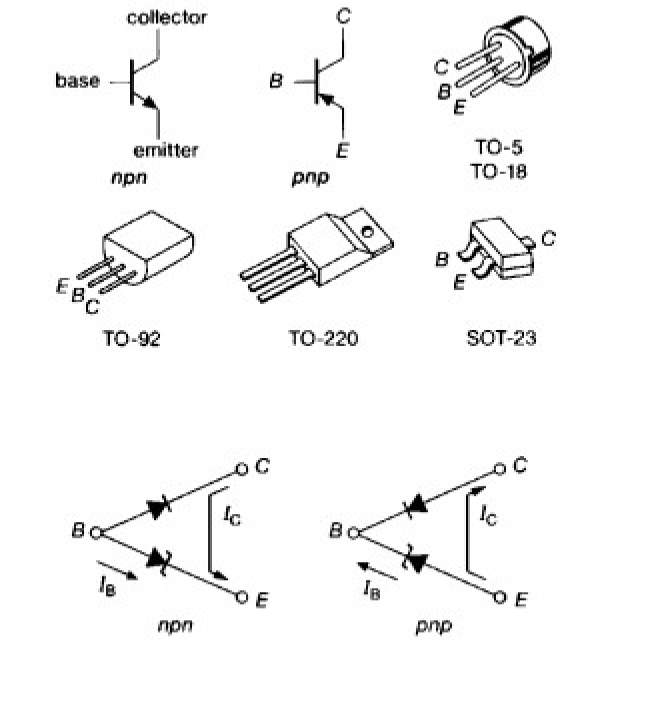


Bipolar Transistors

- 1. Polarity: The collector must be more positive than the
- emitter.
- 2. Junctions: The base-emitter and base-colector circuits behave like diodes in which a small current applied to the base controls a m uch larger current flowing betw een the collector and emitter. Normally the base-em itter diode is conducting, w hereas the base - collector diode is reverse-biased, i.e., the applied voltage is in the opposite direction to easy current flow.
- 3. Maximum ratings: Any given transistor has maximum values o f Ic, I b> and Vce that cannot be exceeded without costing the price of a new transistor! There are also other limits, such as power dissipation (/cF ce). temperature, and Vbe, that you must keep in mind.
- 4. Current amplifier: When rules 1-3 are obeyed, Iq is roughly proportional to / b and can be written as:

Bipolar Transistors

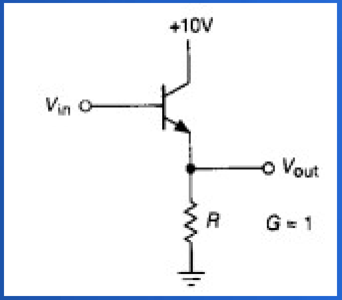
Ic=hFE/B=ßIB, where the current gain (sometimes called ß), is typically about 100. Both / b and / e flow to the emitter.

- Note: the collector current is not due to forward conduction of the base-collector diode; that diode is reverse biased. Just think of it as “transistor action.”
- The simple curcuit opposite is an emitter follower. It seems useless. Until you realise it has a high input impedance, a low output

Impedance – and able to power a heavier load than Vin

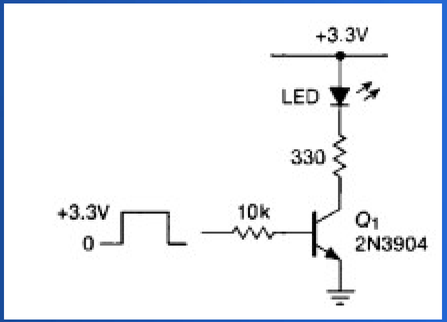
can drive directly. It has no voltage gain – but it has current

and power gain! So it has a use after all!



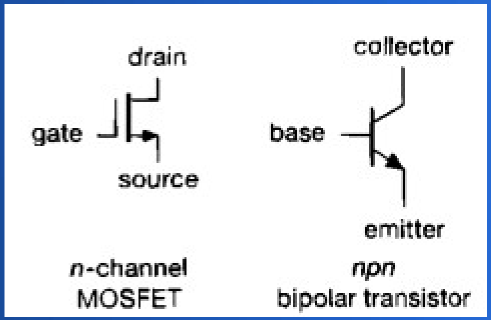
Bipolar Transistors

- Transistor Switch:
- This application, in which a small control current enables a much larger current to flow in another circuit, is called a transistor switch. From the preceding rules it is easy to understand.
- For example, a clock pulse from a CMOS (complementry metal oxide silicon) device will not light a LED (light emitting diode) directly, but this transistor switch outputs enough current to do so easily.



FIELD-EFFECT TRANSISTORS

- Field-effect transistors (FETs) are different from bipolar transistors that we already met.
- Broadly speaking, however, they are similar devices, which we might call charge-control devices: in both cases we have a three-terminal device in which the conduction between two electrodes depends on the availability of charge carriers, which is controlled by a voltage applied to a third control electrode.



FIELD-EFFECT TRANSISTORS

- Here's how they differ: in a bipolar transistor the collector-base junction is back-biased, so no current normally flows. Forward-biasing the base-emitter junction by «0.6V overcomes its diode "contact potential barrier,"causing electrons to enter the base region, where they are strongly attracted to the collector. Although some base current results, most of these "minority carriers" are captured by the collector. This results in a collector current, controlled by a (much smaller) base current. The collector current is proportional to the rate of injection of minority carriers into the base region, which is an exponential function of Vbe (the Ebers-Moll equation). You can think of a bipolar transistor as a current amplifier (with roughly constant current gain j3) or as a transconductance device (Ebers-Moll: collector current programmed by base-emitter voltage).

FIELD-EFFECT TRANSISTORS

- In an FET, as the name suggests, conduction in a channel is controlled by an electric field, produced by a voltage applied to the gate electrode. There are no forwardbiased junctions, so the gate draws no current. This is perhaps the most important advantage of the FET. As with BJTs, there are two polarities, n-channel FETs (conduction by electrons) and p-channel FETs (conduction by holes).
- These two polarities are analogous to the familiar npn and pnp bipolar transistors, respectively. In addition, however, FETs tend to be confusing at first because they can be made with two different kinds of gates (thus JFETs and MOSFETs) and with two different kinds of channel doping (leading to enhancement and depletion modes). We'll talk about these possibilities shortly.

FIELD-EFFECT TRANSISTORS

- The FET’s nonexistent gate current is its best characteristic! The resulting high input impedance (which can be < 10^l4Ω) is essential in many applications,and in any case it simplifies circuit design! For applications like analouge switches and amplifiers of ultrahigh input impedance, FETs have no equal. They can be easily used by themselves or combined with bipolar transistors to make integrated circuit.
- N-channel & p-channel:
- FETs (like BJTs) can be fabricated in both polarities. Thus the mirror twin of our n-channel MOSFET is a p-channel MOSFET. Its behavior is symmetrical, mimicking pnp transistors: the drain is normally negative with respect to the source, and drain current flows if the gate is brought at least a volt or two negative with respect to the source. The symmetry isn’t perfect because the carriers are holes, rather than electrons, with lower mobility and minority carrier lifetime. The consequence is worth remembering - p-channel FETs usually have poorer performance, manifested as a higher gate threshold voltage, higher /? o n , and lower saturation current.

FIELD-EFFECT TRANSISTORS

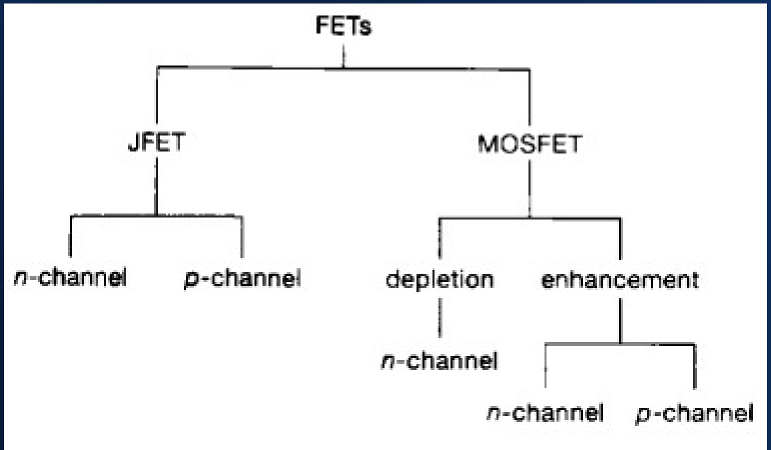
- M O S F E T & J F E T..

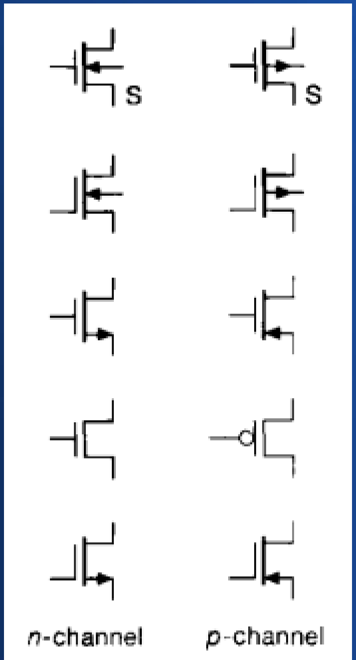
In a MOSFET (“Metal-Oxide-Semiconductor Field-Effect Transistor”) the gate region is separated from the conducting channel by a thin layer of Si02 (glass) grown onto the channel. The gate, which may be either metal or doped silicon, is truly insulated from the sourcedrain circuit, with characteristic input resistance >10^14Ω.

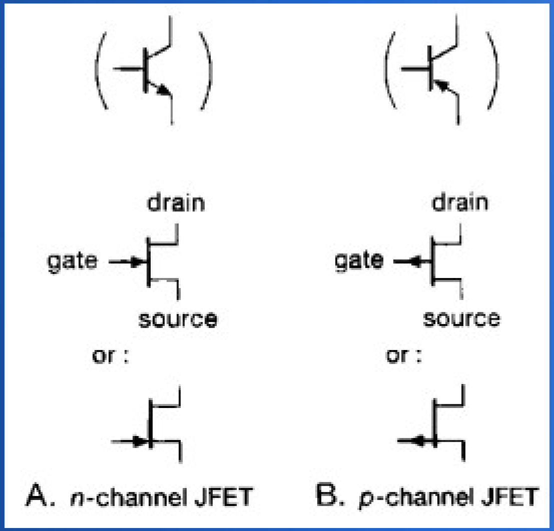
In a JFET (Junction Field-Effect Transistor) the gate forms a semiconductor junction with the underlying channel. This has the important consequence that a JFET gate should not he forward biased with respect to the channel, to prevent gate current. For example, diode conduction will occur as the gate of an n channel JFET approaches +0.6 V with respect to the more negative end of the channel (which is usually the source). The gate is therefore operated reverse-biased with respect to the channel, current (except diode leakage) flows in the gate circuit.

FIELD-EFFECT TRANSISTORS

Once again, we favor the symbol with offset gate to identify the source (though JFETs and small integrated MOSFETs are symmetrical, power MOSFETs are quite asymmetrical, with very different capacitances and breakdown voltages).

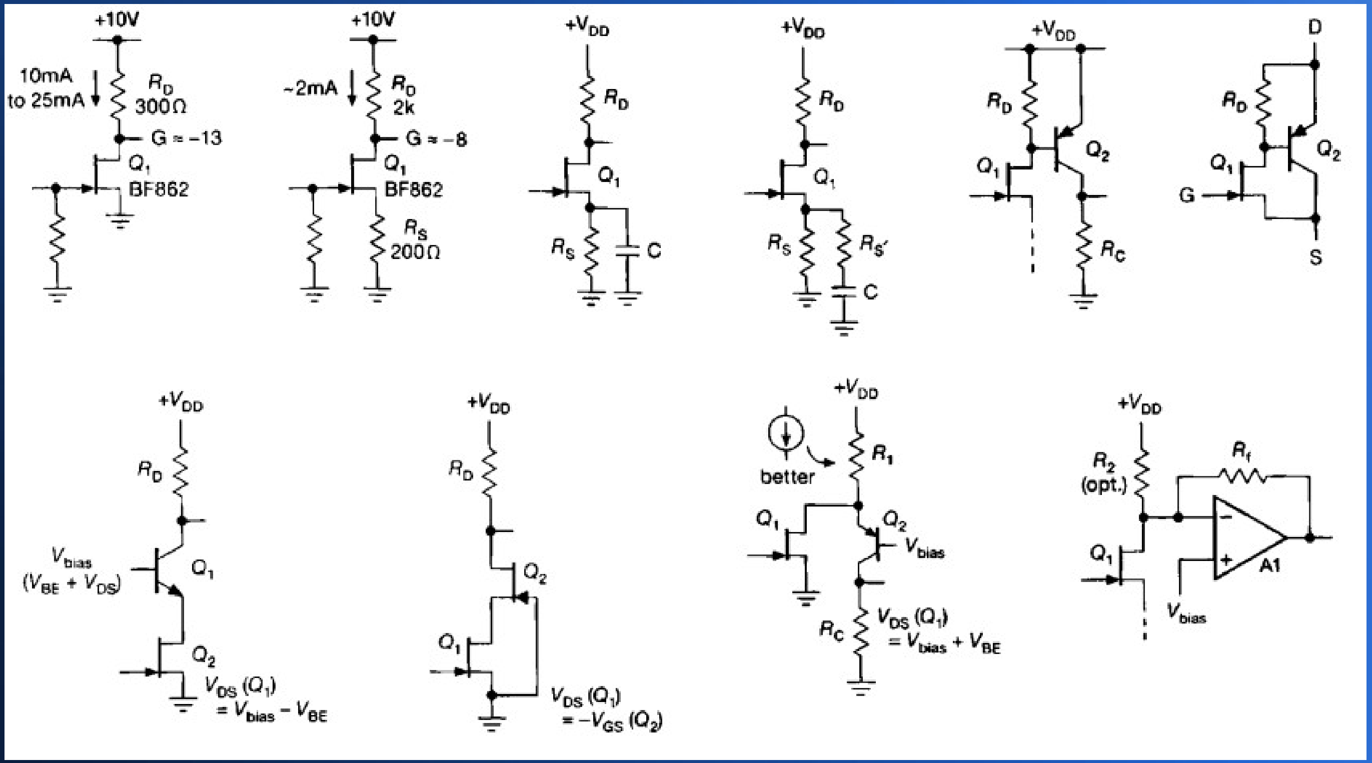






FIELD-EFFECT TRANSISTORS

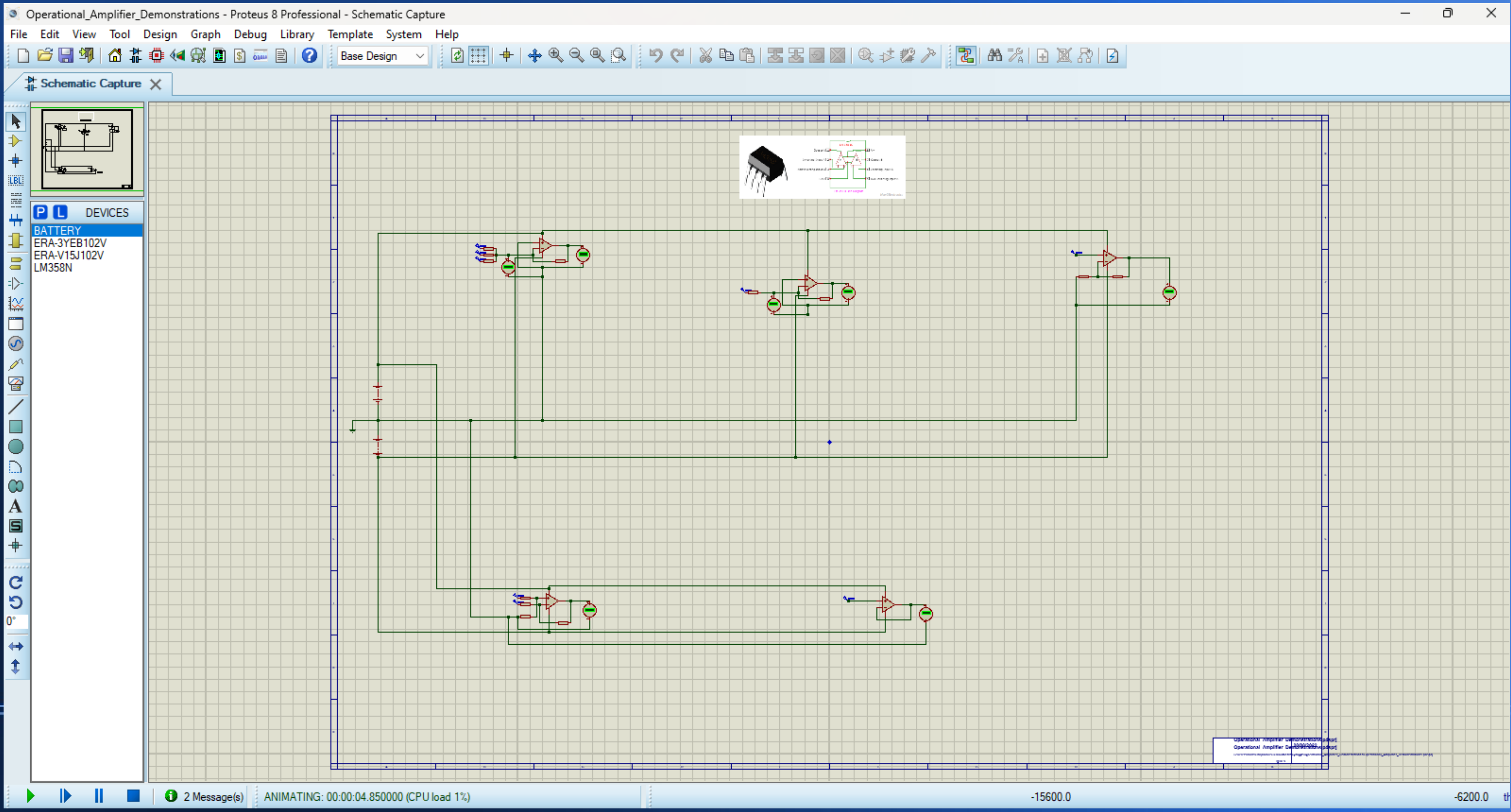
- Some single source JFET amplifier configurations.



FIELD-EFFECT TRANSISTORS

- M O S F E T's work well as saturated switches.
- Power MOSFETs are now available from many manufacturers, making the advantages o f MOSFET's (high input impedance, easy paralleling , absence of “ second breakdown”) applicable to power circuits . Generally speaking, power MOSFET's are easier to use than conventional bipolar power transistors.
- ***Thermal runaway!!!***
- Up to now, we’ve avoided the topic - but “thermal runaway” is quite a nightmare; it refers particularly to circuit configurations in which the power dissipation produces a rise in temperature that in turn raises the power that must be dissipated. Two important examples are the push-pull linear amplifier and the saturated power switch.
- I have seen projects in a laboritory literally go up in smoke due to thermal runaway. No, it was not a laughing matter.

Introducing Amplifiers



Introducing Amplifiers

- I I first learned about amplifier classes before I was even an undergraduate, in order to take my radio amateurs' exam in May 1995. (I passed both sections on the first attempt and gained my UK licence and callsign {G7VDI} in July 1995.
- Amplifiers are almost synonymous with audio applications – but they have other uses too! For example, power transistors are used in my inverter and UPS designs too.

Amplifier Classes

- Not all amplifier designs are the same. There is a clear distinction made between amplifier classes as too the way their output stages are configured and operate. The main operating characteristics of an ideal amplifier are linearity, signal gain, efficiency and power output but in real world amplifiers there is always a trade off between these different characteristics.
- Generally, large signal or power amplifiers are used in the output stages of audio amplifier systems to drive a loudspeaker load. A typical loudspeaker has an impedance of between 4Ω and 8Ω, thus a power amplifier must be able to supply the high peak currents required to drive the low impedance speaker.
- One method used to distinguish the electrical characteristics of different types of amplifiers is by “class”, and as such amplifiers are classified according to their circuit configuration and method of operation. Then Amplifier Classes is the term used to differentiate between the different amplifier types.

Amplifier Classes

- Amplifier Classes represent the amount of the output signal which varies within the amplifier circuit over one cycle of operation when excited by a sinusoidal input signal. The classification of amplifiers range from entirely linear operation (for use in high-fidelity signal amplification) with very low efficiency, to entirely non-linear (where a faithful signal reproduction is not so important) operation but with a much higher efficiency, while others are a compromise between the two.
- Amplifier classes are mainly lumped into two basic groups. The first are the classically controlled conduction angle amplifiers forming the more common amplifier classes of A, B, AB and C, which are defined by the length of their conduction state over some portion of the output waveform, such that the output stage transistor operation lies somewhere between being “fully-ON” and “fully-OFF”.

Amplifier Classes

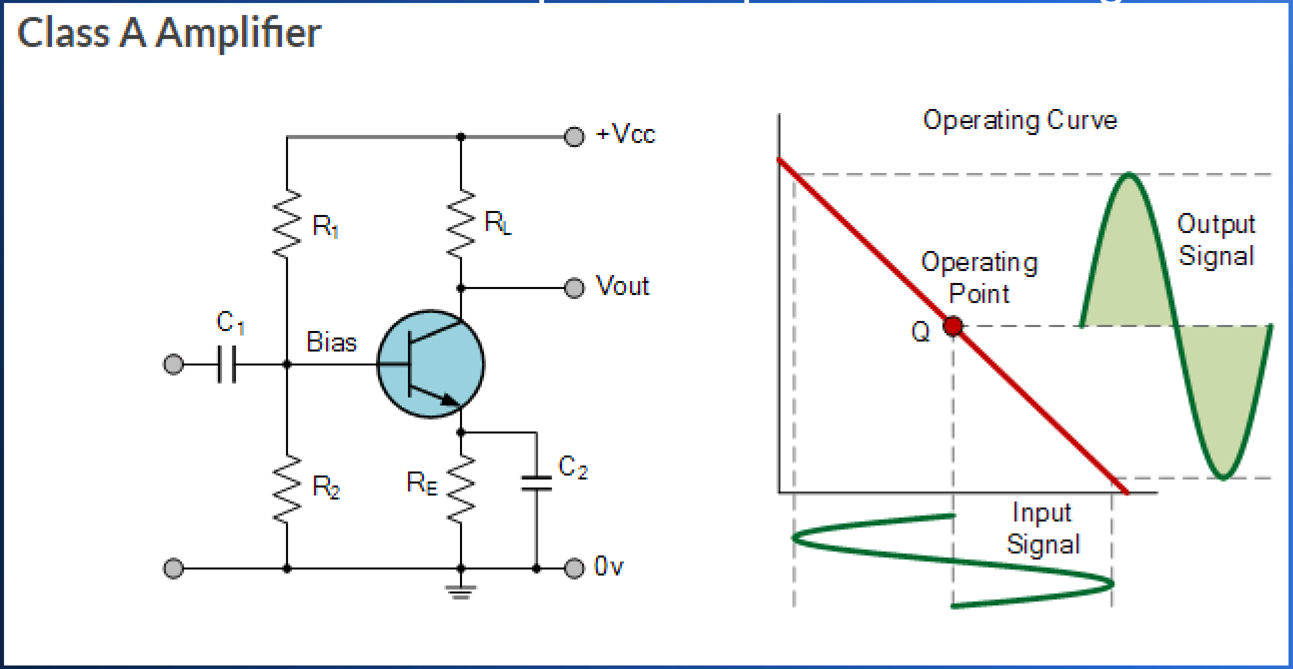
- The second set of amplifiers are the newer so-called “switching” amplifier classes of D, E, F, G, S, T etc, which use digital circuits and pulse width modulation (PWM) to constantly switch the signal between “fully-ON” and “fully-OFF” driving the output hard into the transistors saturation and cut-off regions.
- The most commonly constructed amplifier classes are those that are used as audio amplifiers, mainly class A, B, AB and C and to keep things simple, it is these types of amplifier classes we will look at here in more detail.

Class A Amplifiers

- Class A Amplifier Classes
- Class A Amplifiers are the most common type of amplifier topology as they use just one output switching transistor (Bipolar, FET, IGBT, etc) within their amplifier design. This single output transistor is biased around the Q-point within the middle of its load line and so is never driven into its cut-off or saturation regions thus allowing it to conduct current over the full 360 degrees of the input cycle. Then the output transistor of a class-A topology never turns “OFF” which is its main disadvantage.
- Class “A” amplifiers are considered the best class of amplifier design due mainly to their excellent linearity, high gain and low signal distortion levels when designed correctly. Although seldom used in high power amplifier applications due to thermal power supply considerations, class-A amplifiers are probably the best sounding of all the amplifier classes mentioned here and as such are used in high-fidelity audio amplifier designs.

Class A Amplifiers

- To achieve high linearity and gain, the output stage of a class A amplifier is biased “ON” (conducting) all the time. Then for an amplifier to be classified as “Class A” the zero signal idle current in the output stage must be equal to or greater than the maximum load current required to produce the largest output signal.



Class A Amplifiers

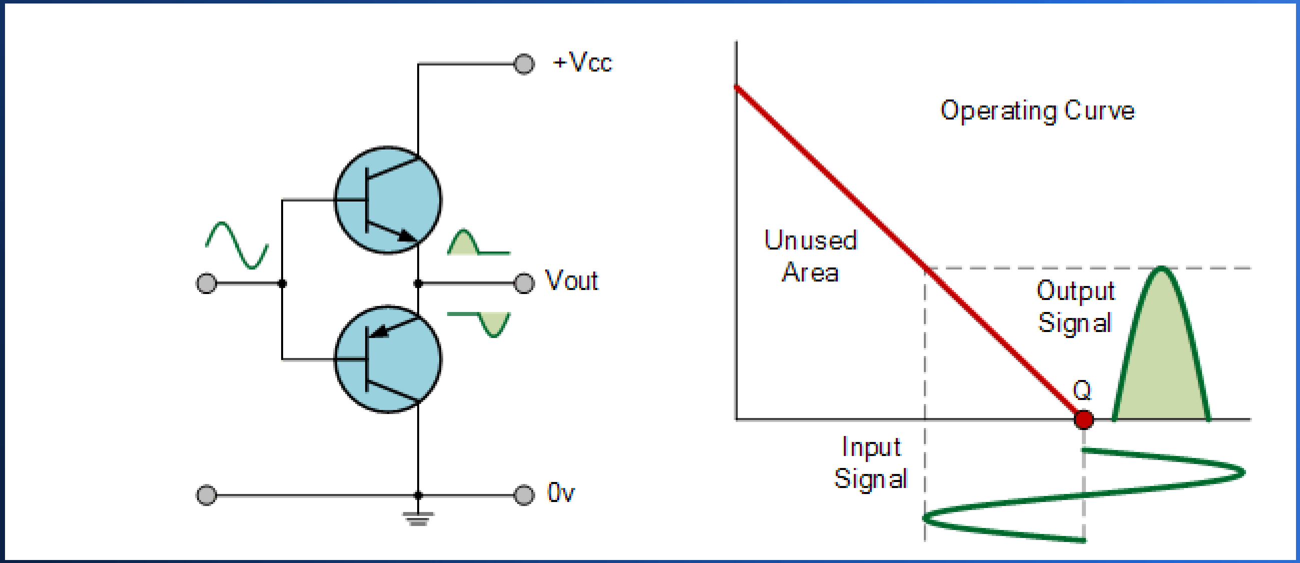
- As a class A amplifier operates in the linear portion of its characteristic curves, the single output device conducts through a full 360 degrees of the output waveform. Then the class A amplifier is equivalent to a current source.
- Since a class A amplifier operates in the linear region, the transistors base (or gate) DC biasing voltage should by chosen properly to ensure correct operation and low distortion. However, as the output device is “ON” at all times, it is constantly carrying current, which represents a continuous loss of power in the amplifier.
- Due to this continuous loss of power class A amplifiers create tremendous amounts of heat adding to their very low efficiency at around 30%, making them impractical for high-power amplifications. Also due to the high idling current of the amplifier, the power supply must be sized accordingly and be well filtered to avoid any amplifier hum and noise. Therefore, due to the low efficiency and over heating problems of Class A amplifiers, more efficient amplifier classes have been developed.

Class B amplifiers

- Class B amplifiers were invented as a solution to the efficiency and heating problems associated with the previous class A amplifier. The basic class B amplifier uses two complimentary transistors either bipolar of FET for each half of the waveform with its output stage configured in a “push-pull” type arrangement, so that each transistor device amplifies only half of the output waveform.
- In the class B amplifier, there is no DC base bias current as its quiescent current is zero, so that the dc power is small and therefore its efficiency is much higher than that of the class A amplifier. However, the price paid for the improvement in the efficiency is in the linearity of the switching device.

Class B amplifiers

- When the input signal goes positive, the positive biased transistor conducts while the negative transistor is switched "OFF". Likewise, when the input signal goes negative, the positive transistor switches "OFF" while the negative biased transistor turns "ON" and conducts the negative portion of the signal.



Class B amplifiers

- Thus the transistor conducts only half of the time, either on positive or negative half cycle of the input signal.Then we can see that each transistor device of the class B amplifier only conducts through one half or 180 degrees of the output waveform in strict time alternation, but as the output stage has devices for both halves of the signal waveform the two halves are combined together to produce the full linear output waveform.
- This push-pull design of amplifier is obviously more efficient than Class A, at about 50%, but the problem with the class B amplifier design is that it can create distortion at the zero-crossing point of the waveform due to the transistors dead band of input base voltages from -0.7V to +0.7.

Class B amplifiers

- We remember from the Transistor tutorial that it takes a base-emitter voltage of about 0.7 volts to get a bipolar transistor to start conducting. Then in a class B amplifier, the output transistor is not “biased” to an “ON” state of operation until this voltage is exceeded.
- This means that the the part of the waveform which falls within this 0.7 volt window will not be reproduced accurately making the class B amplifier unsuitable for precision audio amplifier applications.
- To overcome this zero-crossing distortion (also known as Crossover Distortion) class AB amplifiers were developed.

Class AB Amplifier

- As its name suggests, the Class AB Amplifier is a combination of the “Class A” and the “Class B” type amplifiers we have looked at above. The AB classification of amplifier is currently one of the most common used types of audio power amplifier design.
- The class AB amplifier is a variation of a class B amplifier as described above, except that both devices are allowed to conduct at the same time around the waveforms crossover point eliminating the crossover distortion problems of the previous class B amplifier.
- The two transistors have a very small bias voltage, typically at 5 to 10% of the quiescent current to bias the transistors just above its cut-off point. Then the conducting device, either bipolar of FET, will be “ON” for more than one half cycle, but much less than one full cycle of the input signal. Therefore, in a class AB amplifier design each of the push-pull transistors is conducting for slightly more than the half cycle of conduction in class B, but much less than the full cycle of conduction of class A.

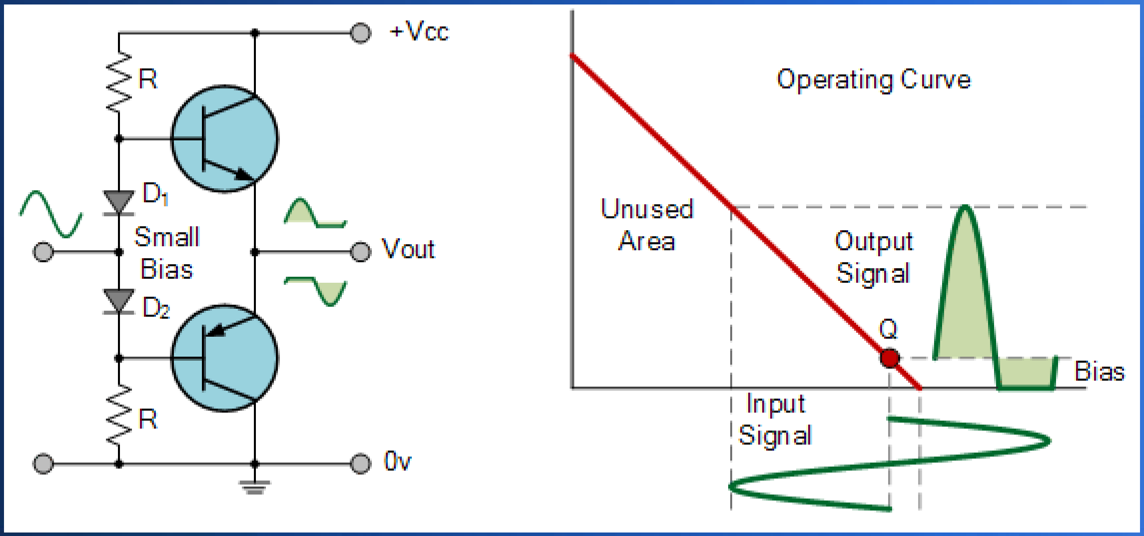
Class AB Amplifier

- In other words, the conduction angle of a class AB amplifier is somewhere between 180o and 360o depending upon the chosen bias point as shown.
- The advantage of this small bias voltage, provided by series diodes or resistors, is that the crossover distortion created by the class B amplifier characteristics is overcome, without the inefficiencies of the class A amplifier design. So the class AB amplifier is a good compromise between class A and class B in terms of

efficiency and linearity, with conversion

efficiencies reaching about 50% to

60%.

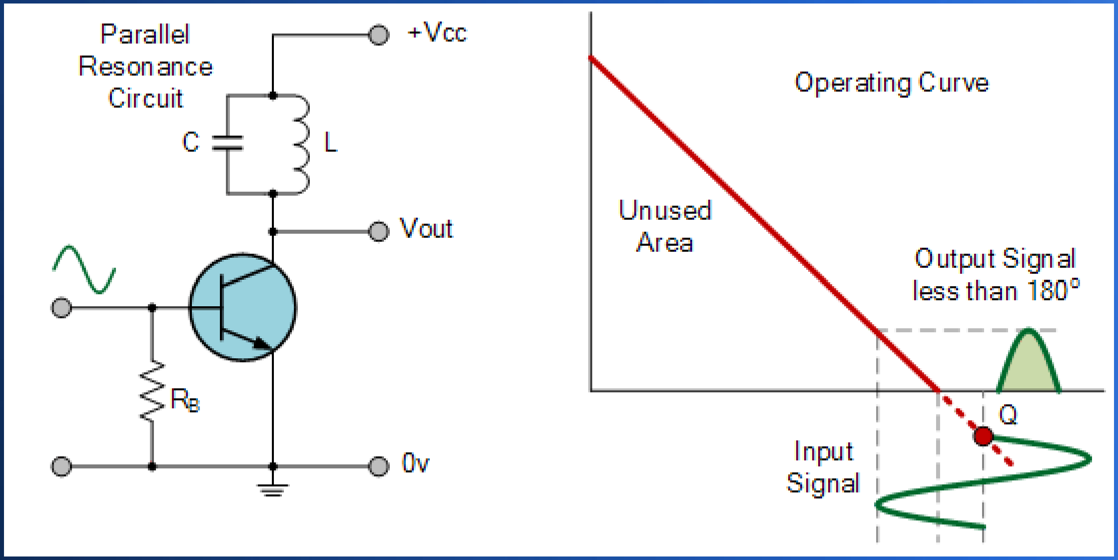


Class C Amplifier

- he Class C Amplifier design has the greatest efficiency but the poorest linearity of the classes of amplifiers mentioned here. The previous classes, A, B and AB are considered linear amplifiers, as the output signals amplitude and phase are linearly related to the input signals amplitude and phase.
- However, the class C amplifier is heavily biased so that the output current is zero for more than one half of an input sinusoidal signal cycle with the transistor idling at its cut-off point. In other words, the conduction angle for the transistor is significantly less than 180 degrees, and is generally around the 90 degrees area.
- While this form of transistor biasing gives a much improved efficiency of around 80% to the amplifier, it introduces a very heavy distortion of the output signal. Therefore, class C amplifiers are not good good audio amplifiers!!!

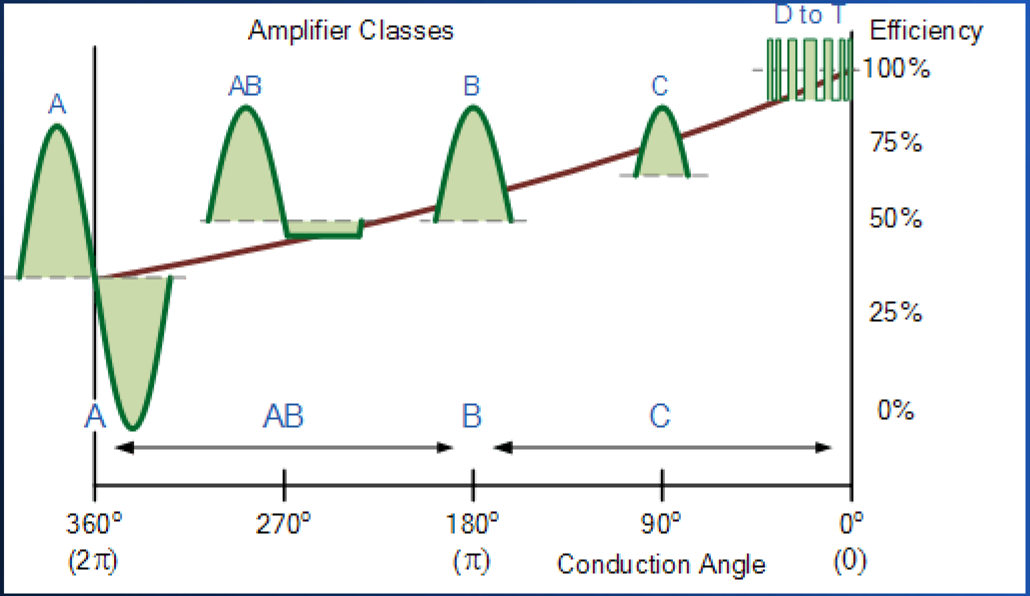
Class C Amplifier

- Due to its heavy audio distortion, class C amplifiers are commonly used in high frequency sine wave oscillators and certain types of radio frequency amplifiers, where the pulses of current produced at the amplifiers output can be converted to complete sine waves of a particular frequency by the use of LC resonant circuits in its collector circuit.



Summary

- We have seen that the quiescent DC operating point (Q-point) of an amplifier determines the amplifier classification. Setting the position of the Q-point at half way on the load line of the amplifiers characteristics curve, the amplifier will operate as a class A amplifier. By moving the Q-point lower down the load line changes the amplifier into a class AB, B or C amplifier.



Other Amplifier Classes

- As well as audio amplifiers there are a number of high efficiency Amplifier Classes relating to switching amplifier designs that use different switching techniques to reduce power loss and increase efficiency. Some amplifier class designs listed below use RLC resonators or multiple power-supply voltages to reduce power loss, or are digital DSP (digital signal processing) type amplifiers which use pulse width modulation (PWM) switching techniques.
- Class D Amplifier – A Class D audio amplifier is basically a non-linear switching amplifier or PWM amplifier. Class-D amplifiers theoretically can reach 100% efficiency, as there is no period during a cycle were the voltage and current waveforms overlap as current is drawn only through the transistor that is on.
-

Other Amplifier Classes

- Class F Amplifier – Class-F amplifiers boost both efficiency and output by using harmonic resonators in the output network to shape the output waveform into a square wave. Class-F amplifiers are capable of high efficiencies of more than 90% if infinite harmonic tuning is used.
- Class G Amplifier – Class G offers enhancements to the basic class AB amplifier design. Class G uses multiple power supply rails of various voltages and automatically switches between these supply rails as the input signal changes. This constant switching reduces the average power consumption, and therefore power loss caused by wasted heat.
- Class I Amplifier – The class I amplifier has two sets of complementary output switching devices arranged in a type of parallel push-pull configuration. Both sets of switching devices sample the same input waveform. Thus one device switches the positive half of the waveform, while the other switches the negative half. This switching action is similar to that for the class B amplifier.

Other Amplifier Classes

- With no input signal applied, or when a signal reaches the zero crossing point, the switching devices are both turned ON and OFF simultaneously with a 50% PWM duty cycle cancelling out any high frequency signals.
- To produce the positive half of the output signal, the output of the positive switching device is increased in duty cycle while the negative switching device is decreased by the same and vice versa. The two switching signal currents are said to be interleaved at the output, giving the class I amplifier the named of: “interleaved PWM amplifier” operating at switching frequencies in excess of 250kHz.
- Class S Amplifier – A class S power amplifier is a non-linear switching mode amplifier similar in operation to the class D amplifier. The class S amplifier converts analogue input signals into digital square wave pulses by a delta-sigma modulator, and amplifies them to increases the output power before finally being demodulated by a band pass filter. As the digital signal of this switching amplifier is always either fully “ON” or “OFF” (theoretically zero power dissipation), efficiencies reaching 100% are possible.

Other Amplifier Classes

- Class T Amplifier – The class T amplifier is another type of digital switching amplifier design. Class T amplifiers are starting to become more popular these days as an audio amplifier design due to the existence of digital signal processing (DSP) chips and multi-channel surround sound amplifiers as it converts analogue signals into digital pulse width modulated (PWM) signals for amplification increasing the amplifiers efficiency. Class T amplifier designs combine both the low distortion signal levels of class AB amplifier and the power efficiency of a class D amplifier.

Operational Amplifiers

- Operational Amplifiers
- Operational Amplifiers, or Op-amps as they are more commonly called, are one of the basic building blocks of Analogue Electronic Circuits.
- In this operational amplifier basics tutorial, we will see that Operational amplifiers are linear devices which have all the properties required for nearly ideal DC amplification. They are used extensively in signal conditioning, filtering or to perform mathematical operations such as add, subtract, integration and differentiation.
- An Operational Amplifier, or op-amp for short, is fundamentally a voltage amplifying device designed to be used with external feedback components such as resistors and capacitors between its output and input terminals. These feedback components determine the resulting function or “operation” of the amplifier and by virtue of the different feedback configurations whether resistive, capacitive or both, the amplifier can perform a variety of different operations, giving rise to its name of “Operational Amplifier”.

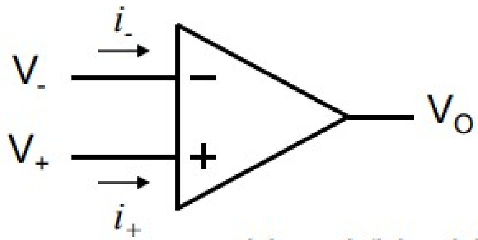
Analouge Electronics

- An Operational Amplifier is basically a three-terminal device which consists of two high impedance inputs. One of the inputs is called the Inverting Input, marked with a negative or “minus” sign, (–). The other input is called the Non-inverting Input, marked with a positive or “plus” sign (+).
- A third terminal represents the operational amplifiers output port which can both sink and source either a voltage or a current. In a linear operational amplifier, the output signal is the amplification factor, known as the amplifiers gain (A) multiplied by the value of the input signal and depending on the nature of these input and output signals, there can be four different classifications of operational amplifier gain.

Ideal Op-Amp charatoristics

Ideal Op-Amp

- infinite voltage gain ($A = \infty$, $V_- = V_+$ for finite V_O)
- infinite input impedance ($i_- = i_+ = 0$)
- zero output impedance
- infinite bandwidth
- zero input offset



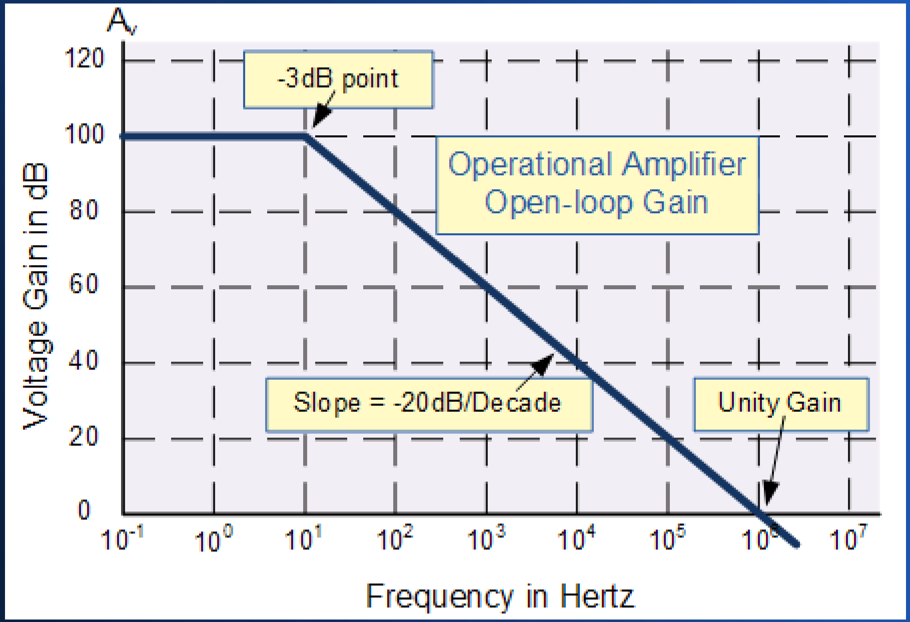
$V_O = A(V_+ - V_-)$

Analouge Electronics

- Operational Amplifier Basics of Classification
- Voltage – Voltage “in” and Voltage “out”
- Current – Current “in” and Current “out”
- Transconductance – Voltage “in” and Current “out”
- Transresistance – Current “in” and Voltage “out”
- Since most of the circuits dealing with operational amplifiers are voltage amplifiers, we will limit the tutorials in this section to voltage amplifiers only, (Vin and Vout).
- The output voltage signal from an Operational Amplifier is the difference between the signals being applied to its two individual inputs. In other words, an op-amps output signal is the difference between the two input signals as the input stage of an Operational Amplifier is in fact a differential amplifier as shown later.

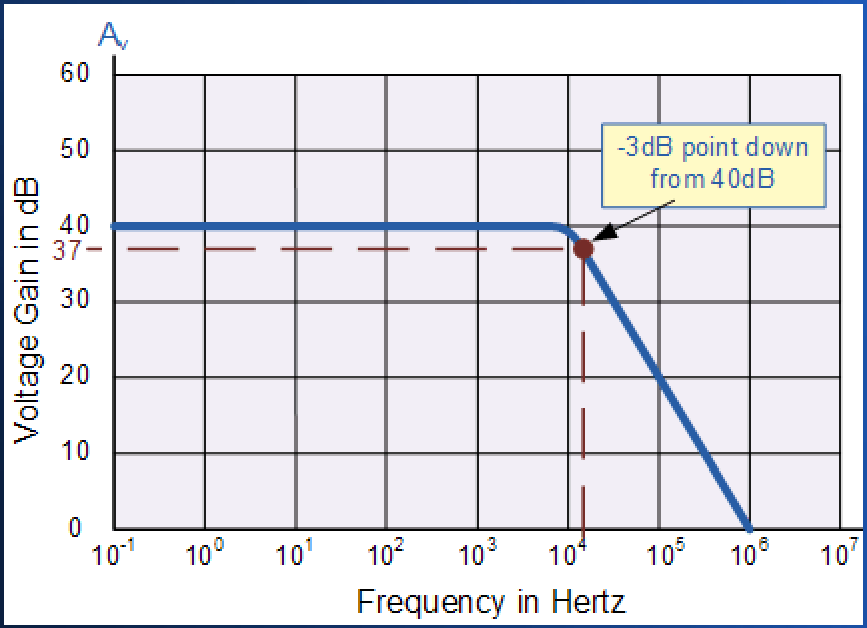
Analouge Electronics

- For a typical operational amplifier, this open loop gain can be as high as 100dB at DC (zero Hz). Generally, an op-amps output gain decreases linearly as frequency increases down to “Unity Gain” or 1, at about 1MHz. This effect is shown in the following open loop gain response curve.



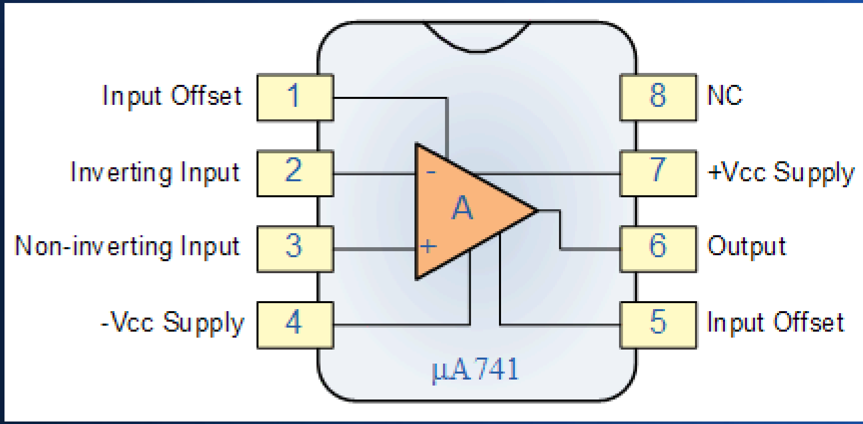
Analouge Electronics

- An Operational Amplifiers Bandwidth
- The operational amplifiers bandwidth is the frequency range over which the voltage gain of the amplifier is above 70.7% or -3dB (where 0dB is the maximum) of its maximum output value as shown below.



Analouge Electronics

- There are a very large number of operational amplifier IC’s available to suit every possible application from standard bipolar, precision, high-speed, low-noise, high-voltage, etc, in either standard configuration or with internal Junction FET transistors.
- Operational amplifiers are available in IC packages of either single, dual or quad op-amps within one single device. The most commonly available and used of all the operational amplifiers in basic electronic kits and projects is the industry standard µA-741.



Operational Amplifiers

- There are three basic operational amplifier configurations.
- Buffer/voltage follower: Has unity gain. What possible use is this? It can isolate a driven circuit from its signal source (provide protection and higher input impedance than the driven circuit itself would provide;
- Inverting amplifier: amplifies voltage, in the opposite polatity to the input
- Non-inverting amplifier: amplifies voltage in the same polarity as the input signal.

Analouge Electronics

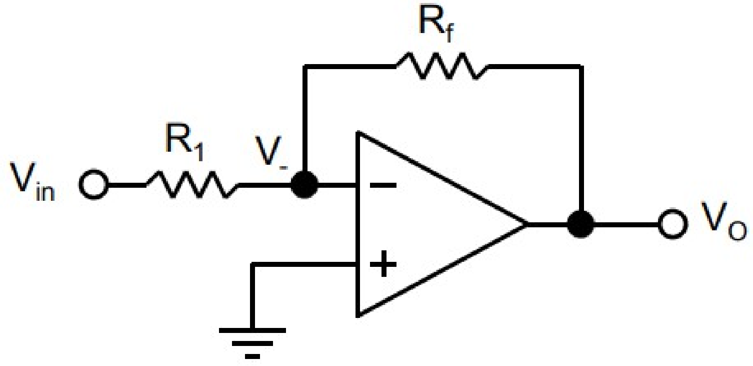
- Inverting Amplifier Configuration
  - current through R_1 = current through R_f ($\because i_- = i_+ = 0$)

    $(V_{in} - V_1) / R_1 = (V_- - V_O) / R_f$

  - and

    $V_- = V_+ = 0$

  - $V_O = -(R_f / R_1) \cdot V_{in}$

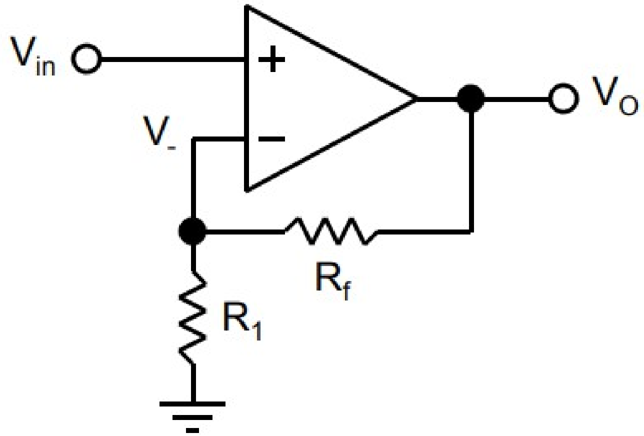


Operational Amplifiers

- Non-Inverting Amplifier Configuration
  - current through R_1 = current through R_f ($\because i_- = i_+ = 0$)

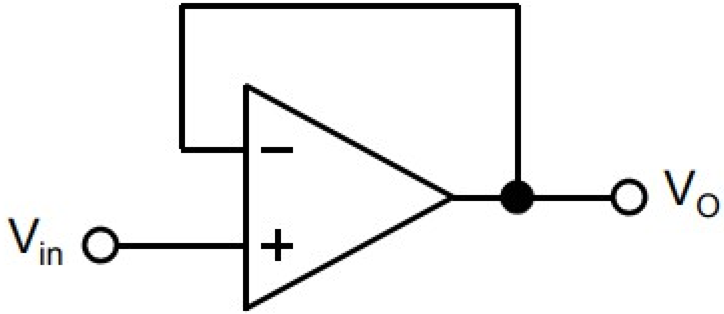
    $(V_O - V_-) / R_f = V_- / R_1$
  - and

    $V_- = V_+ = V_{in}$
  - $V_O = (1 + R_f / R_1) \cdot V_{in}$



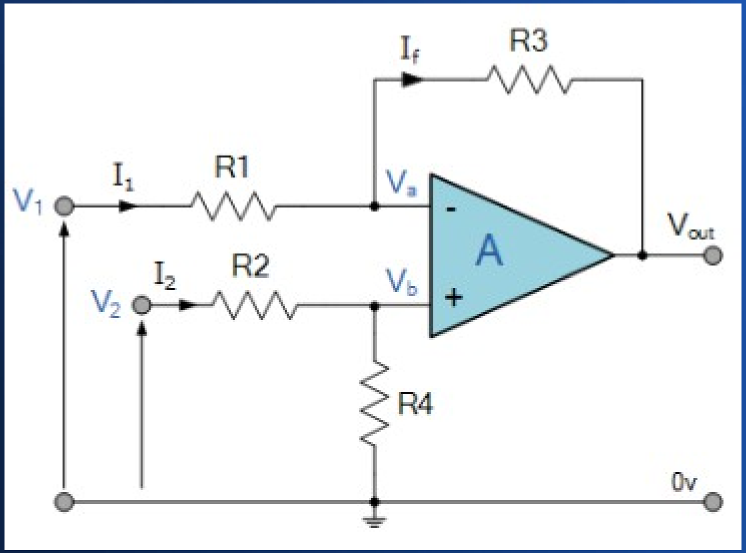
Operational Amplifiers

- Voltage Follower (Unit Gain Buffer)
  - unit-gain non-inverting amplifier
  - $V_O = V_{in}$



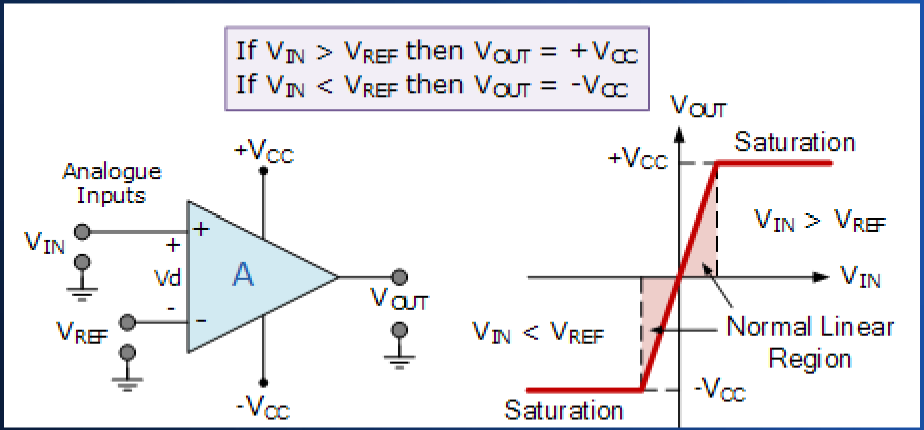
Operational Amplifiers

- The differential amplifier is a voltage subtractor circuit which produces an output voltage proportional to the voltage difference of two input signals applied to the inputs of the inverting and non-inverting terminals of an operational amplifier.



Operational Amplifiers

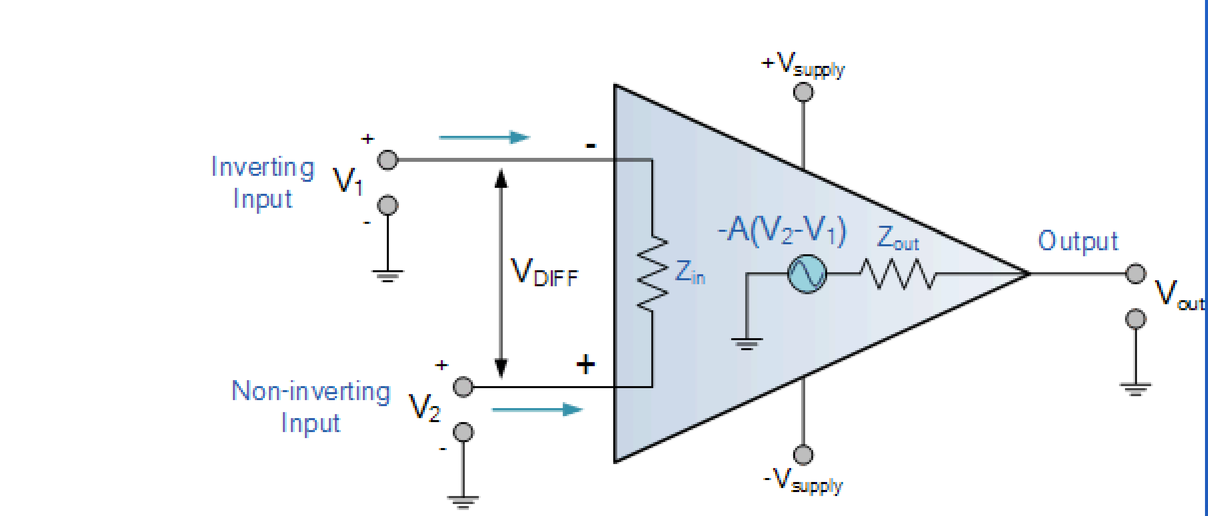
- Comparator: Compares voltages in the two inputs.
- The open-loop op-amp comparator is an analogue circuit that operates in its non-linear region as changes in the two analogue inputs, V+ and V- causes it to behave like a digital bistable device as triggering causes it to have two possible output states, +Vcc or -Vcc. Then we can say that the voltage comparator is essentially a 1-bit analogue to digital converter, as the input signal is analogue but the output behaves digitally.



Operational Amplifiers

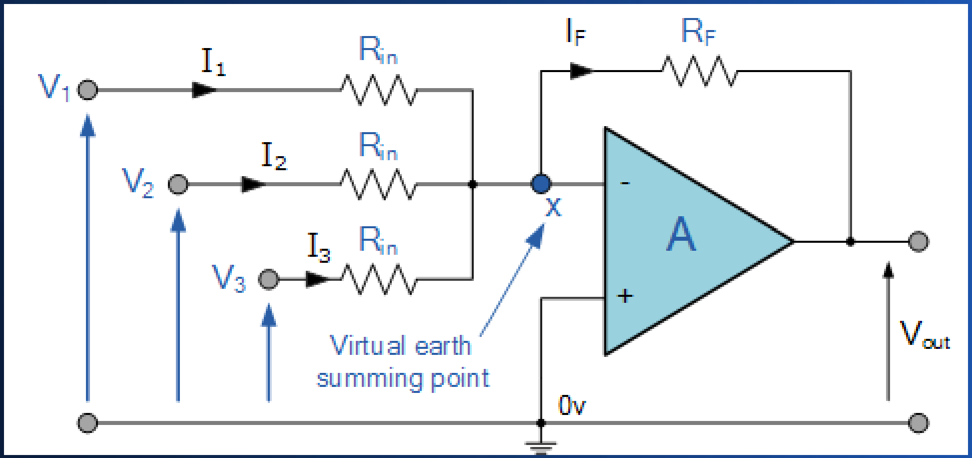
- Op-amp Parameter and Idealised Characteristic
- Open Loop Gain, (Avo)
- Infinite – The main function of an operational amplifier is to amplify the input signal and the more open loop gain it has the better. Open-loop gain is the gain of the op-amp without positive or negative feedback and for such an amplifier the gain will be infinite but typical real values range from about 20,000 to 200,000.

Equivalent Circuit of an Ideal Operational Amplifier



Operational Amplifiers

- Summing amplifier:
- We saw previously in the inverting operational amplifier that the inverting amplifier has a single input voltage, (Vin) applied to the inverting input terminal. If we add more input resistors to the input, each equal in value to the original input resistor, (Rin) we end up with another operational amplifier circuit called a Summing Amplifier, “summing inverter” or even a “voltage adder” circuit as shown below.



Operational Amplifiers

- Input impedance, (ZIN)
- Infinite – Input impedance is the ratio of input voltage to input current and is assumed to be infinite to prevent any current flowing from the source supply into the amplifiers input circuitry (IIN = 0). Real op-amps have input leakage currents from a few pico-amps to a few milli-amps.
- Output impedance, (ZOUT)
- Zero – The output impedance of the ideal operational amplifier is assumed to be zero acting as a perfect internal voltage source with no internal resistance so that it can supply as much current as necessary to the load. This internal resistance is effectively in series with the load thereby reducing the output voltage available to the load. Real op-amps have output impedances in the 100-20kΩ range.
-

Operational Amplifiers

- Bandwidth, (BW)
- Infinite – An ideal operational amplifier has an infinite frequency response and can amplify any frequency signal from DC to the highest AC frequencies so it is therefore assumed to have an infinite bandwidth. With real op-amps, the bandwidth is limited by the Gain-Bandwidth product (GB), which is equal to the frequency where the amplifiers gain becomes unity.
- Offset Voltage, (VIO)
- Zero – The amplifiers output will be zero when the voltage difference between the inverting and the non-inverting inputs is zero, the same or when both inputs are grounded. Real op-amps have some amount of output offset voltage.

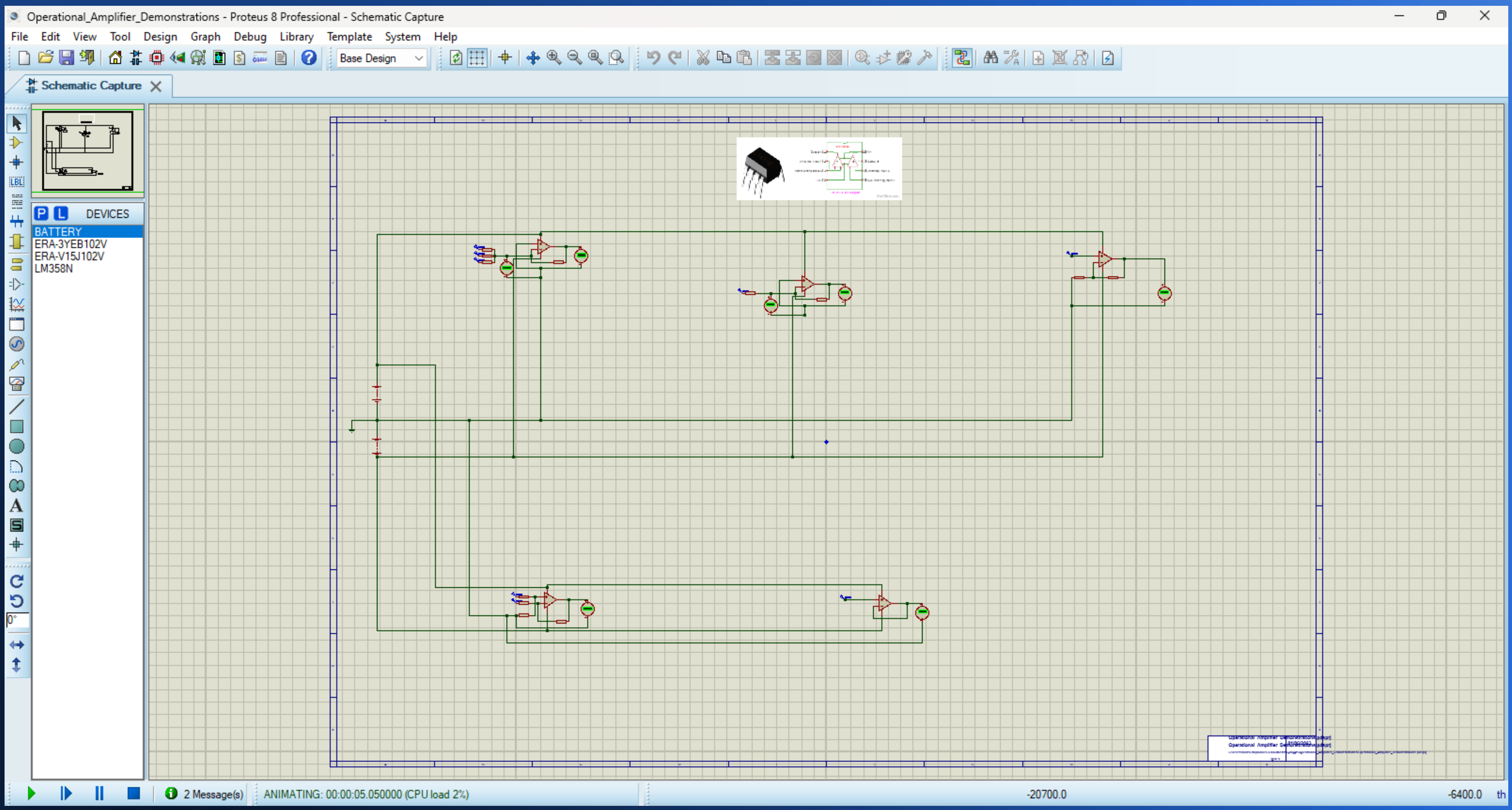
Operational Amplifiers

- From these “idealized” characteristics above, we can see that the input resistance is infinite, so no current flows into either input terminal (the “current rule”) and that the differential input offset voltage is zero (the “voltage rule”). It is important to remember these two properties as they will help us understand the workings of the Operational Amplifier with regards to the analysis and design of op-amp circuits.
- However, real Operational Amplifiers such as the commonly available uA741, for example do not have infinite gain or bandwidth but have a typical “Open Loop Gain” which is defined as the amplifiers output amplification without any external feedback signals connected to it.

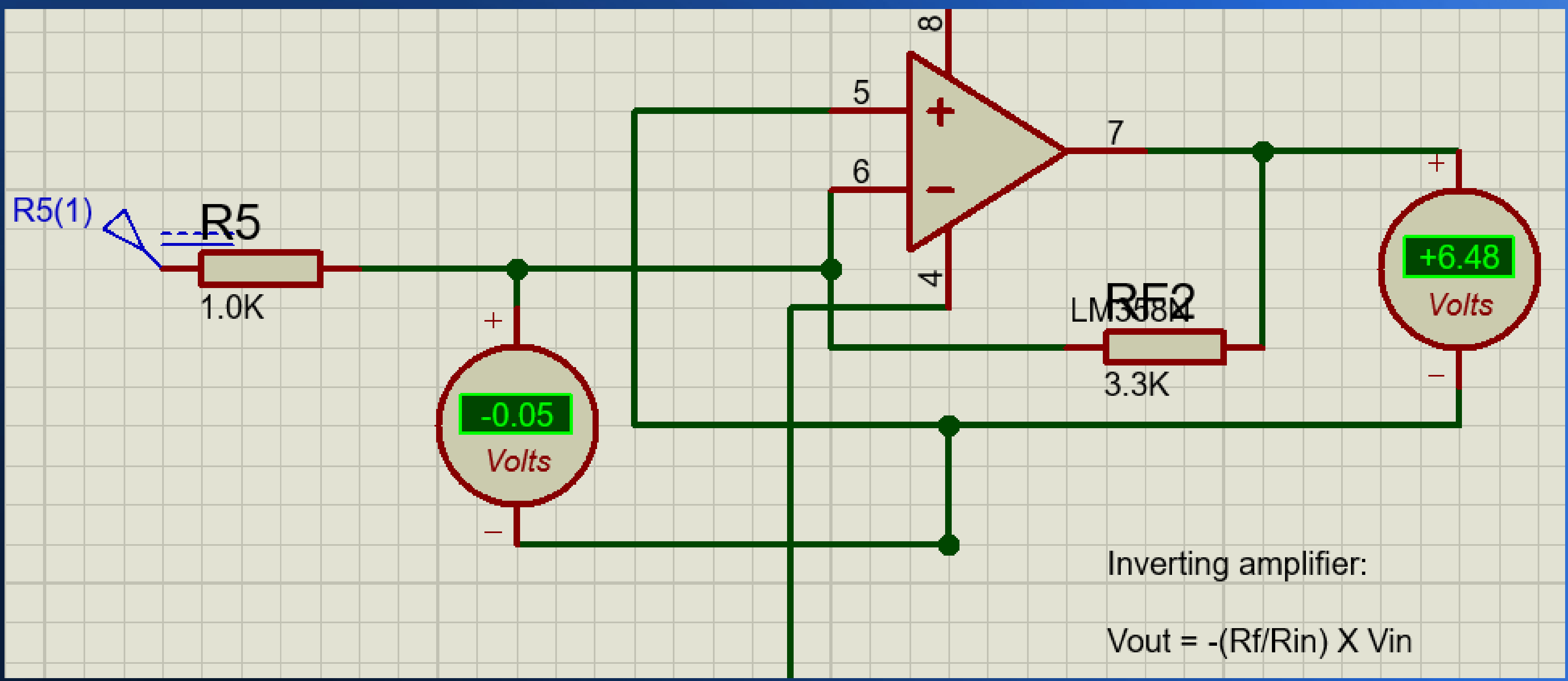
Operational amplifier: formulae

- Comparator: $\bullet\ V_{\text{out}} = \begin{cases} V_{S+} & V_1 > V_2 \\ V_{S-} & V_1 < V_2 \end{cases}$
- Inverting Amplifier: $V_{\text{out}} = -\frac{R_f}{R_{\text{in}}} V_{\text{in}}$
- Non-inverting amplifier: $V_{\text{out}} = V_{\text{in}} \left(1 + \frac{R_2}{R_1}\right)$
- Differential Amplifier: $V_{\text{out}} = \frac{(R_f + R_1)\, R_g}{(R_g + R_2)\, R_1} V_2 - \frac{R_f}{R_1} V_1$
- Voltage follower/buffer: $V_{\text{out}} = V_{\text{in}}$
- Note: there are also many hybrids.

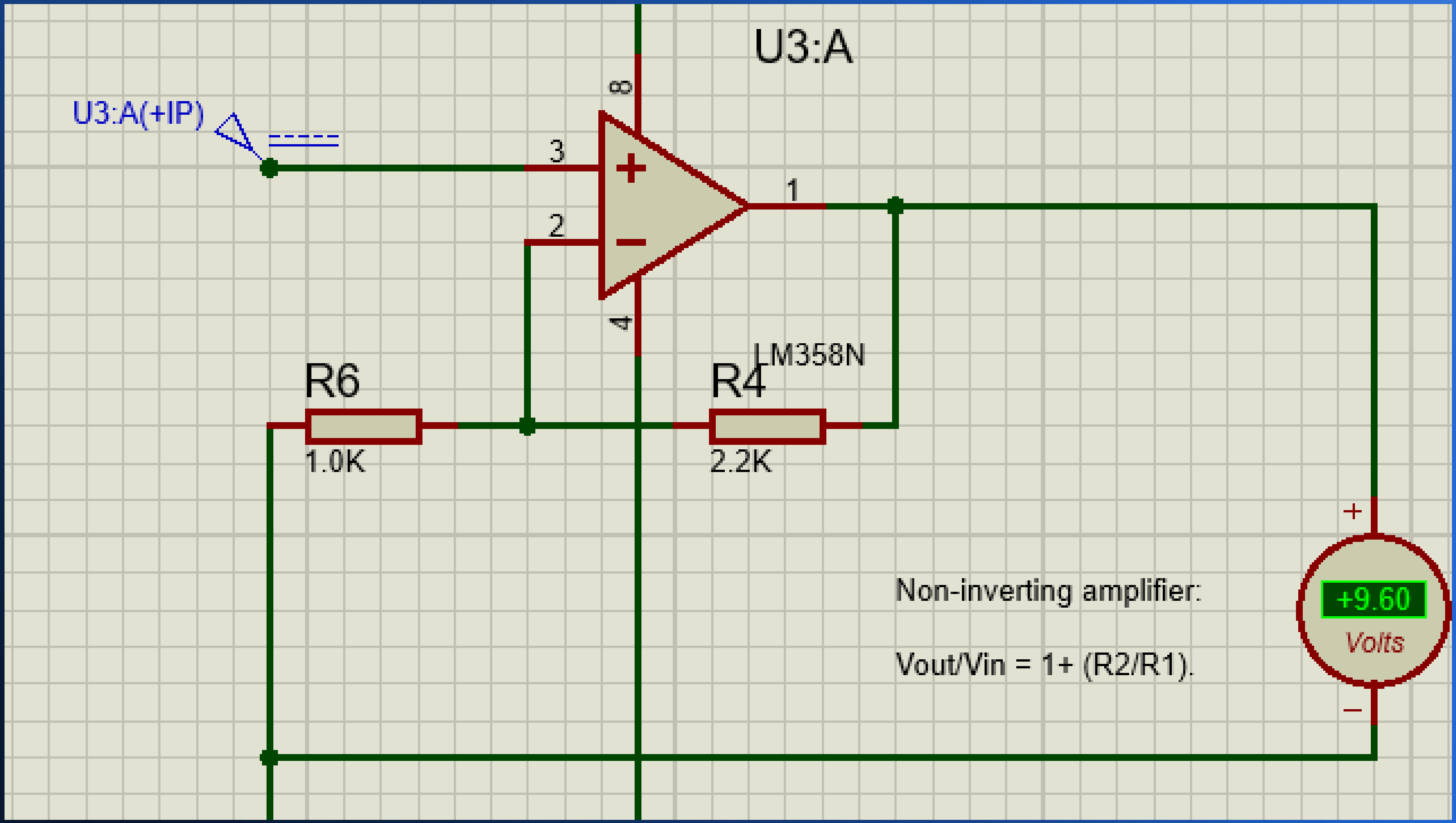
Operational Amplifier Simumation



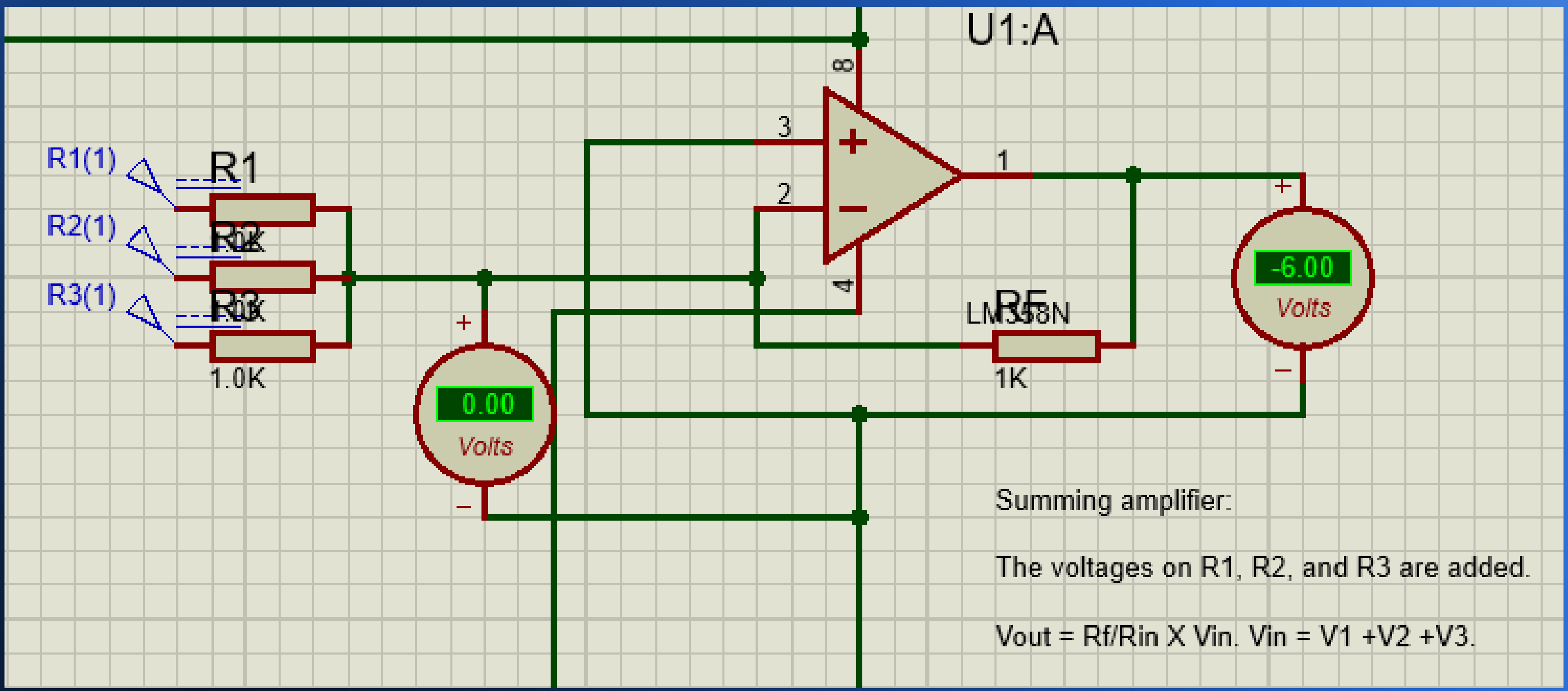
Inverting Amplifier



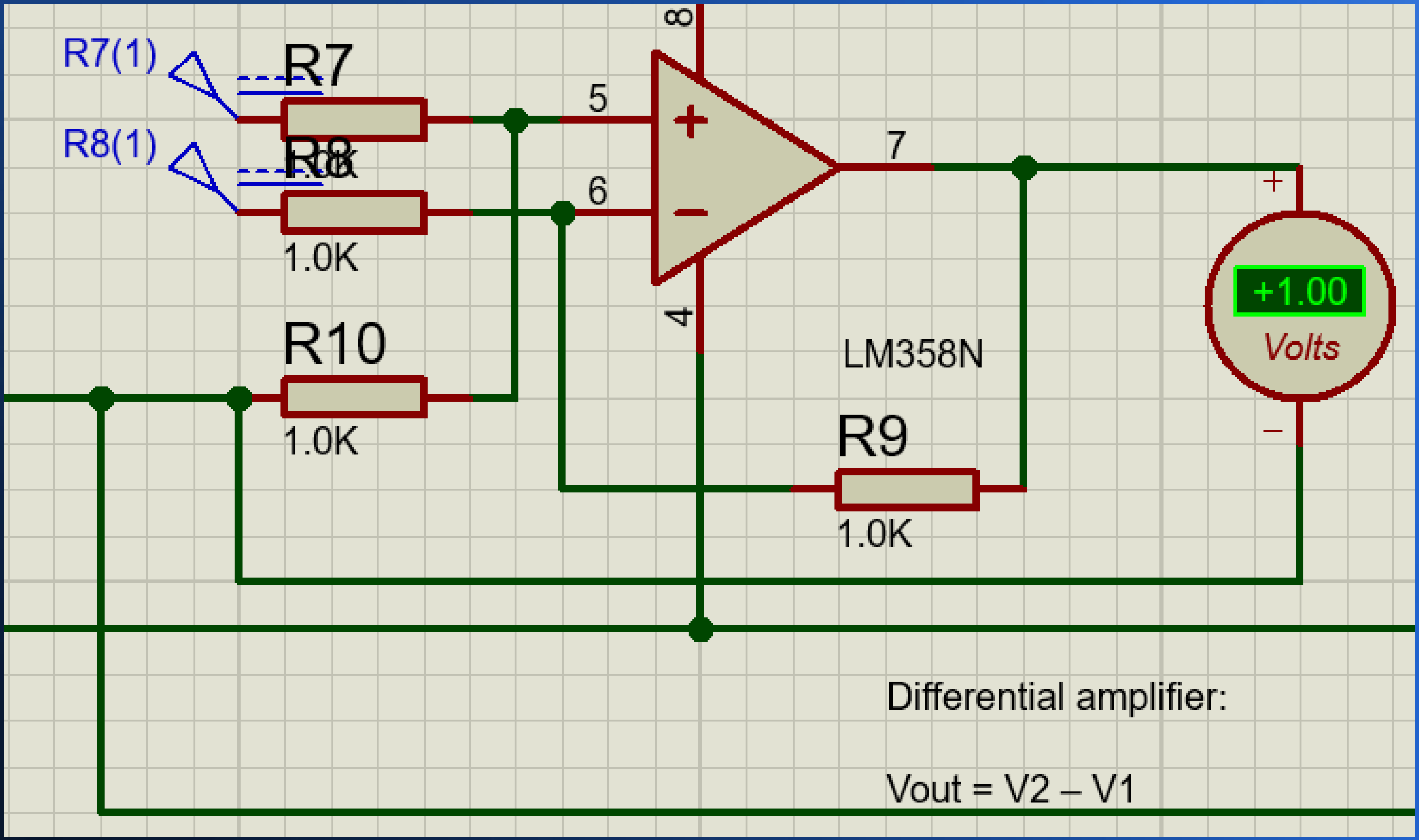
Non-Inverting Amplifier



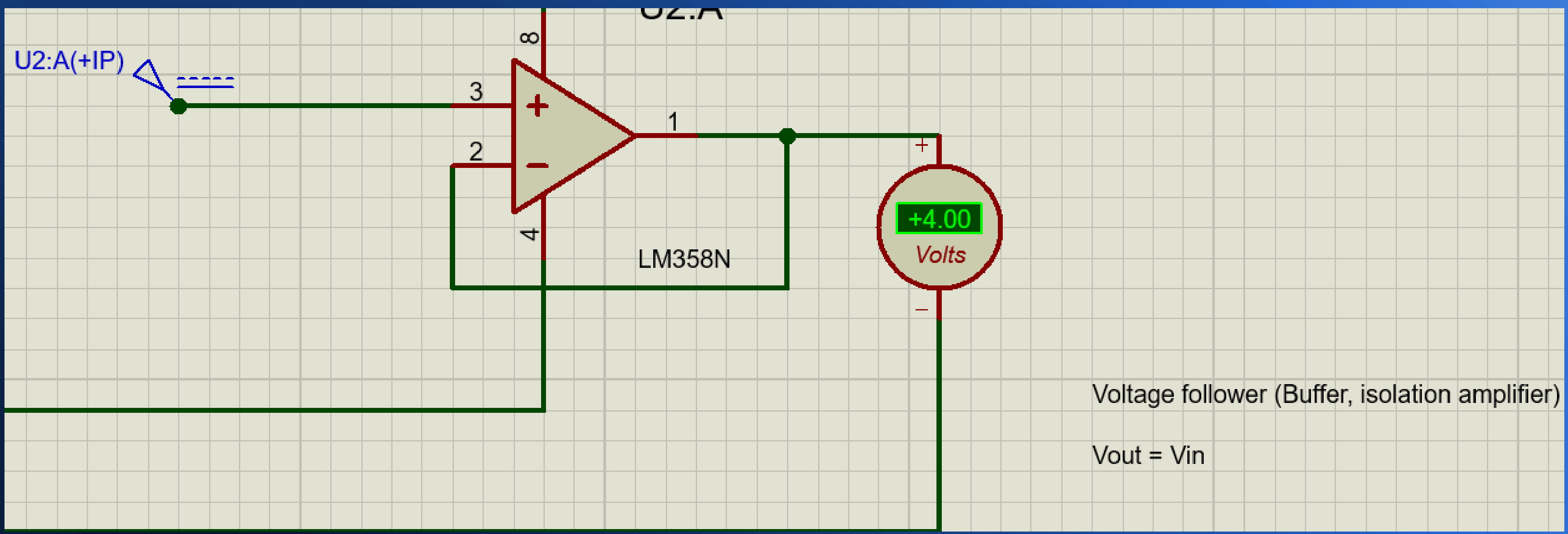
Summing Amplifier



Differential Amplifier



Voltage Follower/Buffer



Oscillators

- Oscillators are electronic circuits that generate a continuous periodic waveform at a precise frequency
- An LC Oscillator converts a DC input (the supply voltage) into an AC output (the waveform). This output waveform can have a wide range of different shapes and frequencies, and can be either complex in shape, or be a simple pure sine wave depending upon the application.
- Oscillators are used in many pieces of test equipment producing either sinusoidal sine waves, square, sawtooth or triangular shaped waveforms or just a train of repetative pulses of a variable or constant width. LC Oscillators are commonly used in radio-frequency circuits because of their good phase noise characteristics and their ease of implementation.
- We met amplifiers already.
- An Oscillator is basically an Amplifier with “Positive Feedback”, or regenerative feedback (in-phase) and one of the many problems in electronic circuit design is stopping amplifiers from oscillating while trying to get oscillators to oscillate.

Oscillators

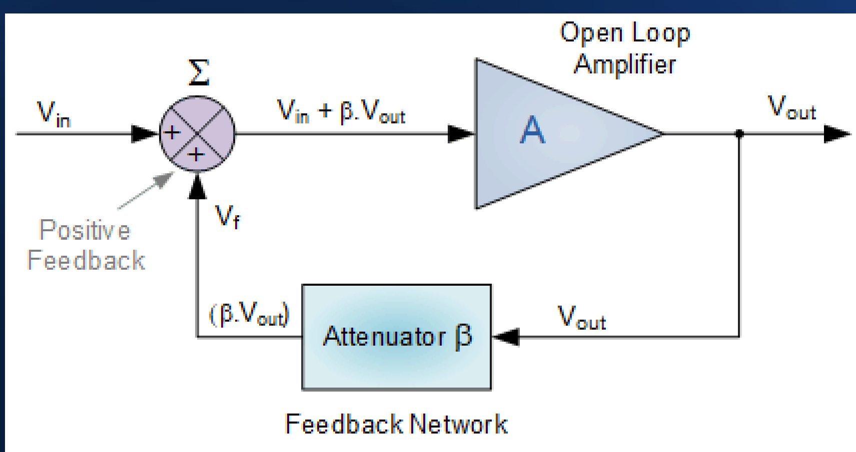
- Oscillators work because they overcome the losses of their feedback resonator circuit either in the form of a capacitor, inductor or both in the same circuit by applying DC energy at the required frequency into this resonator circuit. In other words, an oscillator is a an amplifier which uses positive feedback that generates an output frequency without the use of an externally applied input signal.
- Thus Oscillators are self sustaining circuits generating an periodic output waveform at a single sinusoidal frequency. Thus for any electronic circuit to operate as an oscillator, it must contain the following three characteristics.
- Some form of Amplification
- Positive Feedback (regeneration)
- A Frequency determine feedback network

Oscillators

- So if A = amplification (gain, open loop) and ß = feedback;
- If Aß = 1, the amplifier becomes an oscillator.
- You can prove this using an audio amplifier and a microphone – you get the deafening screech of audio frequency feedback!
- An oscillator has a small signal feedback amplifier with an open-loop gain equal too or slightly greater than one for oscillations to start but to continue oscillations the average loop gain must return to unity. In addition to these reactive components, an amplifying device such as an Operational Amplifier or Bipolar Transistor is required.
- Unlike an amplifier there is no external AC input signal required to cause the Oscillator to work, as the DC supply energy is converted by the oscillator into AC energy at the required frequency.

Oscillators

- This is what happens in an amplifier without any feedback. Here Aß = 1 so the amplifier does not oscillate.
- But when Aß = 1, we have a closed loop....



$$\text{Gain, } A = \frac{V_{OUT}}{V_{IN}}$$

$$\therefore V_{OUT} = A \times V_{IN}$$

Oscillators

- Now we have converted our amplifier into an oscillator! You can make your audio amplifier into an oscillator and holding the microphone near the speaker (and deafen everyone in the room!).

$$A\left(V_{IN} + \beta V_{OUT}\right) = V_{OUT}$$

β is the feedback fraction

$$A \times V_{IN} + A\beta \times V_{OUT} = V_{OUT}$$

$A\beta$ = the voltage loop gain

$$A \times V_{IN} = V_{OUT}(1 - A\beta)$$

$1 - A\beta$ = positive feedback factor

$$\therefore \frac{V_{OUT}}{V_{IN}} = Gv = \frac{A}{1 - A\beta}$$

Gv = the closed loop voltage gain

Oscillators

- Therefore: oscillators are electric circuits that generate a continuous voltage output waveform at a required single frequency. Inductors, capacitors or resistors are used to form a frequency selective resonant circuit, which is basically a passive band-pass filter that allows the desired frequency to pass, and a feedback network.
- The feedback network “feeds” a small percentage of the output signal back to the input side in order to keep the circuit oscillating. The amount of positive feedback used must be large enough to overcome any circuit losses so that oscillations can be sustained indefinately.
- The feedback network is basically an attenuation circuit that has a voltage gain of less than one ($\beta < 1$). Oscillations start when $A\beta > 1$ and then returns to unity ($A\beta = 1$) once oscillations are sustained.

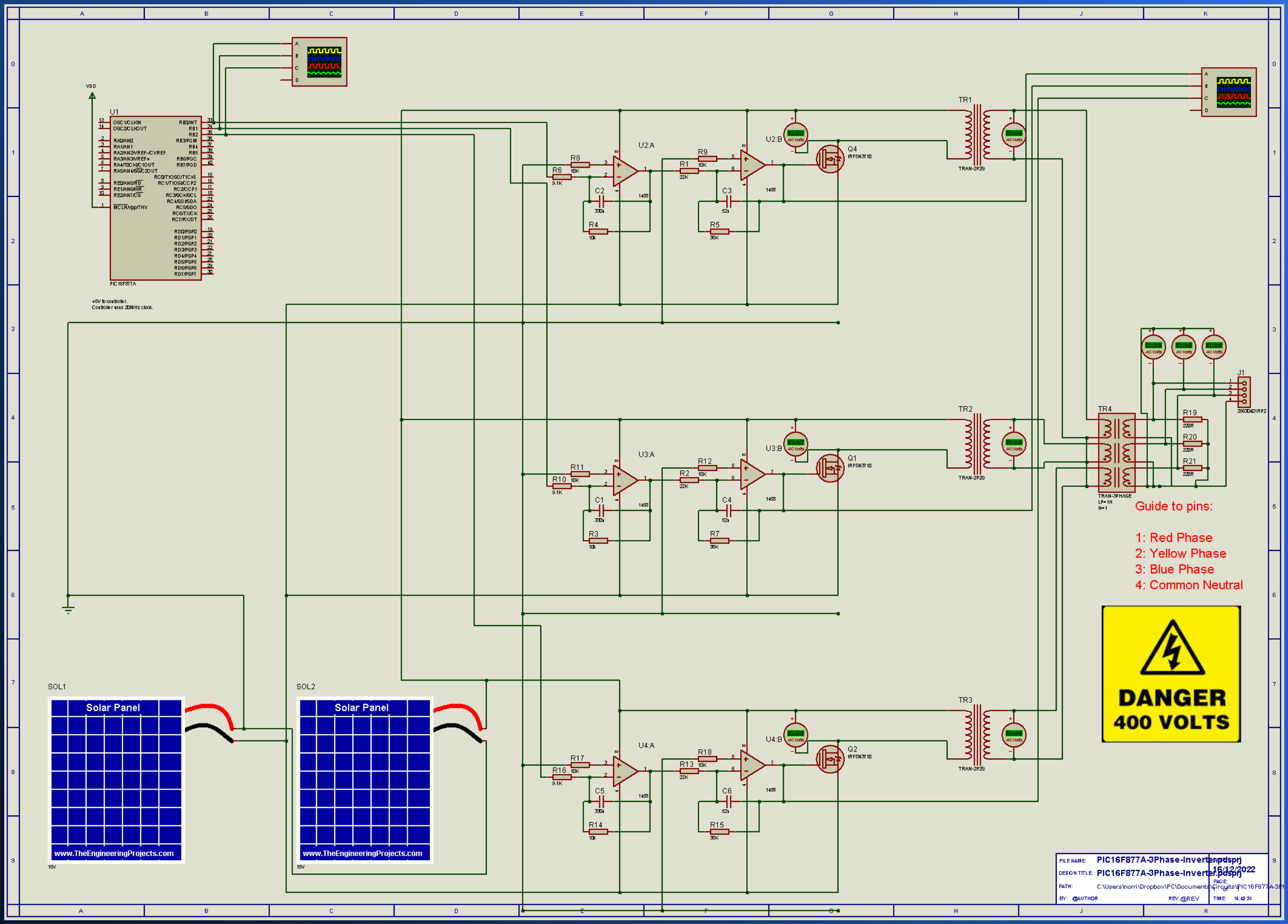
Oscillators

- The LC oscillators frequency is controlled using a tuned or resonant inductive/capacitive (LC) circuit with the resulting output frequency being known as the Oscillation Frequency. By making the oscillators feedback a reactive network the phase angle of the feedback will vary as a function of frequency and this is called Phase-shift.
- Oscillators can be categorised into two types...
- Sinusoidal Oscillators – these are known as Harmonic Oscillators and are generally a “LC Tuned-feedback” or “RC tuned-feedback” type Oscillator that generates a purely sinusoidal waveform which is of constant amplitude and frequency.
- Non-Sinusoidal Oscillators – these are known as Relaxation Oscillators and generate complex non-sinusoidal waveforms that changes very quickly from one condition of stability to another such as “Square-wave”, “Triangular-wave” or “Sawtoothed-wave” type waveforms.

Oscillators

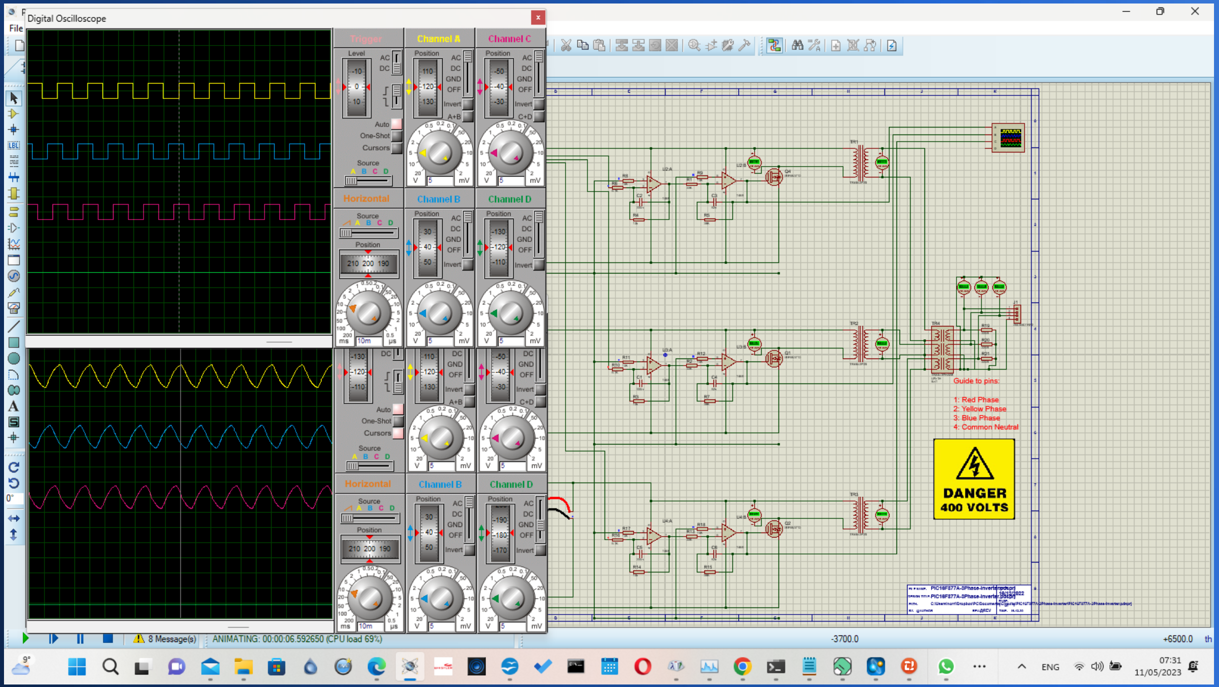
- A case in point...
- The mains supply is sinusoidal. Hence an inverter needs to generate the best approximation to a sinewave possible.
- The frequency (60Hz in North and south America; 50Hz in the rest of the world needs to be accurate;
- A square wave will cause a huge amount of RFI (radio frequency interferance), besides huge inductive distortion where inductive loads are involved, e.g motors and transformers, and my cause failure of switched mode supplies – such as those in computers and USB chargers.
- And some items, such as florescent lighting will make a lot of noise!
- But digital signal generators uch as microcontrollers (used in many of my own designs output a square wave! See next slide.

Oscillators



Oscillators

- In this example, a PIC16F877A microcontroller generates 3 clock signals (this is a three phase design). A microcontroller outputs a square wave at 50Hz exactly. But to convert this to a sinewave a filter cotaining op-amps (which we met already) is utilised. And here is the result... The vitual oscilloscope shows the waveform before and after the filter.



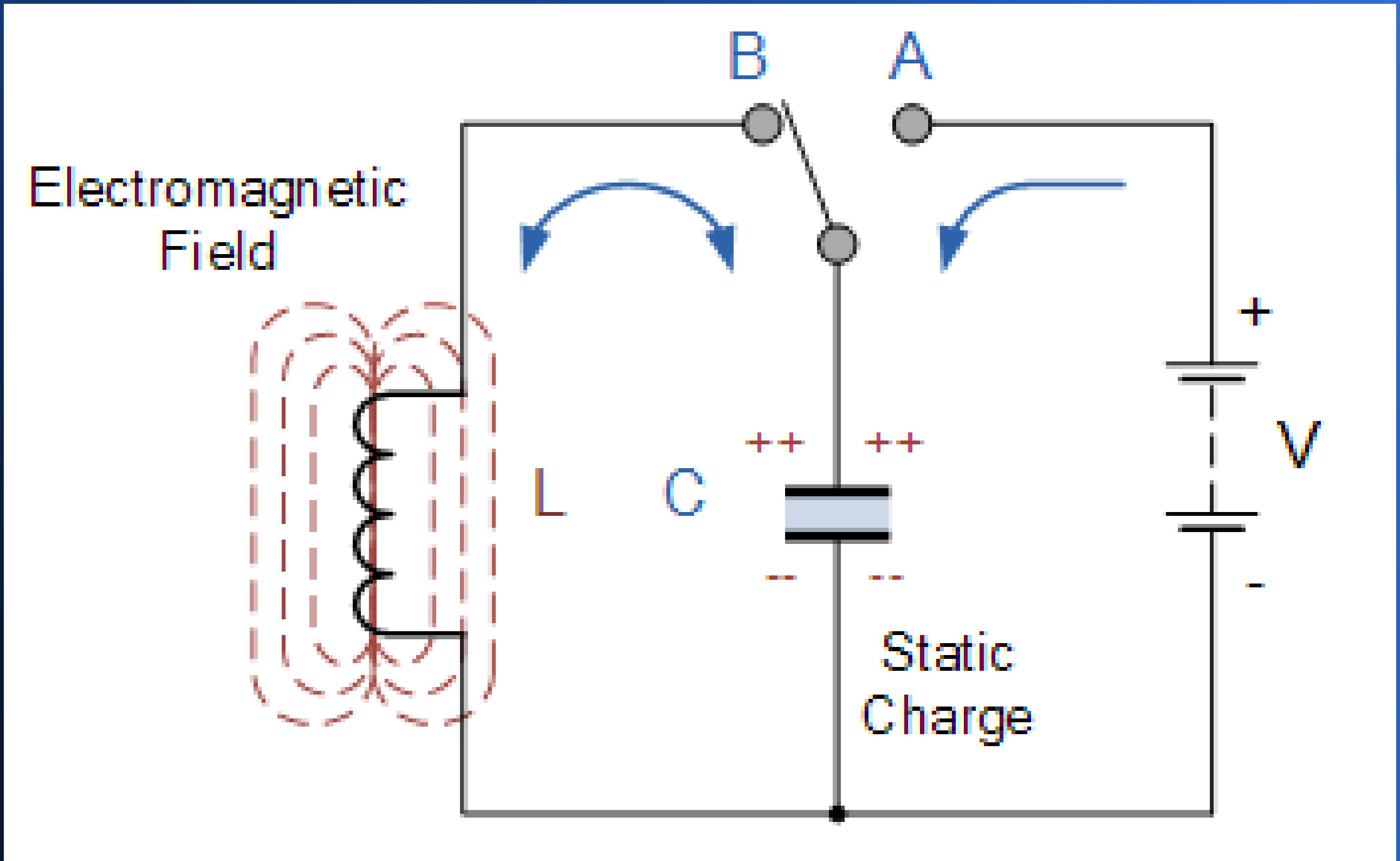
Oscillators

- Note quite a perfect sinewave. BUT a huge improvement over most of the commercially available inverters on the market! I am an old hand at this kind of design. And I intend to sell solar powered inverters lie these in West Africa.
- Oscillator Resonance:
- When a constant voltage of varying frequency is applied to a circuit consisting of an inductor, capacitor and resistor the reactance of both the Capacitor/Resistor and Inductor/Resistor circuits will change both the amplitude and the phase of the output signal as compared to the input signal due to the reactance of the components used.
- At high frequencies the reactance of a capacitor is very low acting as a short circuit while the reactance of the inductor is high acting as an open circuit. At low frequencies the reverse is true, the reactance of the capacitor acts as an open circuit and the reactance of the inductor acts as a short circuit!

Oscillators

- Between these two extremes the combination of the inductor and capacitor produces a “Tuned” or “Resonant” circuit that has a Resonant Frequency, (fr) in which the capacitive and inductive reactance’s are equal and cancel out each other, leaving only the resistance of the circuit to oppose the flow of current. This means that there is no phase shift as the current is in phase with the voltage. Consider this example circuit on the next slide...
- The circuit consists of an inductive coil, L and a capacitor, C. The capacitor stores energy in the form of an electrostatic field and which produces a potential (static voltage) across its plates, while the inductive coil stores its energy in the form of an electromagnetic field. The capacitor is charged up to the DC supply voltage, V by putting the switch in position A. When the capacitor is fully charged the switch changes to position B.

Oscillators



Oscillators

- The charged capacitor is now connected in parallel across the inductive coil so the capacitor begins to discharge itself through the coil. The voltage across C starts falling as the current through the coil begins to rise.
-
- This rising current sets up an electromagnetic field around the coil which resists this flow of current. When the capacitor, C is completely discharged the energy that was originally stored in the capacitor, C as an electrostatic field is now stored in the inductive coil, L as an electromagnetic field around the coils windings.
- As there is now no external voltage in the circuit to maintain the current within the coil, it starts to fall as the electromagnetic field begins to collapse. A back emf is induced in the coil ($e = -L\,di/dt$) keeping the current flowing in the original direction.

Oscillators

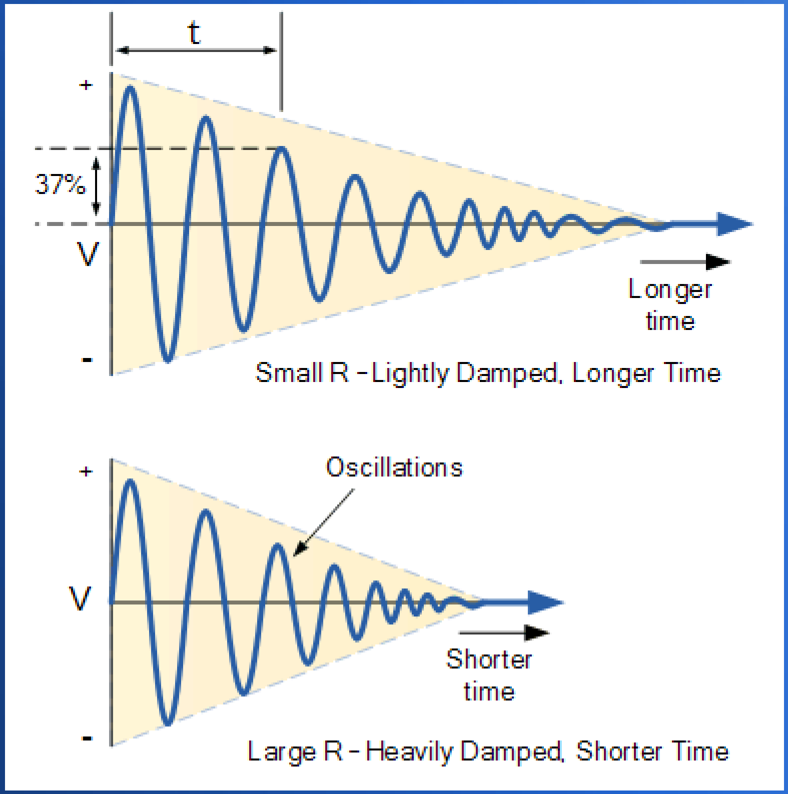
- This current charges up capacitor, C with the opposite polarity to its original charge. C continues to charge up until the current reduces to zero and the electromagnetic field of the coil has collapsed completely.
- The energy originally introduced into the circuit through the switch, has been returned to the capacitor which again has an electrostatic voltage potential across it, although it is now of the opposite polarity. The capacitor now starts to discharge again back through the coil and the whole process is repeated. The polarity of the voltage changes as the energy is passed back and forth between the capacitor and inductor producing an AC type sinusoidal voltage and current waveform.
- This process then forms the basis of an LC oscillators tank circuit and theoretically this cycling back and forth will continue indefinitely. However, things are not perfect and every time energy is transferred from the capacitor, C to inductor, L and back from L to C some energy losses occur which decay the oscillations to zero over time.

Oscillators

- This oscillatory action of passing energy back and forth between the capacitor, C to the inductor, L would continue indefinitely if it was not for energy losses within the circuit. Electrical energy is lost in the DC or real resistance of the inductors coil, in the dielectric of the capacitor, and in radiation from the circuit so the oscillation steadily decreases until they die away completely and the process stops.
- Then in a practical LC circuit the amplitude of the oscillatory voltage decreases at each half cycle of oscillation and will eventually die away to zero. The oscillations are then said to be “damped” with the amount of damping being determined by the quality or Q-factor of the circuit.
- This is the equivilent of a mass oscillating on a spring. (It is said to be a second order system since it has the ball and spring represent two storage elements of energy, just like our indoctor and capacitor here. Friction provides damping just like the resistor in this circuit).

Oscillators

- This is a second order response.
- A first order system has a time constant...
- But no oscillitory response.
- For example, a RC filter.



Oscillators

- The frequency of the oscillatory voltage depends upon the value of the inductance and capacitance in the LC tank circuit. We now know that for resonance to occur in the tank circuit, there must be a frequency point were the value of XC, the capacitive reactance is the same as the value of XL, the inductive reactance (XL = XC) and which will therefore cancel out each other out leaving only the DC resistance in the circuit to oppose the flow of current.
- If we now place the curve for inductive reactance of the inductor on top of the curve for capacitive reactance of the capacitor so that both curves are on the same frequency axes, the point of intersection will give us the resonance frequency point, (ƒr or ωr) as shown next...

Oscillators

- Where: fr is in Hertz, L is in Henries and C is in Farads.
- Then the frequency at which this will happen is given as:

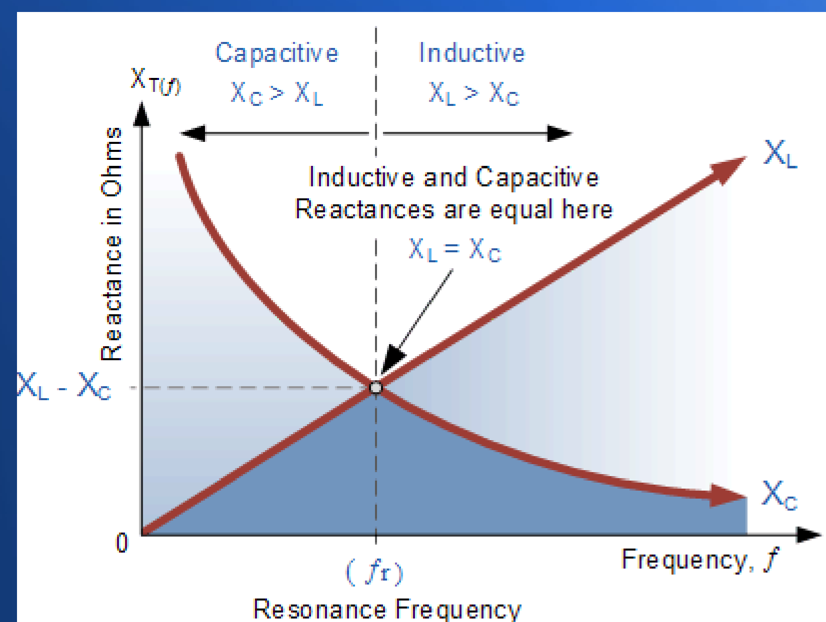
$$X_L = 2\pi f L \quad \text{and} \quad X_C = \frac{1}{2\pi f C}$$

$$\text{at resonance:} \quad X_L = X_C$$

$$\therefore \; 2\pi f L = \frac{1}{2\pi f C}$$

$$2\pi f^2 L = \frac{1}{2\pi C}$$

$$\therefore \; f^2 = \frac{1}{(2\pi)^2 LC}$$



Oscillators

- (A good program to write expressions is Mathcast).
- So for resonant frequency, we have: $$f = \frac{\sqrt{1}}{\sqrt{(2\pi)^2 LC}}$$
- Or: $$f_r = \frac{1}{2\pi\sqrt{LC}}$$
- You need to remember these.
- Where:
- L is the Inductance in Henries
- C is the Capacitance in Farads
- fr is the Output Frequency in Hertz.

Oscillators

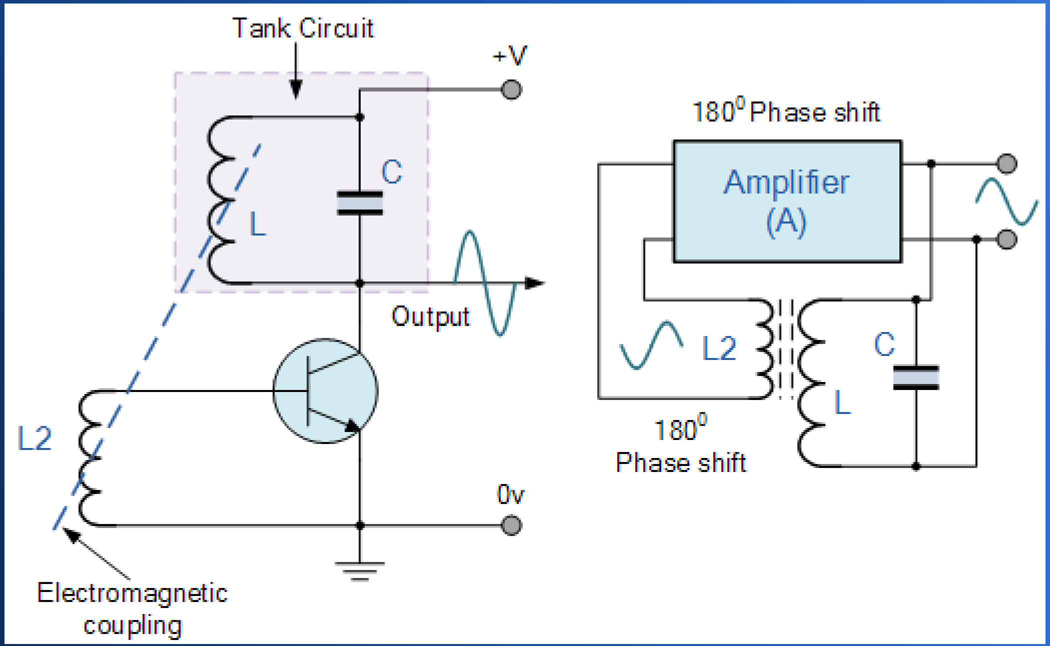
- This equation shows that if either L or C are decreased, the frequency increases. This output frequency is commonly given the abbreviation of (fr) to identify it as the “resonant frequency”.
-
- To keep the oscillations going in an LC tank circuit, we have to replace all the energy lost in each oscillation and also maintain the amplitude of these oscillations at a constant level. The amount of energy replaced must therefore be equal to the energy lost during each cycle.
- If the energy replaced is too large the amplitude would increase until clipping of the supply rails occurs. Alternatively, if the amount of energy replaced is too small the amplitude would eventually decrease to zero over time and the oscillations would stop.

Oscillators

- The simplest way of replacing this lost energy is to take part of the output from the LC tank circuit, amplify it and then feed it back into the LC circuit again. This process can be achieved using a voltage amplifier using an op-amp, FET or bipolar transistor as its active device. However, if the loop gain of the feedback amplifier is too small, the desired oscillation decays to zero and if it is too large, the waveform becomes distorted.
- To produce a constant oscillation, the level of the energy fed back to the LC network must be accurately controlled. Then there must be some form of automatic amplitude or gain control when the amplitude tries to vary from a reference voltage either up or down.
- To maintain a stable oscillation the overall gain of the circuit must be equal to one or unity. Any less and the oscillations will not start or die away to zero, any more the oscillations will occur but the amplitude will become clipped by the supply rails causing distortion. Consider the circuit next.

Oscillators

- A Bipolar Transistor is used as the LC oscillators amplifier with the tuned LC tank circuit acts as the collector load. Another coil L2 is connected between the base and the emitter of the transistor whose electromagnetic field is “mutually” coupled with that of coil L.



Oscillators

- “Mutual inductance” exists between the two circuits and the changing current flowing in one coil circuit induces, by electromagnetic induction, a potential voltage in the other (transformer effect) so as the oscillations occur in the tuned circuit, electromagnetic energy is transferred from coil L to coil L2 and a voltage of the same frequency as that in the tuned circuit is applied between the base and emitter of the transistor. In this way the necessary automatic feedback voltage is applied to the amplifying transistor.
- The amount of feedback can be increased or decreased by altering the coupling between the two coils L and L2. When the circuit is oscillating its impedance is resistive and the collector and base voltages are 180o out of phase. In order to maintain oscillations (called frequency stability) the voltage applied to the tuned circuit must be “in-phase” with the oscillations occurring in the tuned circuit.

Oscillators

- Therefore, we must introduce an additional 180o phase shift into the feedback path between the collector and the base. This is achieved by winding the coil of L2 in the correct direction relative to coil L giving us the correct amplitude and phase relationships for the Oscillators circuit or by connecting a phase shift network between the output and input of the amplifier.
- The LC Oscillator is therefore a “Sinusoidal Oscillator” or a “Harmonic Oscillator” as it is more commonly called. LC oscillators can generate high frequency sine waves for use in radio frequency (RF) type applications with the transistor amplifier being of a Bipolar Transistor or FET.
- Harmonic Oscillators come in many different forms because there are many different ways to construct an LC filter network and amplifier with the most common being the Hartley LC Oscillator, Colpitts LC Oscillator, Armstrong Oscillator and Clapp Oscillator to name a few.

Oscillators

- An example:
- An inductance of 200mH and a capacitor of 10pF are connected together in parallel to create an LC oscillator tank circuit. Calculate the frequency of oscillation....
- Then we can see from the below example that by decreasing the value of either the capacitance, C or the inductance, L will have the effect of increasing the frequency of oscillation of the LC tank circuit.

$$f = \frac{1}{2\pi\sqrt{LC}} = \frac{1}{2\pi\sqrt{200mH \times 10pF}} = 112.5\ kHz$$

Oscillators – to sumarise..

- The basic conditions required for an LC oscillator resonant tank circuit are given as follows.
- For oscillations to exist an oscillator circuit MUST contain a reactive (frequency-dependant) component either an “Inductor”, (L) or a “Capacitor”, (C) as well as a DC power source.
- In a simple inductor-capacitor, LC circuit, oscillations become damped over time due to component and circuit losses.
- Voltage amplification is required to overcome these circuit losses and provide positive gain.
- The overall gain of the amplifier must be greater than one, unity.
- Oscillations can be maintained by feeding back some of the output voltage to the tuned circuit that is of the correct amplitude and in-phase, (0o).
- Oscillations can only occur when the feedback is “Positive” (self-regeneration).
- The overall phase shift of the circuit must be zero or 360o so that the output signal from the feedback network will be “in-phase” with the input signal.

Filters

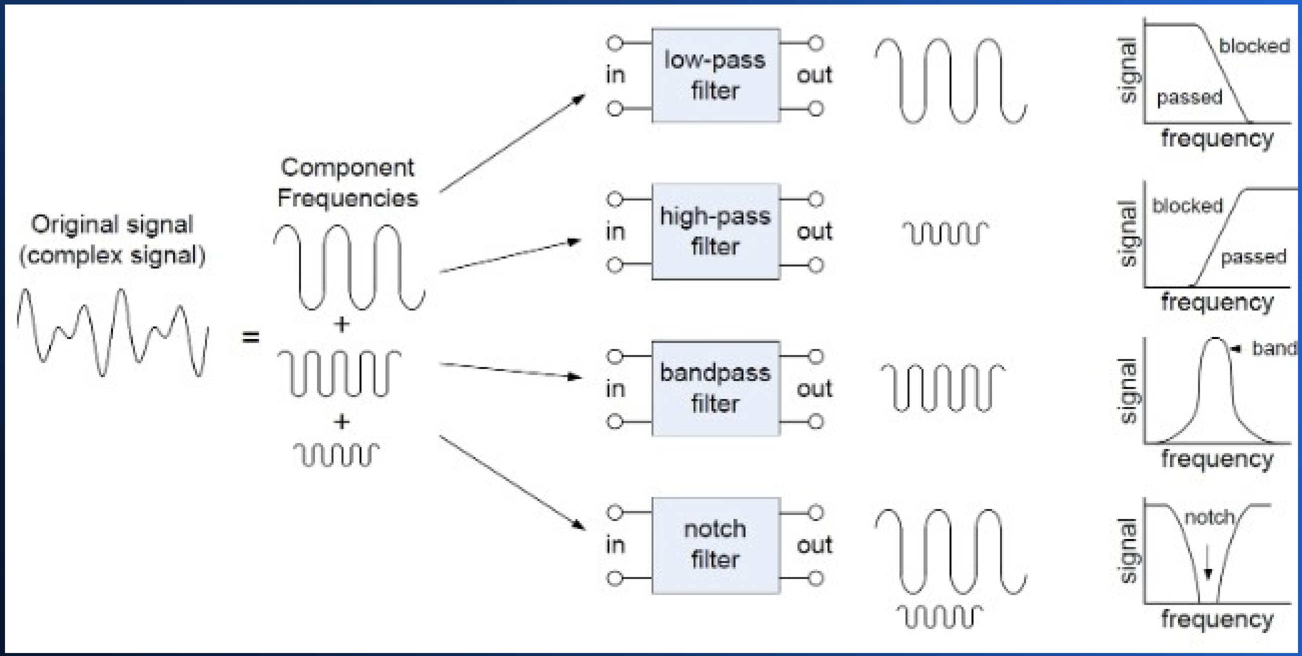
- What is an electronic filter?
- A filter is a circuit capable of passing (or amplifying) certain frequencies while attenuating other frequencies. Thus, a filter can extract important frequencies from signals that also contain undesirable or irrelevant frequencies. They are used extensively in audo circuits (think of your graphic equaliser on your sound system) and in radio circuits (to only transit or receive on the desired frequency).
- In the field of electronics, there are many practical applications for filters.
- There are four main types of filter. The four primary types of filters include the low-pass filter, the high-pass filter, the band-pass filter, and the notch filter (or the band-reject or band-stop filter).

Filters

- In the field of electronics, there are many practical applications for filters. Examples include:
- Radio communications: Filters enable radio receivers to only "see" the desired signal while rejecting all other signals (assuming that the other signals have different frequency content).
- DC power supplies: Filters are used to eliminate undesired high frequencies (i.e., noise) that are present on AC input lines. Additionally, filters are used on a power supply's output to reduce ripple.
- Audio electronics: A crossover network is a network of filters used to channel low-frequency audio to woofers, mid-range frequencies to midrange speakers, and high-frequency sounds to tweeters.
- Analogue-to-digital conversion: Filters are placed in front of an ADC input to minimise aliasing.

Filters

- This is a depiction of the four major filter types. A lowpass filter passes frequencies below a cutoff frequency, a highpass filter passes frequencies above a cutoff frequency, a bandpass filter passes a range of frequencies, and a bandstop or notch filter rejects a frequency range.



Filters

- Passive and Active Filters:
- Filters can be placed in one of two categories: passive or active.
- Passive filters include only passive components—resistors, capacitors, and inductors. In contrast, active filters use active components, such as op-amps, in addition to resistors and capacitors, but not inductors.
- Passive filters are most responsive to a frequency range from roughly 100 Hz to 300 MHz. The limitation on the lower end is a result of the fact that at low frequencies the inductance or capacitance would have to be quite large. The upper-frequency limit is due to the effect of parasitic capacitances and inductances. Careful design practices can extend the use of passive circuits well into the gigahertz range.

Filters

- Active filters are capable of dealing with very low frequencies (approaching 0 Hz), and they can provide voltage gain (passive filters cannot). Active filters can be used to design high-order filters without the use of inductors; this is important because inductors are problematic in the context of integrated-circuit manufacturing techniques. However, active filters are less suitable for very-high-frequency applications because of amplifier bandwidth limitations. Radio-frequency circuits must often utilize passive filters.
- Response curves are used to describe how a filter behaves. A response curve is simply a graph showing an attenuation ratio (VOUT / VIN) versus frequency (see Figure 2 below). Attenuation is commonly expressed in units of decibels (dB). Frequency can be expressed in two forms: either the angular form ω (units are rad/s) or the more common form of f (units of Hz, i.e., cycles per second). These two forms are related by $\omega = 2\pi f$. Finally, filter response curves may be plotted in linear-linear, log-linear, or log-log form. The most common approach is to have decibels on the y-axis and logarithmic frequency on the x-axis.

Filters

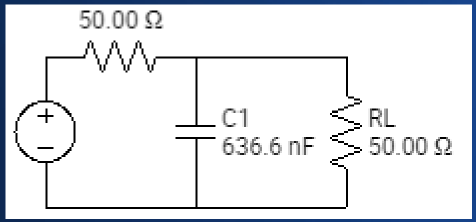
- Below are some technical terms that are commonly used when describing filter response curves:
- -3dB Frequency (f_{3dB}). This term, pronounced "minus 3dB frequency", corresponds to the input frequency that causes the output signal to drop by -3dB relative to the input signal. The -3dB frequency is also referred to as the cutoff frequency, and it is the frequency at which the output power is reduced by one-half (which is why this frequency is also called the "half-power frequency"), or at which the output voltage is the input voltage multiplied by $1/\sqrt{2}$. For low-pass and high-pass filters there is only one -3dB frequency. However, there are two -3dB frequencies for band-pass and notch filters—these are normally referred to as f_1 and f_2.
- Center frequency (f_0). The center frequency, a term used for band-pass and notch filters, is a central frequency that lies between the upper and lower cutoff frequencies. The center frequency is commonly defined as either the arithmetic mean (see equation below) or the geometric mean of the lower cutoff frequency and the upper cutoff frequency.

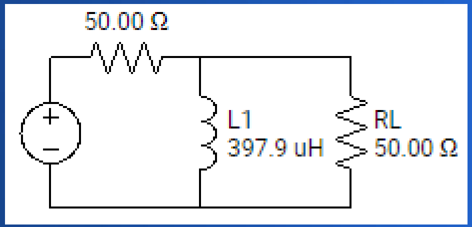
Filters

- For band-pass and notch filters, two stopband frequencies exist. The frequencies between these two stopband frequencies are referred to as the stopband.
- Quality factor (Q): The quality factor of a filter conveys its damping characteristics. In the time domain, damping corresponds to the amount of oscillation in the system's step response. In the frequency domain, higher Q corresponds to more (positive or negative) peaking in the system's magnitude response. For a bandpass or notch filter, Q represents the ratio between the center frequency and the -3dB bandwidth (i.e., the distance between f1 and f2).
- For both band-pass and notch filters:
- $Q = F_0 / (F_2 – F_1)$.

Filters

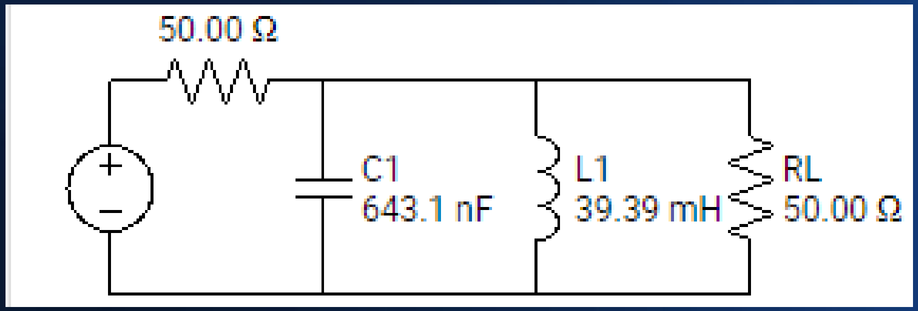
- We will use simple first order filters as an illistration. Cutoff freqiency is considered to be the frequency at which voltage is reduced by half (3dB) and power by a factor of four, (6dB).
- This is due to ohms law.
- Left: a lowpass filter, right, a highpass filter.
- Both are first order Butterworth filters with a cutoff frequency of 10kHz.

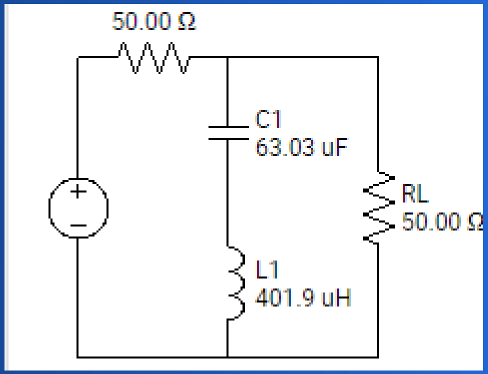




Filters

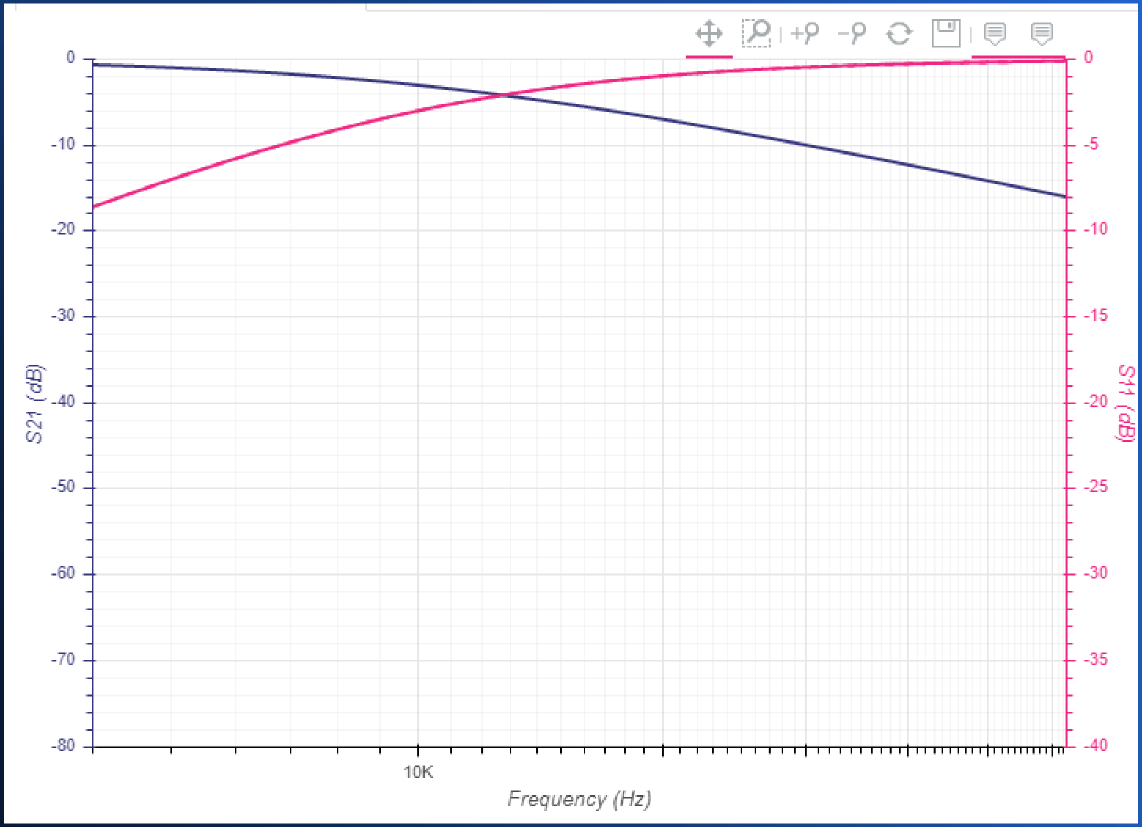
- Left: a first order butterworth lowpass filter, right, a first order butterworth highpass filter. Both filters have a lower cutoff frequency of 100Hz and an upper cutoff frequency of 10kHz.





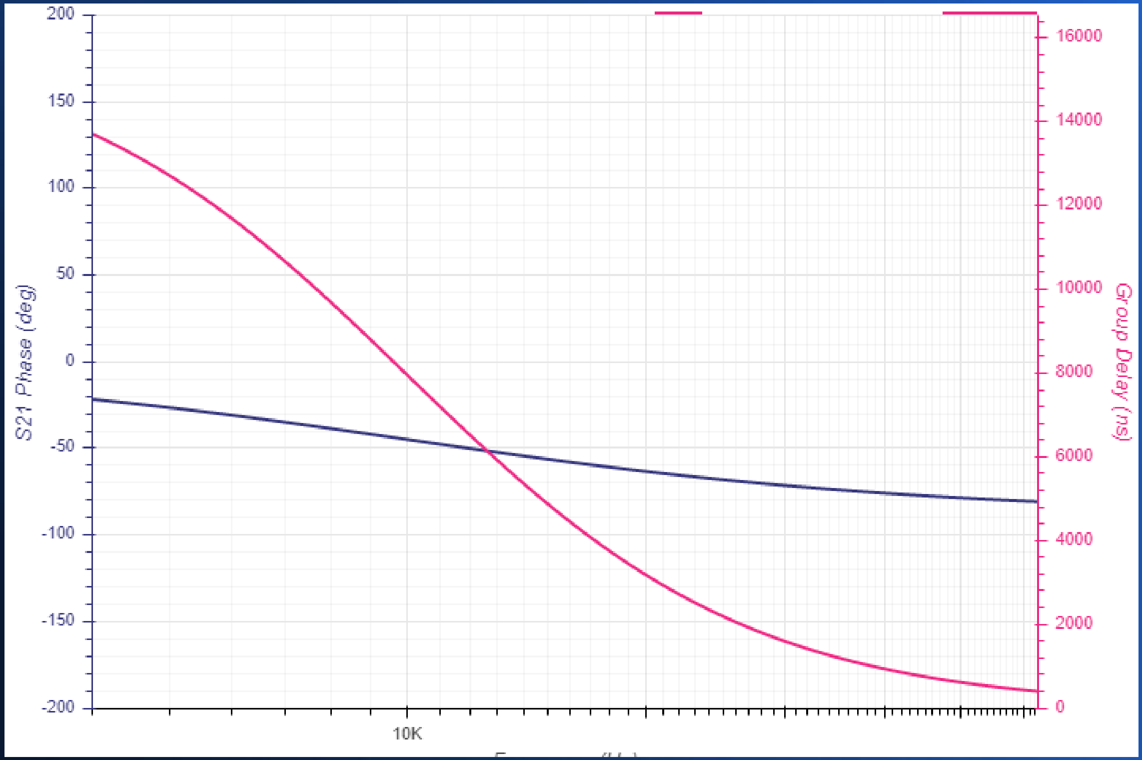
Filters

- First Order Butterworth lowpass gain.



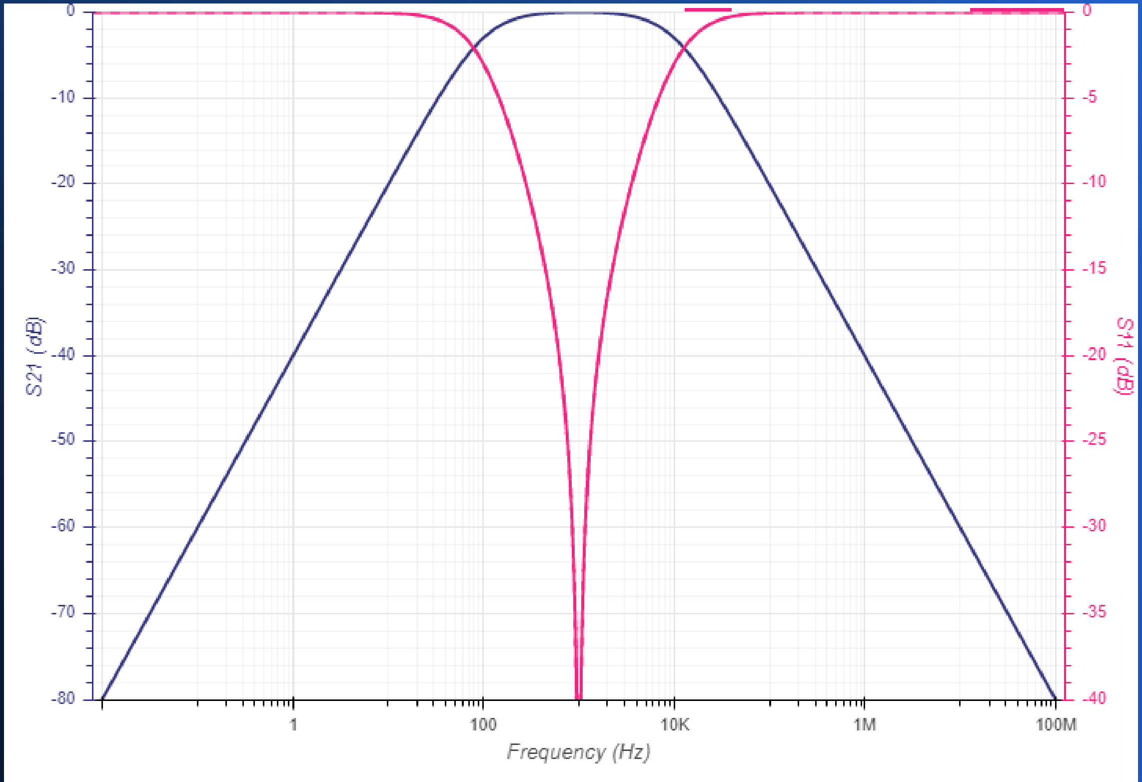
Filters

- First Order Butterworth lowpass phase angle.



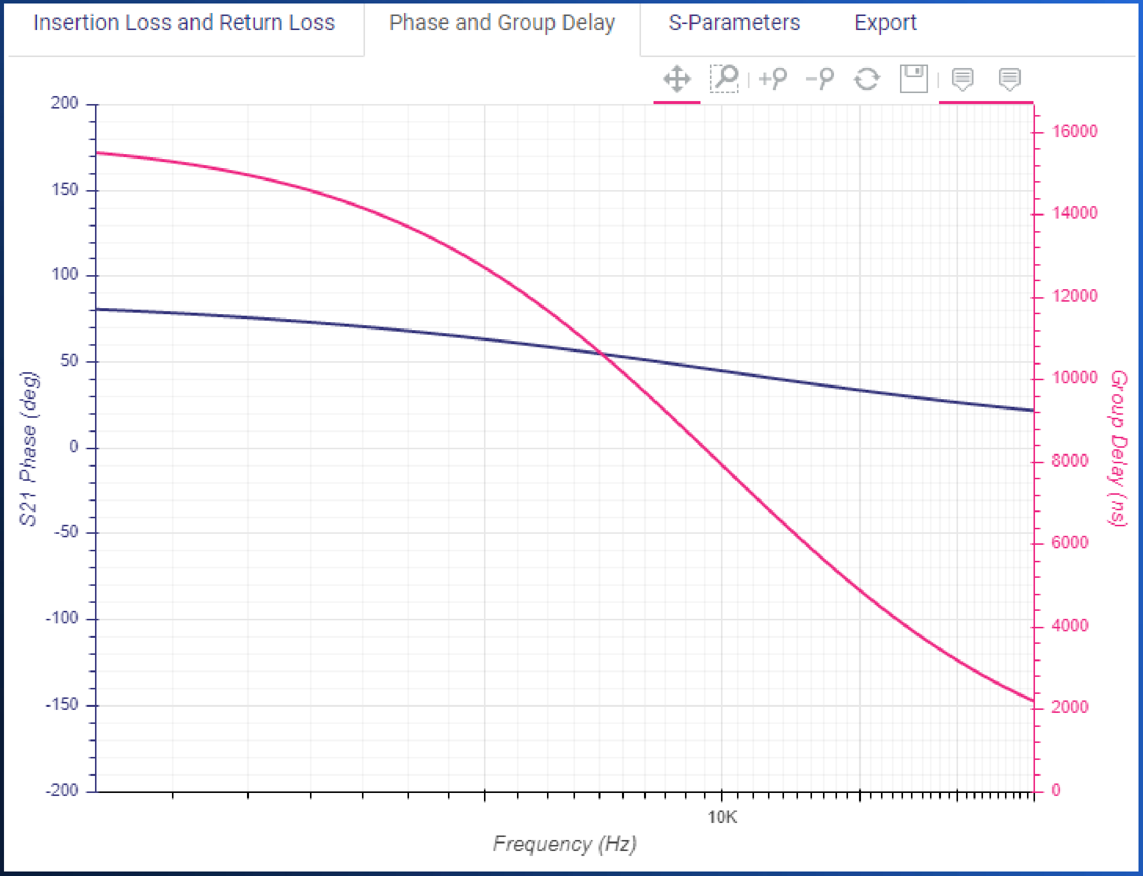
Filters

- First Order Butterworth highpass gain:



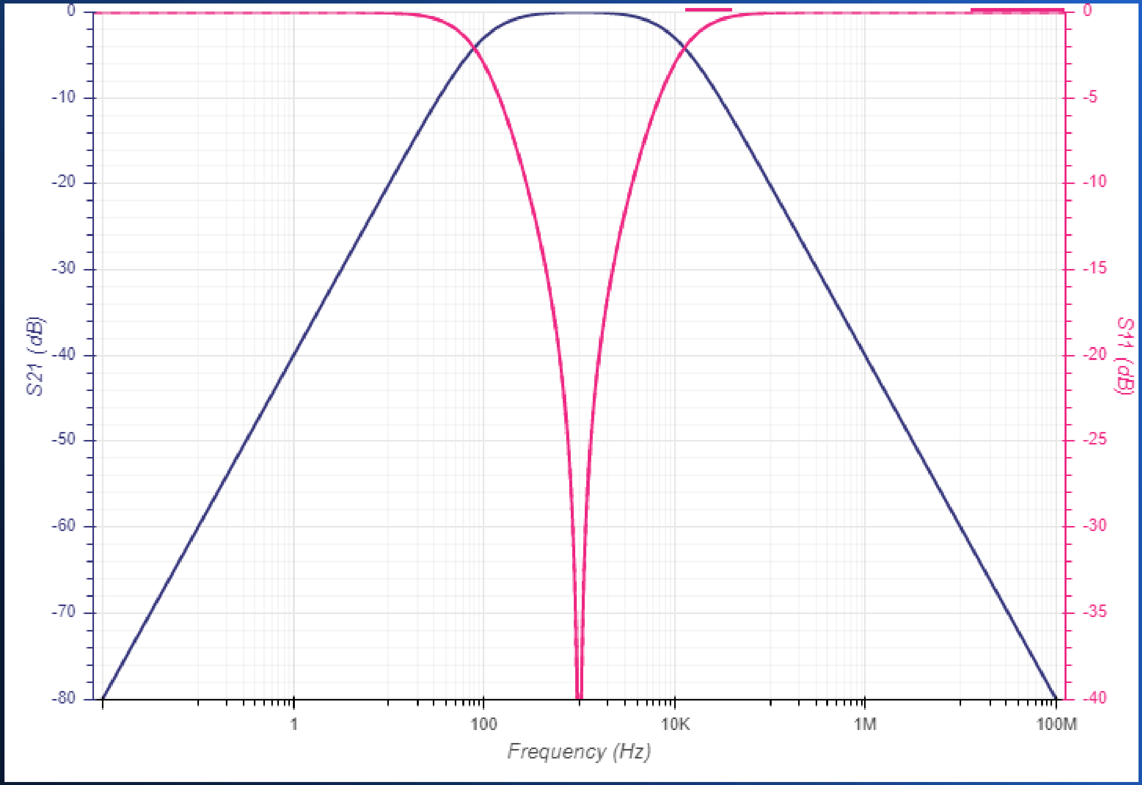
Filters

- First Order Butterworth highpass phase angle.



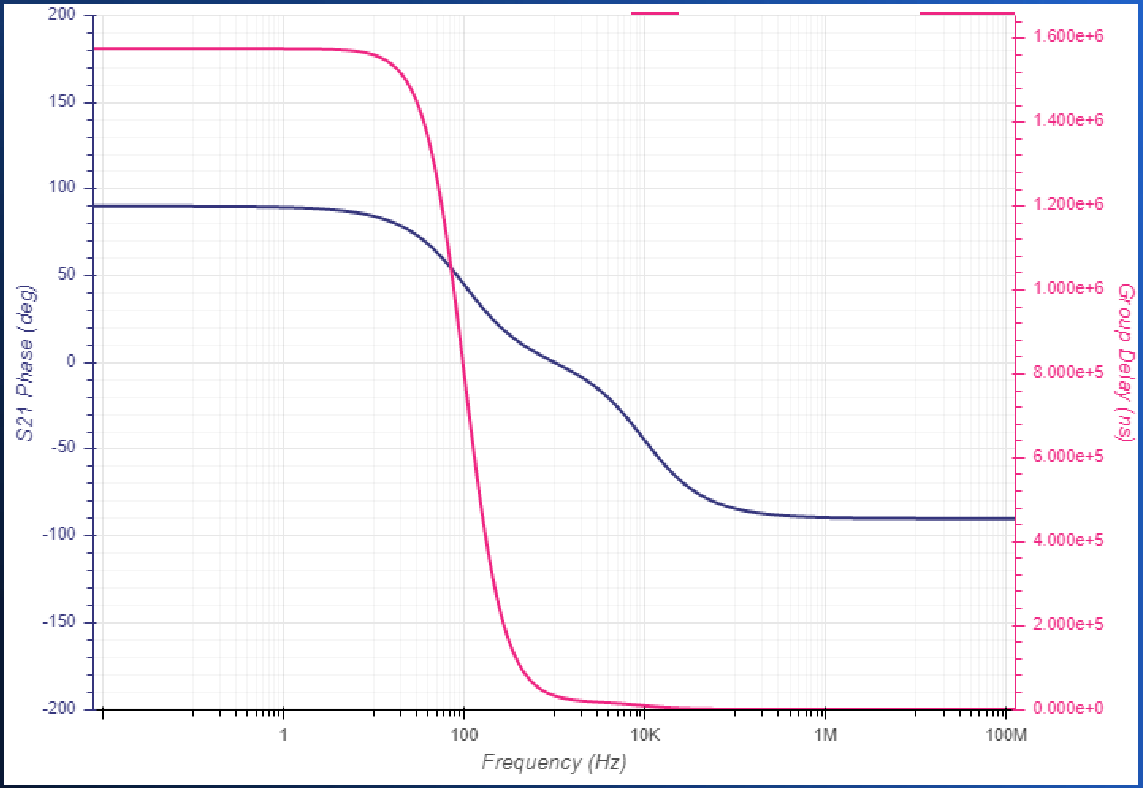
Filters

- First Order Butterworth bandpass gain.



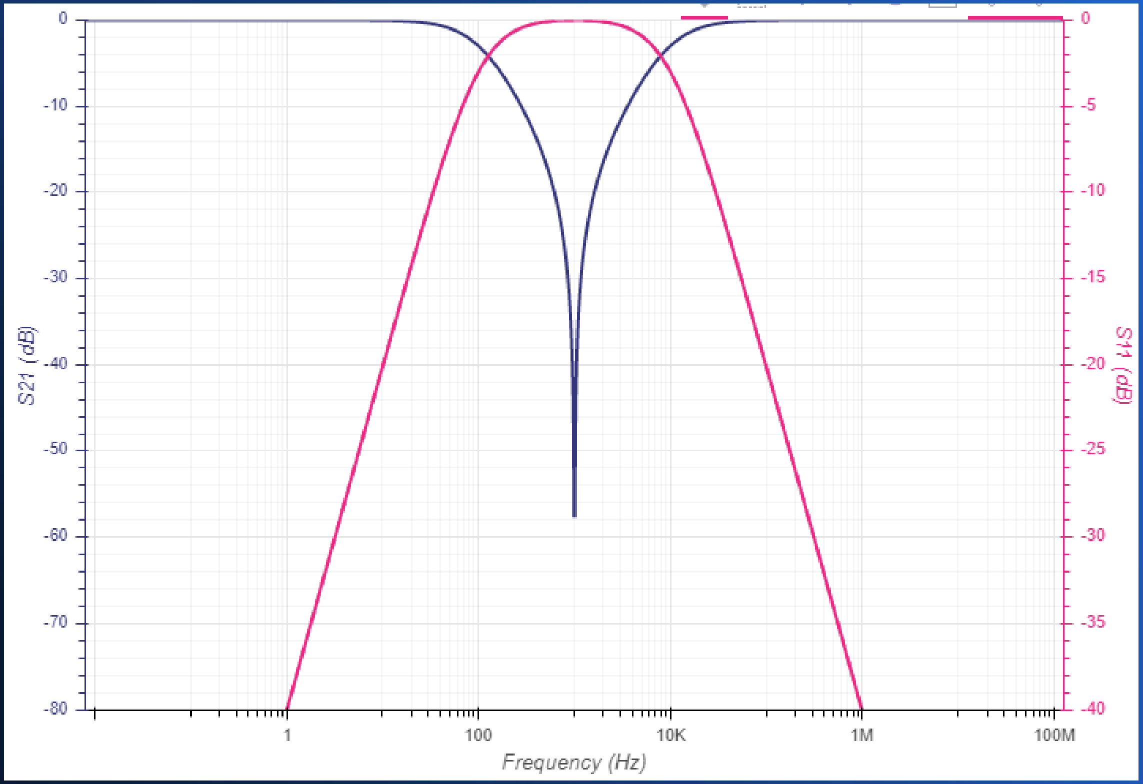
Filters

- First Order Butterworth bandpass phase angle.



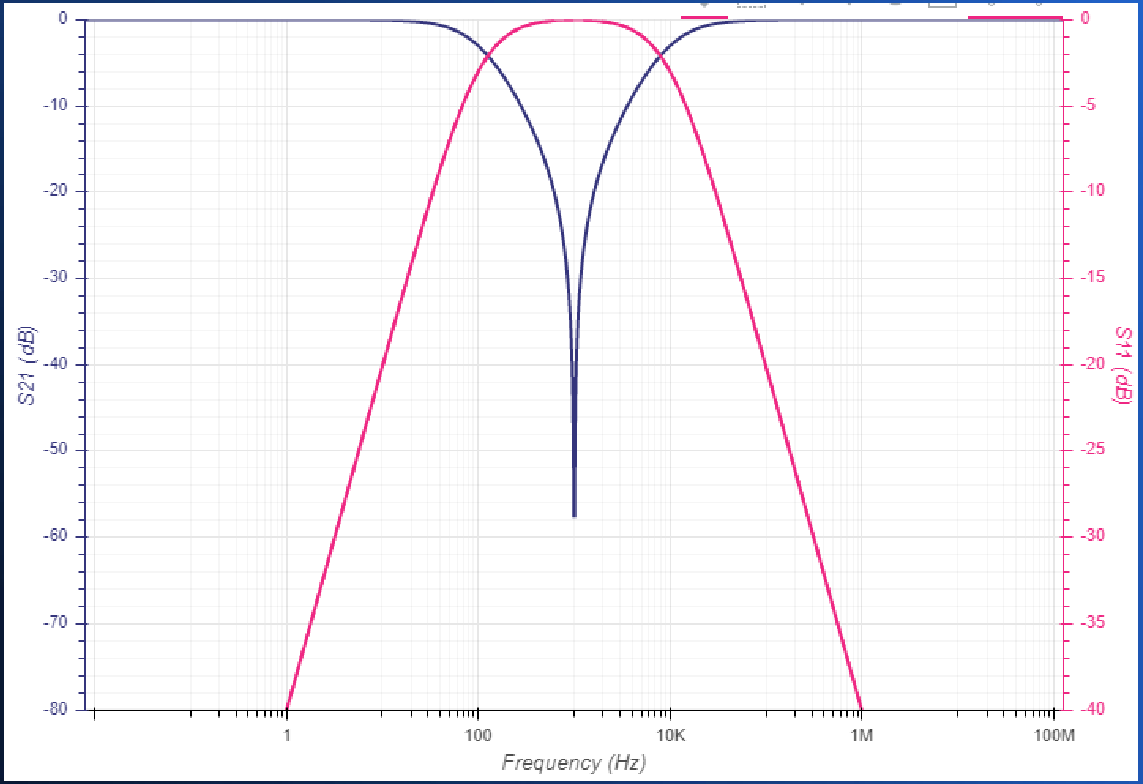
Filters

- First Order Butterworth bandstop gain.



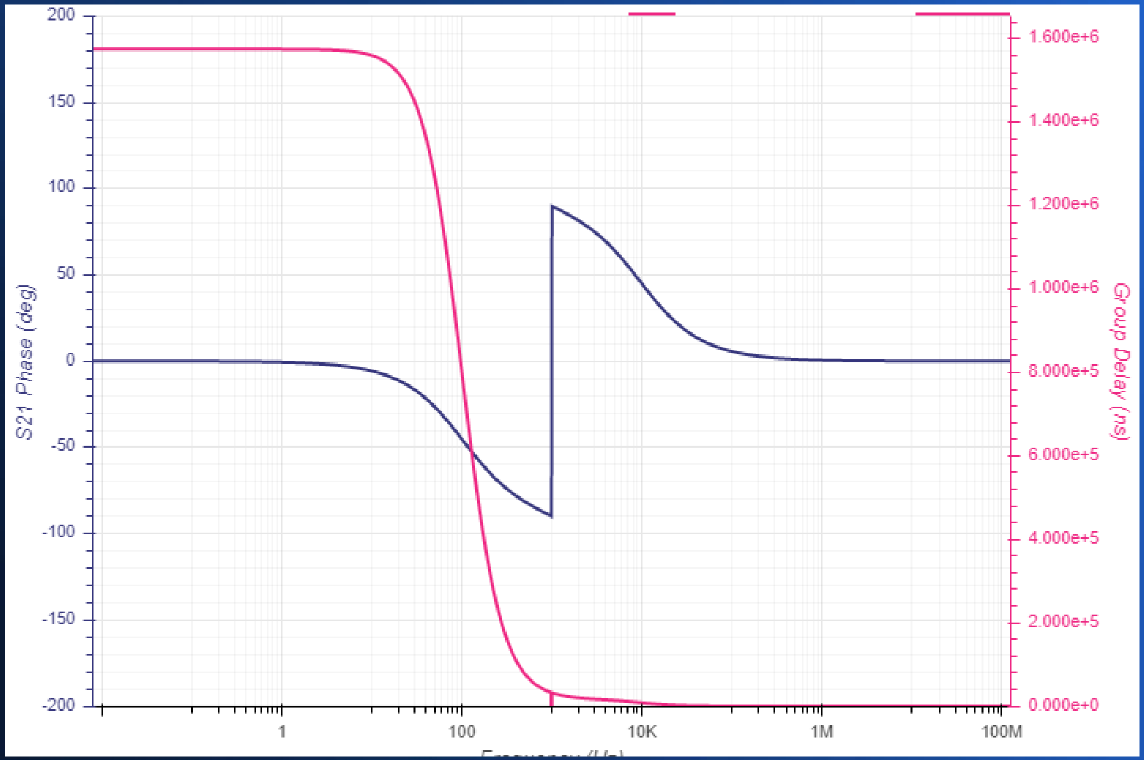
Filters

- First Order Butterworth bandstop gain.



Filters

- First Order Butterworth bandstop phase angle.



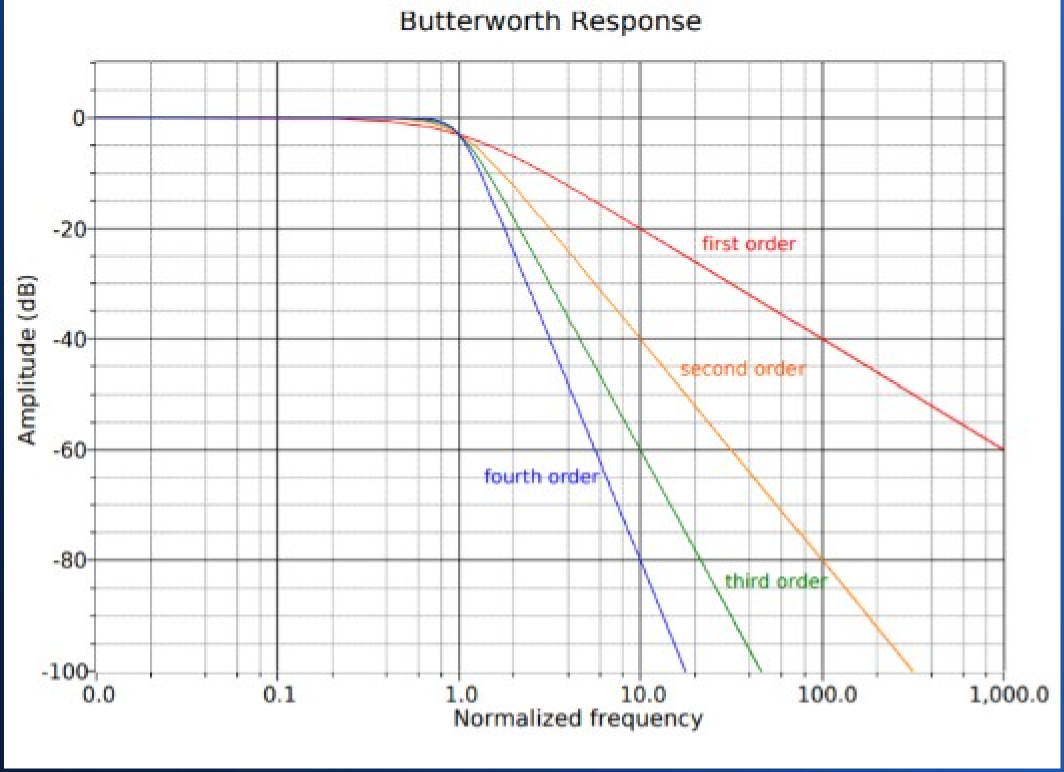
Filters

- Order of a filter:
- The order of a filter is given as an integer value and is derived from the filter's transfer function. As an example, all other factors being equal, a fourth-order filter will roll off twice as fast as a second-order filter, and four times faster than a first-order unit. See also the graph next...
- For example:

Filter Order	Voltage gain (dB)	Power gain (dB)
1	-3	-6
2	-6	-12
3	-9	-18
4	-12	-24
5	-15	-30

Filters

- Butterworth filter response for first, second, third and fourth orders.



Feedback Mechanisms

- <u>Positive and Negative feedback mechanisms...</u>
- is used in amplifiers and some types of fiter.
- Positive feedback:
- If the feedback signal is in the phase with the input signal, the effective input to the circuit is increased and this type of feedback is called positive, regenerative or direct feedback. It provides increased gain but it also increases distortion and leads to poor stability of gain. Positive feedback is used in oscillators where it is needed to sustain the oscillations in the output.
- Negative feedback in amplifier:
- In negative feedback, the feedback signal is opposite in phase to the input signal and this lead to a net decrease in the input signal resulting in a reduced gain. Negative feedback is also called degenerative or inverse feedback. It is used in amplifiers to improve the stability of gain, reduce distortion, transform input and output impedances and improve frequency response.

Feedback Mechanisms

- Advantages of negative feedback
- Despite reduction of gain, negative feedback is standard design practice in amplifiers because it improves the functioning of the amplifier in many ways. Its advantages are as follows:
- It improves the stability of gain.
- It increases the input impedance and reduces the output impedance.
- It increases the bandwidth of the amplifier.
- It reduces distortion and noise.

Feedback Mechanisms

- Types of feedback connections in an amplifier:
- Since both voltage and current can be fed back to the input using a series or parallel connection, there are four different types of feedback connections:
- A. Voltage series feedback.
- B. Current series feedback.
- C. Voltage shunt feedback.
- D. Current shunt feedback.
- Series feedback generally increases the input impedance whereas shunt feedback decreases the input impedance. Further, current feedback generally increases the output impedance whereas voltage feedback decreases the output impedance. Most multistage amplifiers require high input and low output impedances and voltage series feedback connection fulfils this requirement.

Feedback Mechanisms

- $$V_o = AV_i = A\left(V_s - V_f\right) \ldots\ldots$$
- $$A = \frac{V_o}{V_S}$$

feedback signal V_f is applied to the input , then

- $$V_i = V_s - V_f$$
- $$V_o = AV_i = A\left(V_s - V_f\right) \ldots\ldots$$
- $$A = \frac{V_o}{V_s}$$

$$A = \frac{V_o}{V_S}$$

Feedback Mechanisms

- In electronics, feedback is defined as the process of returning part of the signal output from a circuit or device back to the input of that circuit or device. Feedback systems are widely used in amplifier circuits, oscillators, process control systems, and in many other areas.
- Benefits of a feedback system include the ability to precisely control gain (e.g., amplification of a signal in an op amp), improve linear response, reduce signal distortion, and to control signal fluctuations. Feedback is sometimes referred to as a “closed loop” system. This means that an output signal is attached to the input of a device or system, forming a “loop.” The opposite of a closed loop or feedback loop is an “open loop” system where there is no feedback, and thus no corrective action based upon the output signal or what’s currently happening.

Feedback Mechanisms

- In control systems, the desired result is either to increase the input (positive or regenerative feedback) or to decrease the input (negative or degenerative feedback). In positive feedback control systems, the positive feedback is in phase with the input, which makes the output larger, i.e., results in system gain. An early use of positive feedback was regenerative circuits invented in 1914 for the amplification and reception of very weak radio signals. These circuits allowed radio signals to be amplified up to 100,000 times in one stage, although early circuits tended to be unstable and oscillated (remember, how do we make an oscillator!?).
- Today, positive feedback is primarily used in electronic oscillators to increase gain and narrow bandwidth. Positive feedback adds to the signal that needs correction, based on the output. One example is a radiator with a hot water valve and thermostat. If the temperature in a room gets too cold, the valve needs to increase the amount of hot water to the radiator to get the temperature back to a comfortable level. Here, the relationship of the input signal to the output signal (to the valve) is inverse (negative feedback), and so if the valve is naturally closed, or normally closed, increasing the amplitude of the signal to the valve provides more energy to further open the hot water valve and heat the room. The thermostat is the feedback mechanism and the temperature in the room is the feedback. In an open loop system, the valve would be connected to nothing – and the heating would run at full power forever!

Feedback Mechanisms

- In negative feedback systems, the negative feedback is out of phase with the input, which reduces the signal, making the output smaller. If we apply the hot water radiator example here, and this time the valve is normally open in its de-energized state, then less voltage applied to the valve would cause the valve to open more. As an aside to control theory, a normally open hot water valve would mean that when the temperature is satisfied, the valve would have maximum voltage applied so that the radiator would no longer heat the room.
- Negative feedback is the most common form of feedback control used in all manner of systems. Electronic devices are inherently non-linear but can be made more linear with the use of negative feedback. The negative feedback amplifier was invented in 1927 by Harold Black while working at Bell Labs. Bell used his invention to reduce the overcrowding of phone lines and to extend its long-distance network. Negative feedback was later used by the military to design accurate fire-control systems in World War II. Later uses included operational amplifiers (op amps) and precise variable-frequency audio oscillators.

Feedback Mechanisms

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Feedback Mechanisms

- Op amps may or may not include an external feedback loop, depending on what function it’s configured to do, but if external feedback is not available it is likely configured at a set value in the integrated chip. External feedback allows the designer to adjust it's gain.
- Consider a voltage series negative feedback amplifier. If there is no feedback, then the voltage gain is called internal gain and is given by:
- $A = V_0 / V_s$.

If the feedback signal Vf is applied to the input , then $V_i = V_s – V_f$.

So $V0 = AV_i = A(V_s - V_f)$...

$V_f = \beta V_0$ which is the feedback ratio.

Feedback Mechanisms

- For negative closed loop feedback the voltage gain Vo = $A_f = \frac{V_o}{V_s} = \frac{A}{1 + A\beta}$
- For positive closed loop feedback the voltage gain Vo = $A_f = \frac{V_o}{V_s} = \frac{A}{1 - A\beta}$
- This shows that positive feedback increases the voltage gain.
- The factor (-Aβ) is called the loop gain or loop transmission or return ratio or feedback factor.
- The difference between unity and feedback factor (1+Aβ) is called return difference.

Introduction to Analogue Communication Systems

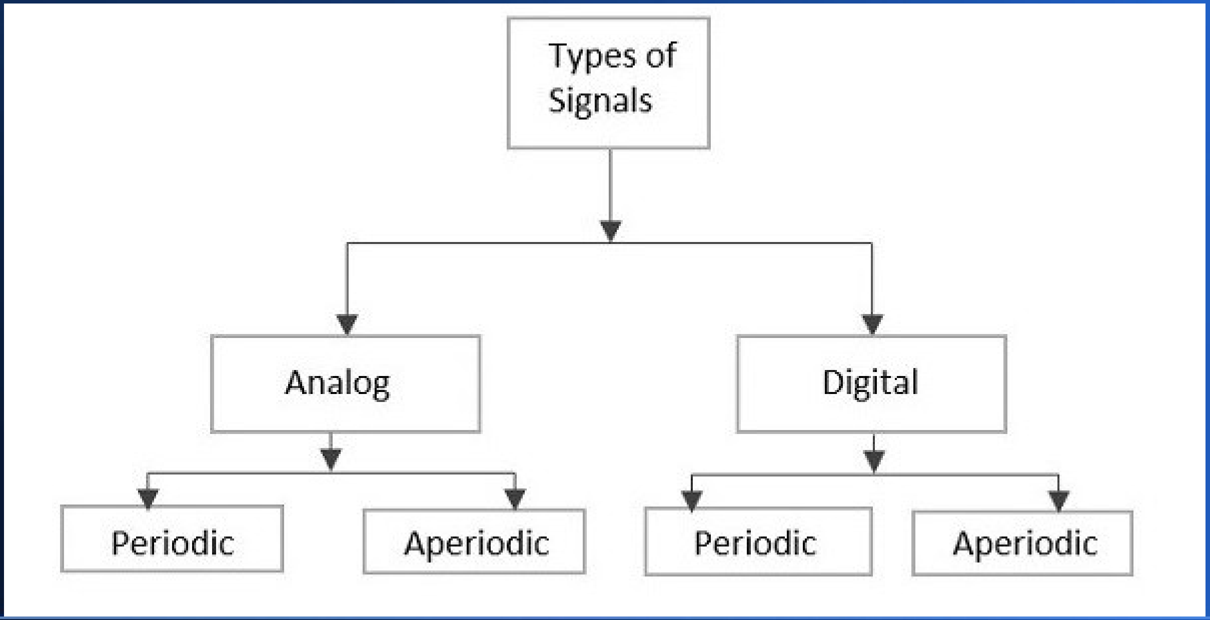
- The word communication arises from the Latin word commūnicāre, which means “to share”. Communication is the basic step for exchange of information.
-
- For example, a baby in a cradle, communicates with a cry when she needs her mother. A person communicates with the help of a language. Communication is the means to share.
- Communication can be defined as the process of exchange of information through means such as words, actions, signs, etc., between two or more individuals.

Introduction to Analogue Communication Systems

- Parts of a Communication System:
- Sender is the person who sends a message. It could be a transmitting station from where the signal is transmitted.
- Channel is the medium through which the message signals travel to reach the destination.
- Receiver is the person who receives the message. It could be a receiving station where the transmitted signal is being received.
- Types of Signals:
- Conveying an information by some means such as gestures, sounds, actions, etc., can be termed as signaling. Hence, a signal can be a source of energy which transmits some information. This signal helps to establish a communication between the sender and the receiver.

Introduction to Analogue Communication Systems

- Analogue Signal
- A continuous time varying signal, which represents a time varying quantity can be termed as an Analog Signal. This signal keeps on varying with respect to time, according to the instantaneous values of the quantity, which represents it.



Introduction to Analogue Communication Systems

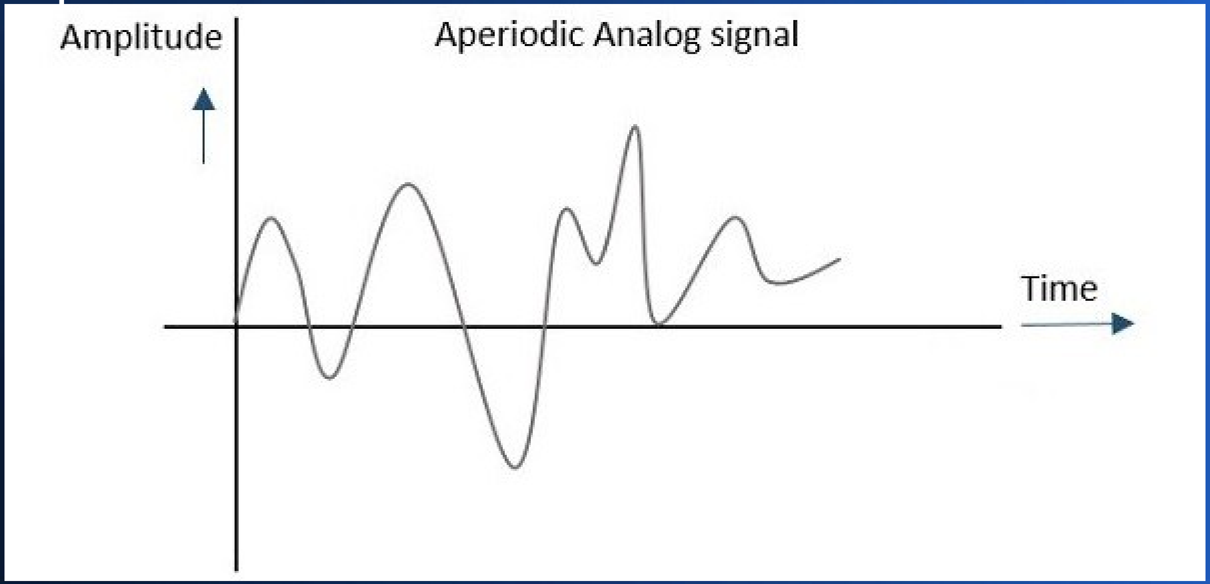
- Digital Signal
- A signal which is discrete in nature or which is non-continuous in form can be termed as a Digital signal. This signal has individual values, denoted separately, which are not based on the previous values, as if they are derived at that particular instant of time.
- Anglouge signals include speech, music, voltage, current, sound and ight intensity.
- Digital signals consist of discrete states. For example, TTL (transistor transistor logic) uses 0V to represent binary 0, and 5V to represent binary 1. This is because it is very difficult to design a circuit which recognises more than two 'states'.

Introduction to Analogue Communication Systems

- These values can be considered individually and separately or discretely, hence they are called as discrete values.
- The binary digits which has only 1s and 0s are mostly termed as digital values. Hence, the signals which represent 1s and 0s are also called as digital signals. The communication based on digital signals and digital values is called as Digital Communication.
- Periodic Signal
- Any analog or digital signal, that repeats its pattern over a period of time, is called as a Periodic Signal. This signal has its pattern continued repeatedly and is easy to be assumed or to be calculated.

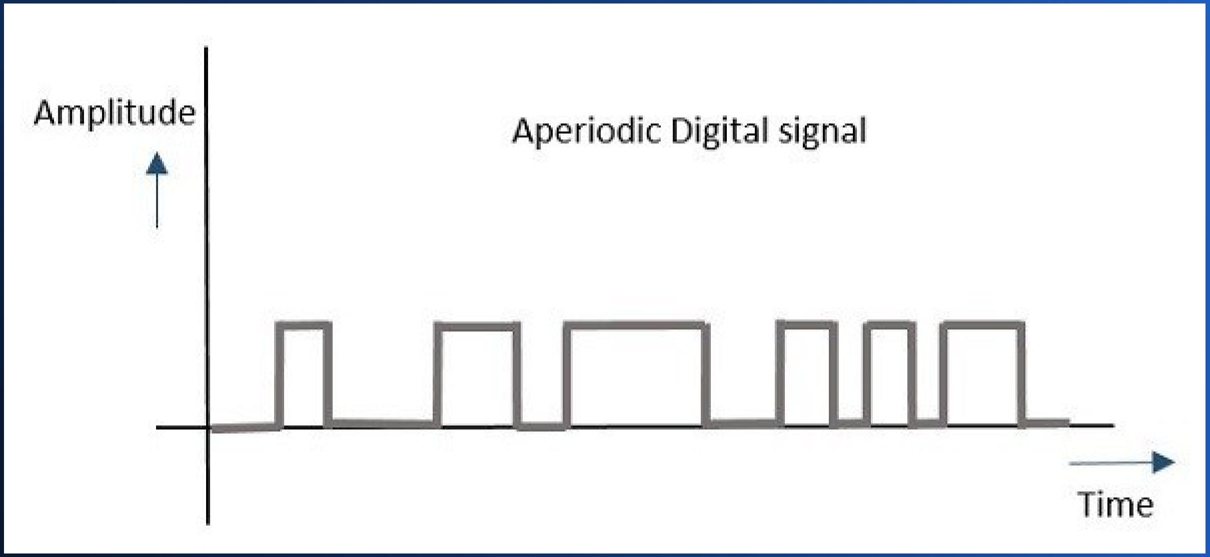
Introduction to Analogue Communication Systems

- Aperiodic Signal
- Any analog or digital signal, that doesn't repeat its pattern over a period of time is called as Aperiodic Signal. This signal has its pattern continued but the pattern is not repeated. It is also not so easy to be assumed or to be calculated. Such a process whether considered analog or digital, can be graphically represented as follows.



Introduction to Analogue Communication Systems

- In general, the signals which are used in communication systems are analog in nature, which are transmitted in analog or converted to digital and then transmitted, depending upon the requirement.
-



Introduction to Analogue Communication Systems

- Modulation:
- For a signal to be transmitted to a distance, without the effect of any external interferences or noise addition and without getting faded away, it has to undergo a process called as Modulation. It improves the strength of the signal without disturbing the parameters of the original signal.
- What is Modulation?
- A message carrying a signal has to get transmitted over a distance and for it to establish a reliable communication, it needs to take the help of a high frequency signal which should not affect the original characteristics of the message signal.

Introduction to Analogue Communication Systems

- The characteristics of the message signal, if changed, the message contained in it also alters. Hence, it is a must to take care of the message signal. A high frequency signal can travel up to a longer distance, without getting affected by external disturbances. We take the help of such high frequency signal which is called as a carrier signal to transmit our message signal. Such a process is simply called as Modulation.
- Modulation is the process of changing the parameters of the carrier signal, in accordance with the instantaneous values of the modulating signal.
- Need for Modulation:
- Baseband signals are incompatible for direct transmission. For such a signal, to travel longer distances, its strength has to be increased by modulating with a high frequency carrier wave, which doesn’t affect the parameters of the modulating signal.

Introduction to Analogue Communication Systems

- Advantages of Modulation
- The antenna used for transmission, had to be very large, if modulation was not introduced. The range of communication gets limited as the wave cannot travel a distance without getting distorted.
- Following are some of the advantages for implementing modulation in the communication systems.
- Reduction of antenna size
- No signal mixing
- Increased communication range
- Multiplexing of signals
- Possibility of bandwidth adjustments
- Improved reception quality

Introduction to Analogue Communication Systems

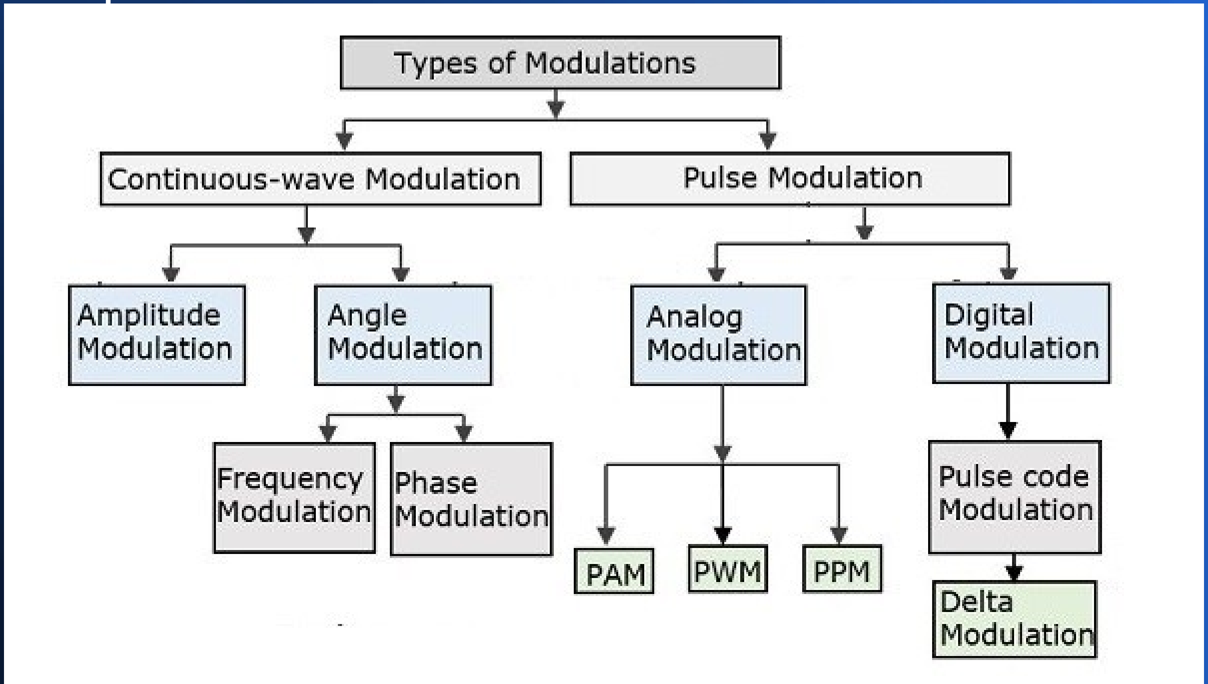
- Signals in the Modulation Process
- Following are the three types of signals in the modulation process.
- Message or Modulating Signal:
- The signal which contains a message to be transmitted, is called as a message signal. It is a baseband signal, which has to undergo the process of modulation, to get transmitted. Hence, it is also called as the modulating signal.
- Carrier Signal:
- The high frequency signal, which has a certain amplitude, frequency and phase but contains no information is called as a carrier signal. It is an empty signal and is used to carry the signal to the receiver after modulation.

Introduction to Analogue Communication Systems

- Modulated Signal
- The resultant signal after the process of modulation is called as a modulated signal. This signal is a combination of modulating signal and carrier signal.
- Types of Modulation
- There are many types of modulations. Depending upon the modulation techniques used, they are classified as shown in the following slide.....

Introduction to Analogue Communication Systems

- As a licenced radio amateur (ham) I am well aware of these!
- The types of modulations are broadly classified into continuous-wave modulation and pulse modulation.



- Continuous-wave Modulation
- In continuous-wave modulation, a high frequency sine wave is used as a carrier wave. This is further divided into amplitude and angle modulation.
- If the amplitude of the high frequency carrier wave is varied in accordance with the instantaneous amplitude of the modulating signal, then such a technique is called as Amplitude Modulation.
- If the angle of the carrier wave is varied, in accordance with the instantaneous value of the modulating signal, then such a technique is called as Angle Modulation. Angle modulation is further divided into frequency modulation and phase modulation.
- If the frequency of the carrier wave is varied, in accordance with the instantaneous value of the modulating signal, then such a technique is called as Frequency Modulation.
- If the phase of the high frequency carrier wave is varied in accordance with the instantaneous value of the modulating signal, then such a technique is called as Phase Modulation.

Introduction to Analogue Communication Systems

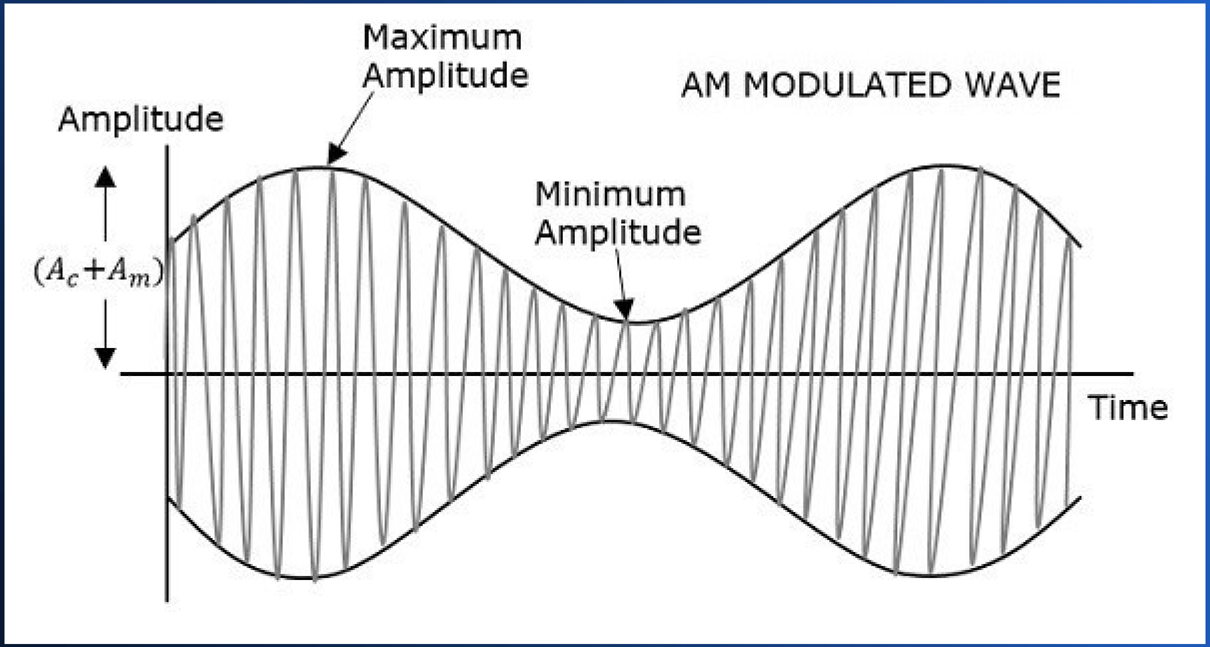
- Pulse Modulation
- In Pulse modulation, a periodic sequence of rectangular pulses, is used as a carrier wave. This is further divided into analog and digital modulation.
- In analog modulation technique, if the amplitude or duration or position of a pulse is varied in accordance with the instantaneous values of the baseband modulating signal, then such a technique is called as Pulse Amplitude Modulation (PAM) or Pulse Duration/Width Modulation (PDM/PWM), or Pulse Position Modulation (PPM).
- In digital modulation, the modulation technique used is Pulse Code Modulation (PCM) where the analog signal is converted into digital form of 1s and 0s. As the resultant is a coded pulse train, this is called as PCM. This is further developed as Delta Modulation (DM).

Introduction to Analogue Communication Systems

- A continuous-wave goes on continuously without any intervals and it is the baseband message signal, which contains the information. This wave has to be modulated.
- According to the standard definition, “The amplitude of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal.” Which means, the amplitude of the carrier signal containing no information varies as per the amplitude of the signal containing information, at each instant. This can be well explained by the following...

Introduction to Analogue Communication Systems

- The low frequency is the modulating signal. Often audio. The high frequency is the carrier. For example, our ears are sensitive to 20Hz to 20kHz (when young; the ability to hear the higher frequencies decreases with age).



Introduction to Analogue Communication Systems

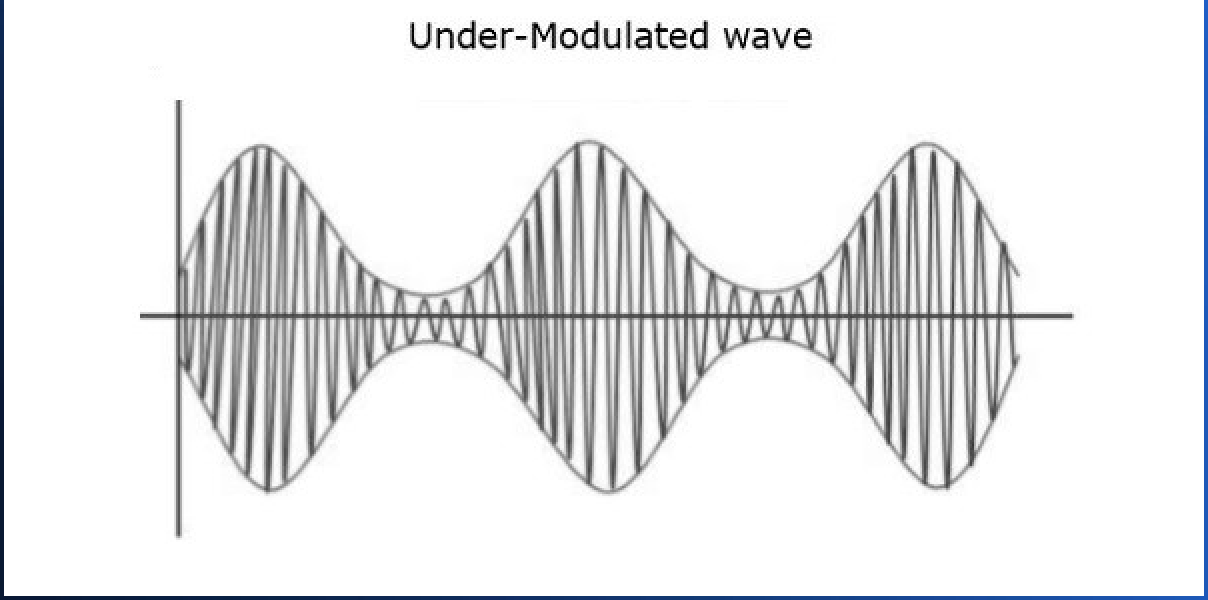
- It can be observed that the positive and negative peaks of the carrier wave, are interconnected with an imaginary line. This line helps recreating the exact shape of the modulating signal. This imaginary line on the carrier wave is called as Envelope. It is the same as that of the message signal.
- Mathematical Expressions:
- Following are the mathematical expressions for these waves.
- Time-domain Representation of the Waves
- Let the modulating signal be $m(t)=A_m\cos(2\pi f_m t)$
- and the carrier signal: $c(t)=A_c\cos(2\pi f_c t)$
- Where, Am and Ac are the amplitude of the modulating signal and the carrier signal respectively. f_m and f_c are the frequency of the modulating signal and the carrier signal respectively.

Introduction to Analogue Communication Systems

- Then, the equation of Amplitude Modulated wave will be:
- $s(t)=[Ac+A_m \cos(2\pi f_m t)]\cos(2\pi f_c t)$.
- Modulation Index
- A carrier wave, after being modulated, if the modulated level is calculated, then such an attempt is called as Modulation Index or Modulation Depth, μ. It states the level of modulation that a carrier wave undergoes.
- $\Rightarrow \mu=(Amax-A_{min})/(Amax+A_{min})$.
- the two formulas for Modulation index. The modulation index or modulation depth is often denoted in percentage called as Percentage of Modulation. We will get the percentage of modulation, just by multiplying the modulation index value with 100. For a perfect modulation, the value of modulation index should be 1, which implies the percentage of modulation should be 100%.

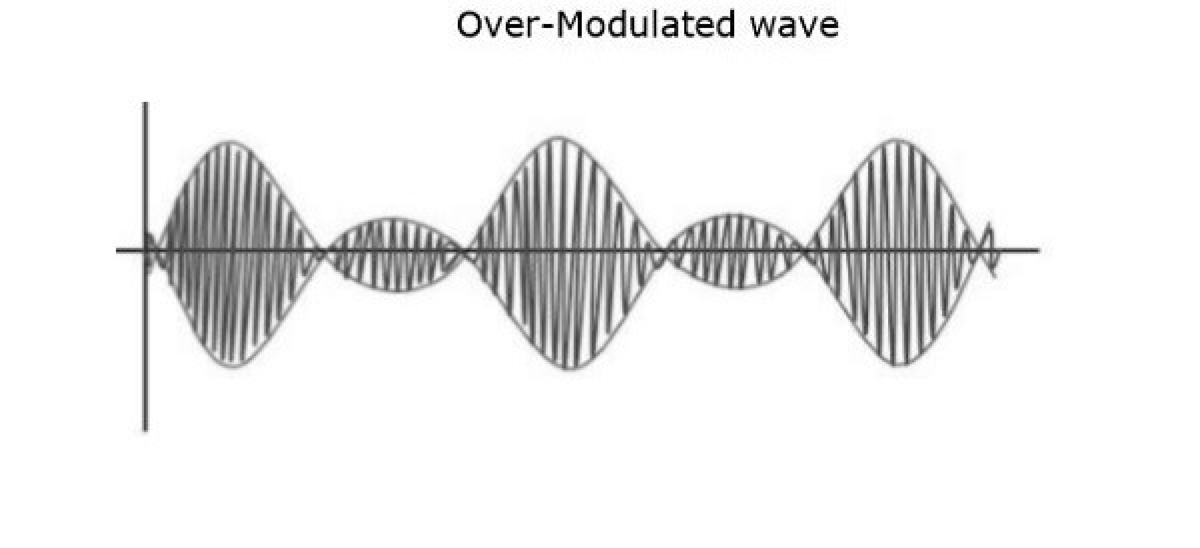
Introduction to Analogue Communication Systems

- For instance, if this value is less than 1, i.e., the modulation index is 0.5, then the modulated output would look like the following figure. It is called as Under-modulation. Such a wave is called as an under-modulated wave.
- For AM, too little modulation makes a signal appear weak; too much will cause interferance to other signals.



Introduction to Analogue Communication Systems

- If the value of the modulation index is greater than 1, i.e., 1.5 or so, then the wave will be an over-modulated wave. It would look like the following figure.
- As the value of the modulation index increases, the carrier experiences a 180o phase reversal, which causes additional sidebands and hence, the wave gets distorted. Such an over-modulated wave causes interference, which cannot be eliminated.



Introduction to Analogue Communication Systems

- Bandwidth of AM Wave
- Bandwidth (BW) is the difference between the highest and lowest frequencies of the signal. Mathematically, we can write it as:
- $BW = f_{max} - f_{min} \Rightarrow BW = 2f_m$
- Power of AM wave is equal to the sum of powers of carrier, upper sideband, and lower sideband frequency components.
- For carrier power: $$P_c = \frac{(A_c/\sqrt{2})^2}{R} = \frac{A_c^2}{2R}$$
- For upper sideband power: $$P_{USB} = \frac{(A_c\mu/2\sqrt{2})^2}{R} = \frac{A_c^2\mu^2}{8R}$$
- For lower sideband power: $$P_{LSB} = \frac{A_c^2\mu^2}{8R}$$

Introduction to Analogue Communication Systems

- Now, let us add these three powers in order to get the power of AM wave.
- We can use the above formula to calculate the power of AM wave, when the carrier power and the modulation index are known.

$$P_t = \frac{A_c^2}{2R} + \frac{A_c^2 \mu^2}{8R} + \frac{A_c^2 \mu^2}{8R}$$

$$\Rightarrow P_t = \left(\frac{A_c^2}{2R}\right)\left(1 + \frac{\mu^2}{4} + \frac{\mu^2}{4}\right)$$

$$\Rightarrow P_t = P_c\left(1 + \frac{\mu^2}{2}\right)$$

If the modulation index μ=1;

then the power of AM wave is equal to 1.5 times

the carrier power. So, the power required for

transmitting an AM wave is 1.5 times the carrier

power for a perfect modulation.

Introduction to Analogue Communication Systems

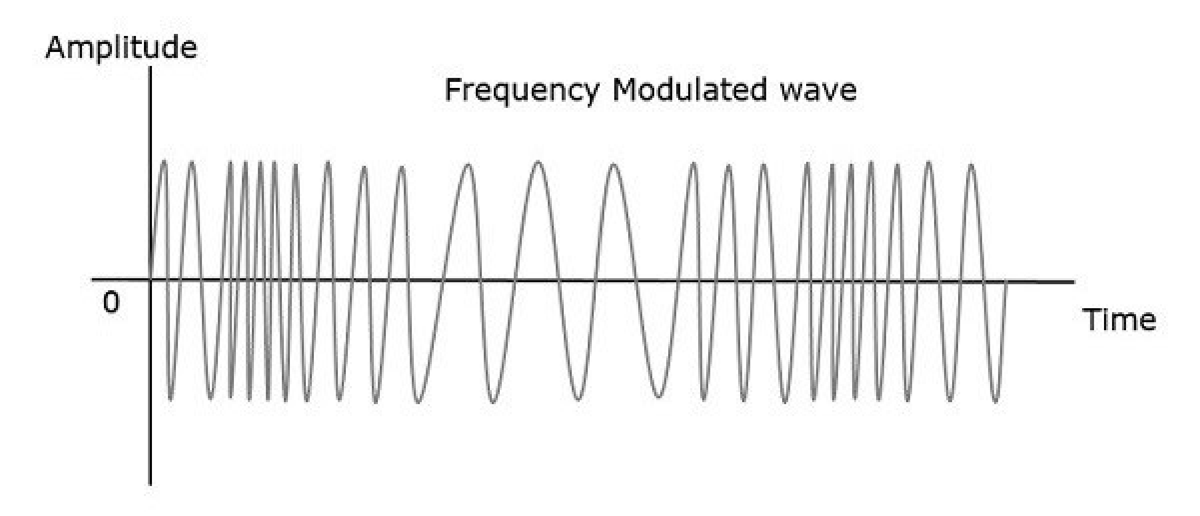
- Frequency Modulation:
- The other type of modulation in continuous-wave modulation is Angle Modulation. Angle Modulation is the process in which the frequency or the phase of the carrier signal varies according to the message signal.
- The standard equation of the angle modulated wave is $s(t)=A_c\cos\theta_i(t)$.
- Where, Ac is the amplitude of the modulated wave, which is the same as the amplitude of the carrier signal;
- $\theta_i(t)$ is the angle of the modulated wave.
- Angle modulation is further divided into frequency modulation and phase modulation.
- Angle modulation is further divided into frequency modulation and phase modulation.

Introduction to Analogue Communication Systems

- Frequency Modulation is the process of varying the frequency of the carrier signal linearly with the message signal.
- Phase Modulation is the process of varying the phase of the carrier signal linearly with the message signal.
- Frequency Modulation
- In amplitude modulation, the amplitude of the carrier signal varies. Whereas, in Frequency Modulation (FM), the frequency of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal.
- Hence, in frequency modulation, the amplitude and the phase of the carrier signal remains constant. This can be better understood by observing the following slide//.

Introduction to Analogue Communication Systems

- The frequency of the modulated wave increases, when the amplitude of the modulating or message signal increases. Similarly, the frequency of the modulated wave decreases, when the amplitude of the modulating signal decreases. Note that, the frequency of the modulated wave remains constant and it is equal to the frequency of the carrier signal, when the amplitude of the modulating signal is zero.



Introduction to Analogue Communication Systems

- Mathematical Representation:
- The equation for instantaneous frequency in FM modulation is:
- $fi=f_c+kfm(t)$
- Where, f_c is the carrier frequency, k_t is the frequency sensitivity;
- m(t) is the message signal.
- We know the relationship between angular frequency ω_i
- and angle $\theta_i(t)$ as $\omega i=d\theta_i(t)/dt$.
- The difference between FM modulated frequency (instantaneous frequency) and normal carrier frequency is termed as Frequency Deviation. It is denoted by Δf , which is equal to the product of k_f and A_m.

Introduction to Analogue Communication Systems

- FM can be divided into Narrowband FM and Wideband FM based on the values of modulation index β.
- Narrowband FM
- Following are the features of Narrowband FM.
- This frequency modulation has a small bandwidth when compared to wideband FM.
- The modulation index β is small, i.e., less than 1.
- Its spectrum consists of the carrier, the upper sideband and the lower sideband.
- This is used in mobile communications such as police wireless, ambulances, taxicabs, etc which do not need to deliver high quality audio.

Introduction to Analogue Communication Systems

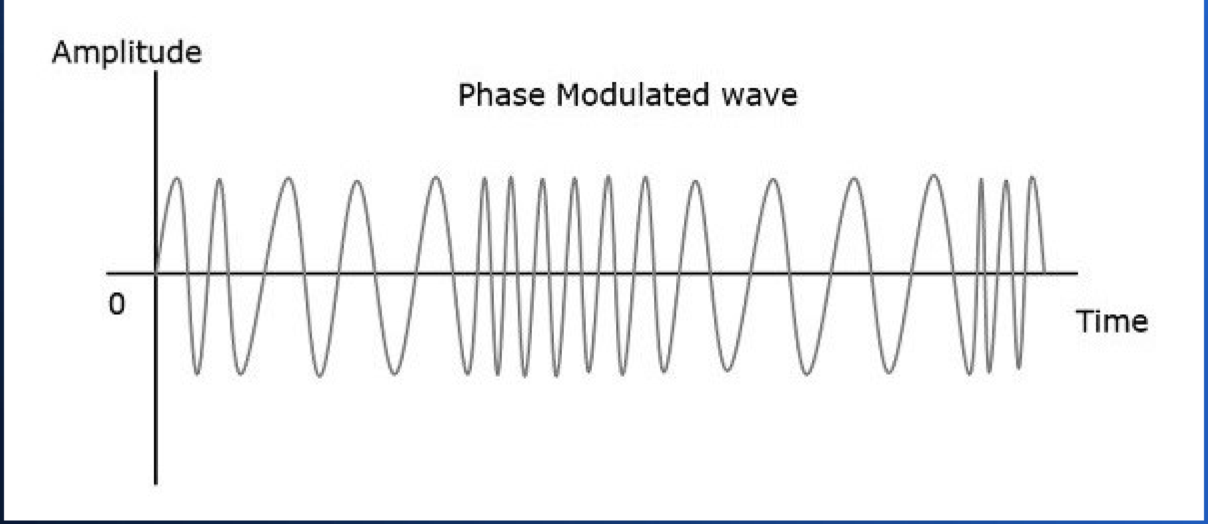
- Wideband FM
- Following are the features of Wideband FM.
-
- This frequency modulation has infinite bandwidth.
- The modulation index β is large, i.e., higher than 1.
- Its spectrum consists of a carrier and infinite number of sidebands, which are located around it. This was used in entertainment, broadcasting applications such as FM radio, TV, etc. (Most countries still use WFM for sound broadcasting around 100MHz, but have discontinued analouge television, for example the UK did so in 2012).

Introduction to Analogue Communication Systems

- Phase Modulation
- In frequency modulation, the frequency of the carrier varies. Whereas, in Phase Modulation (PM), the phase of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal.
- So, in phase modulation, the amplitude and the frequency of the carrier signal remains constant. This can be better understood by observing the following slide...
-

Introduction to Analogue Communication Systems

- The phase of the modulated wave has got infinite points, where the phase shift in a wave can take place. The instantaneous amplitude of the modulating signal changes the phase of the carrier signal. When the amplitude is positive, the phase changes in one direction and if the amplitude is negative, the phase changes in the opposite direction.



Introduction to Analogue Communication Systems

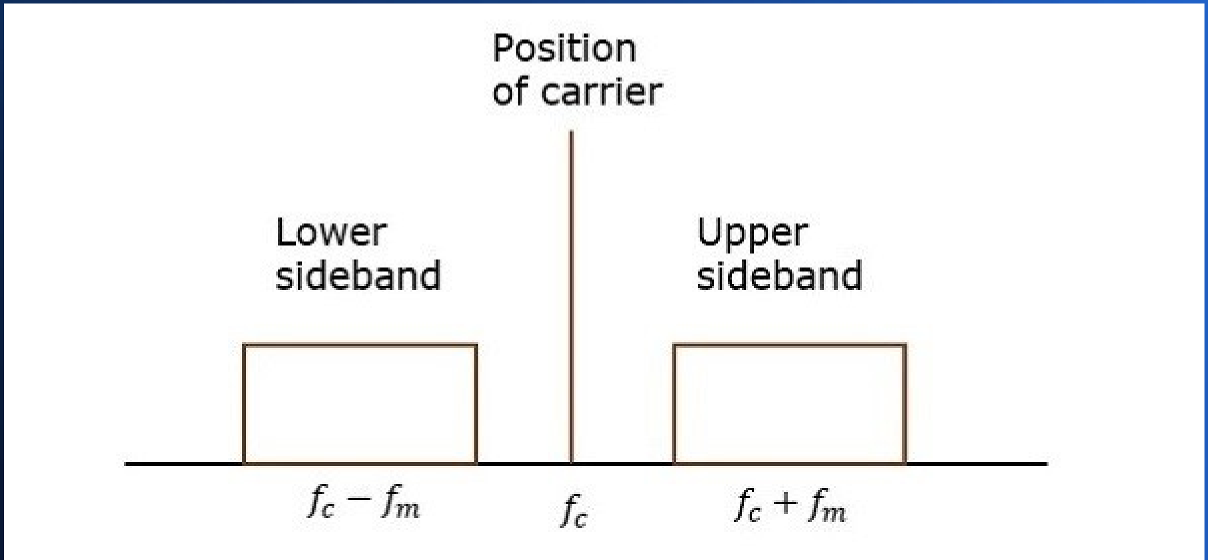
- Mathematical Representation:
- The equation for instantaneous phase ϕ_i in phase modulation is:
- $\phi_i = k_p m(t)$
- Where, k_p is the phase sensitivity, m(t), is the message signal.
- The equation of PM wave will be: $s(t) = A_c \cos(2\pi f_c t + \beta \cos(2\pi f_m t))$.
- Where, β = modulation index = $\Delta\phi = k_p A_m$
- Δ_ϕ is phase deviation.
- Phase modulation is used in mobile communication systems, while frequency modulation is used mainly for FM broadcasting.

Introduction to Analogue Communication Systems

- Single Sideband (SSB) and DSBSC (Double Sideband, Suppressed Carrier).
- In the process of Amplitude Modulation, the modulated wave consists of the carrier wave and two sidebands. The modulated wave has the information only in the sidebands. The sidebands is nothing but a band of frequencies, containing power, which are the lower and higher frequencies of the carrier frequency.
- The only advantage of this simple mode of communication is the simplicity of the circuitry needed. But as we shall see, it is inefficient.
- The transmission of a signal, which contains a carrier along with two sidebands can be termed as Double Sideband Full Carrier system or simply DSBFC. It is plotted as shown in the following slide...

Introduction to Analogue Communication Systems

- The reason that such a transmission is inefficient is because, two-thirds of the power is being wasted in the carrier, which carries no information.
- And the sidebands contain a mirror image of the same modulation. It follows that the carrier need not be transmitted, and optionally, one of the sidebands can also be ommited.

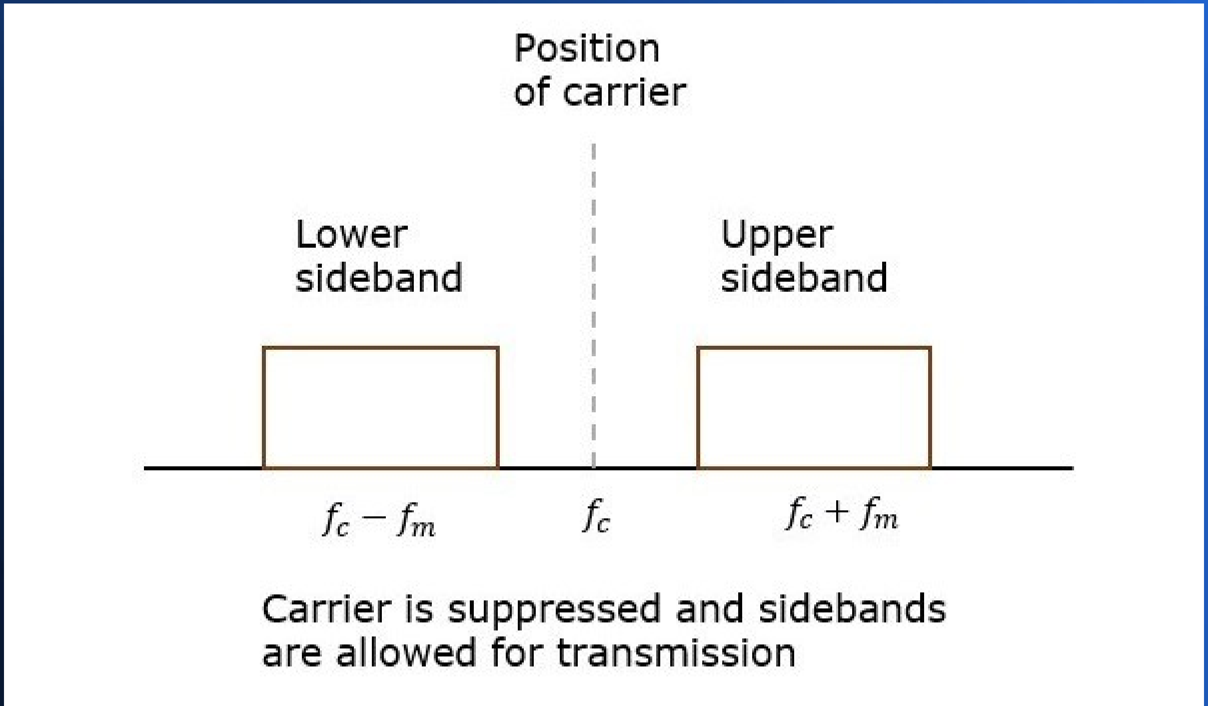


Introduction to Analogue Communication Systems

- For single sideband, we have upper and lower sideband (USB and LSB respectively).
- For amateur radio, an early convention was adopted where we use LSB on bands below 10MHz and USB on the HF bands above 10MHz. HF stands for high frequency, considered to lie between 3 and 30MHz, or wavelengths from 10 to 100 metres.
- Maritime and aeronautical comminucations on HF use USB throughout.
- Commercial systems often use DSBFC (double sideband, suppressed carrier).
- If this carrier is suppressed and the saved power is distributed to the two sidebands, then such a process is called as Double Sideband Suppressed Carrier system or simply DSBSC. It is plotted as shown in the following slide.

Introduction to Analogue Communication Systems

- Mathematical Expressions
- Let us consider the same mathematical expressions for modulating and carrier signals as we have considered in the earlier notes.



Introduction to Analogue Communication Systems

- Modulating signal: $m(t)=A_m\cos(2\pi f_m t)$
- Carrier signal: $c(t)=A_c\cos(2\pi f_c t)$
- Mathematically, we can represent the equation of DSBSC wave as the product of modulating and carrier signals.
- $s(t)={}_m(t)_c(t)$
- $\Rightarrow s(t)=AmAc\cos(2\pi fmt)\cos(2\pi fct)$
- Bandwidth of DSBSC Wave
- We know the formula for bandwidth (BW) is
- $BW=f_{max}-f_{min}$.

Introduction to Analogue Communication Systems

- Consider the equation of DSBSC modulated wave.
- The DSBSC modulated wave has only two frequencies. So, the maximum and minimum frequencies are f_c+f_m and f_c-f_m respectively.
- $f_{max}=fc+f_m$ and $f_{min}=f_c-f_m$.
- $\Rightarrow BW=2f_m$
- Power of DSBSC wave is equal to the sum of powers of upper sideband and lower sideband frequency components.
- Power of DSBSC:
- First, let us find the powers of upper sideband and lower sideband...

Introduction to Analogue Communication Systems

- Upper and lower sideband power must be added to get a power equation:
- Adding upper and lower sideband respectively gives: $P_t = \frac{A_m{}^2 A_c{}^2}{8R} + \frac{A_m{}^2 A_c{}^2}{8R}$
- In turn yielding: $\Rightarrow P_t = \frac{A_m{}^2 A_c{}^2}{4R}$
- Therefore, the power required for transmitting DSBSC wave is simply equal to the power of both of the sidebands (as the carrier is suppressed).
- For DSBSC power is ~ 1/3 tht of AM;
- For SSB power is @ 1/6 that for AM.
- And a SSB signal for voice only communication can be accomodated in just 3kHz of bandwidth, compared to as much as 10KHz for AM, and 150kHz for WFM for broadcast of high fidelity sound.

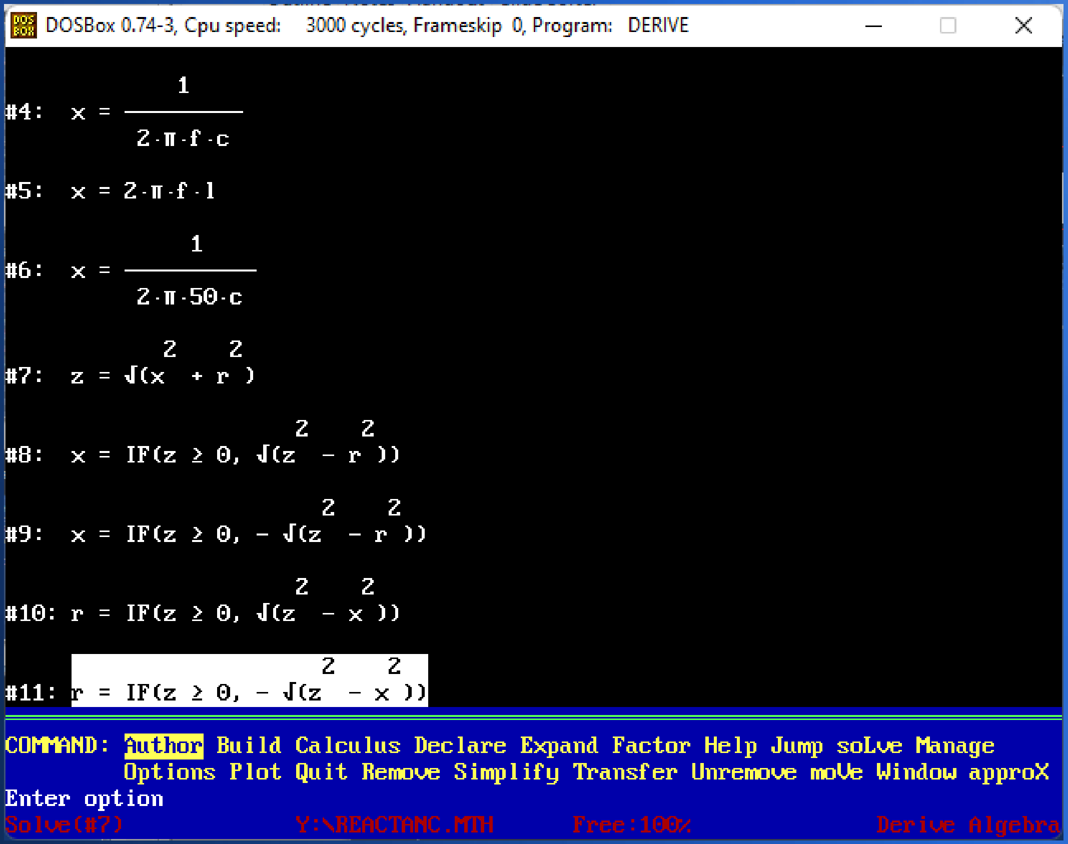
Introduction to Analogue Communication Systems

The following material is left here as it might be useful for reference.

- This is not part of the course, but it is there for potential reference in other study. It will not be in the exam..

Remember, a package such as Derive can always check your results!

- I used Derive as an Undergraduate - but other programs are available for all platorms.



AC Wattage

- In a DC circuit, power (W) = Voltage(V) X Current(I).
- For AC, be aware that reactance complicates matters. If a device includes an inductive and/or capacitance element, the apparent power and real power are different!
- This is because power can only be dissipated in a resistance.
- So the dissipation – except where a load is wholy resistive – can be less than voltage X current! This leads to the term 'power factor'.
- Imagine I put a large capacitor across the AC mains supply. I would draw a huge current, yet dissipate no power. However, it would nonetheless incur losses in the distribution system! Be aware that your electricity provider bills you for apparent power! So - improving power factor is important, particularly for large industrial customers.
- This is why substations include capacitors – transformers are inductive!

This is an example of a capacitor bank, for correcting power factor.

- Power factor will not be in the exam, but be aware that AC power is calculated as $P(W) = P_f VI$. Capacitor banks compensate as a counterbalance for large motors and transformers – which can have huge inductance values!

Wire Resistance

- What is the actual resistance of a piece of wire? The resistance value of a wire depends on all three of the following parameters: resistivity, length and diameter. The formula to calculate wire resistance is as follows:
- $R = \rho\ (l / A)$
- in which R is the resistance in (Ω),
- ρ is the resistivity of the material (Ω·m),
- l is the length of the material (m),
- A is the cross-sectional area of the material (m^2).
- It follows that a long thin wire has a much higher resistance than a short length of thick cable of the same material.
- In practice performance and economic considerations determine the type of cable used in a given application.

Electrical Resistivity – a brief introduction

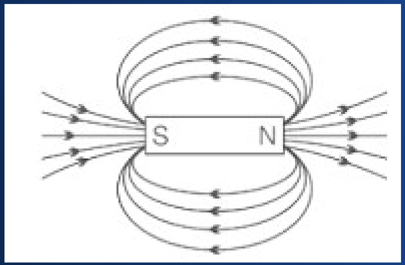
- Electrical resistivity is a measure of a material's property to oppose the flow of electric current. This is expressed in Ohm-meters (Ω·m). The symbol of resistivity is usually the Greek letter ρ (rho). A high resistivity means that a material does not conduct electric charge well.
- Electrical resistivity is defined as the relation between the electric field inside a material, and the electric current through it as a consequence:
- $\rho = E/J$ where:
- in which ρ is the resistivity of the material (Ω·m),
- E is the magnitude of the electrical field in the material (V/m),
- J is the magnitude of the electric current density in the material (A/m2)
- If the electric field (E) through a material is very large and the flow of current (J) is very small, it means that the material has a high resistivity.
- As an example, copper wire has a lower resistivity than nichrome wire (used to make heating elements.

Examples of Resistivity

Material	ρ (Ω·m) at 20°C	σ (S/m) at 20°C	Temperature coefficient (1/°C) x10^{-3}
Silver	1.59×10^{-8}	6.30×10^{7}	3.8
Copper	1.68×10^{-8}	5.96×10^{7}	3.9
Gold	2.44×10^{-8}	4.10×10^{7}	3.4
Aluminum	2.82×10^{-8}	3.5×10^{7}	3.9
Tungsten	5.60×10^{-8}	1.79×10^{7}	4.5
Zinc	5.90×10^{-8}	1.69×10^{7}	3.7
Nickel	6.99×10^{-8}	1.43×10^{7}	6
Lithium	9.28×10^{-8}	1.08×10^{7}	6
Iron	1.0×10^{-7}	1.00×10^{7}	5
Platinum	1.06×10^{-7}	9.43×10^{6}	3.9
Tin	1.09×10^{-7}	9.17×10^{6}	4.5
Lead	2.2×10^{-7}	4.55×10^{6}	3.9
Manganin	4.82×10^{-7}	2.07×10^{6}	0.002
Constantan	4.9×10^{-7}	2.04×10^{6}	0.008
Mercury	9.8×10^{-7}	1.02×10^{6}	0.9
Nichrome	1.10×10^{-6}	9.09×10^{5}	0.4
Carbon (amorphous)	5×10^{-4} to 8×10^{-4}	1.25 to 2×10^{3}	-0.5

Electromagnetism

- A magnet is a material or object that creates a magnetic field. While the magnetic field is completely invisible, it creates a force that pulls on other ferromagnetic materials, such as iron, steel, nickel, and cobalt. It can also attract or repel other magnets (like poles repel, unlike poles attract). While a magnet attracts these examples of magnetic materials, non-magnetic materials, such as rubber, coins, feather and leather, are not attracted. This is a diagram of the field around a simple bar magnet:



Electromagnetism

- Electric currents typically consist of huge numbers of electric charges that move in a coordinated, overall motion. However, unless you see it heat up and start glowing, it is not easy to tell from the outside whether a wire is carrying a current or not.
- This is because a conductor remains electrically neutral while electrons move through it. Any excess electrons that enter a segment of the wire on one end will simultaneously be made up for by electrons leaving that segment on the other end. Remember, the conductor contains equally many positive charges in the nuclei of its atoms, as there are electrons in it.
- Electromagnetism is the best way to detect and quantify how many amperes of current is going through a circuit. It is created by the motion of the negatively-charged electrons that make up the current, whereas the positively-charged nuclei have no magnetic effect because they are not moving! So while the electric influences of electrons and nuclei cancel out as seen from the outside, their magnetic effects do not.

Electromagnetism

- In a galvanometer (the core of an ammeter), a magnetic field is converted into a force that moves a needle. This is done by exploiting the effect discussed in this text – a current-carrying wire feels a force when it is surrounded by a magnetic field.
- Motors are the most common application of magnetic force on current-carrying wires. Motors have loops of wire in a magnetic field. When current is passed through the loops, the magnetic field exerts torque on the loops, which rotates a shaft. Electrical energy is converted to mechanical work in the process.
- Metres, such as the galvanometer, are another common application of magnetic torque on a current-carrying loop. This finally answers the question how ammeters actually work (we treated them as black boxes in the discussion of circuits earlier). As with motors, the basic idea is to convert the magnetic force into a twisting action, also called torque. This is how the indicator arrow on a meter is made to rotate to a given position, indicating how many amperes are flowing through the meter.

Electromagnetism

- How are elecmagnetic fields created in the first place, and what determines their strength?
- An electromagnet uses an electric current to create the same magnetic forces we have just discussed. We use electromagnets for everything from a crane in a scrapyard which lifts scrapped cars, to controlling the beam of a particle accelerator. But if you look at an electromagnet closely, it is nothing but a loop coil of wire, just like the coils we just mentioned in motors and metres.
- How can we use the same device (a coil) for two different purposes: creating a force on a current, as in a motor, and turning a current into a magnetic field?
- Recall Newton's Third Law: every action creates an equal and opposite reaction. In a motor, a magnet created a force on the current-carrying coil via the magnetic field. By Newton's Third Law, the current-carrying coil must simultaneously be exerting a force on the magnet. That is the magnetic field created by the coil, and it makes the coil into an electromagnet.

Electromagnetism and Ampere's Law

- To quantify the strength and direction of the magnetic field created by flowing currents, it is best to start with the simplest case of a straight wire.
- In all cases, the magnetic field is proportional to the current. But the way the magnetic field behaviour depends on your position relative to the wires is very much affected by the geometry of the wires.
- The simplest behaviour is found for a straight wire: the magnetic field in this case decreases inversely with the distance measured perpendicular to the wire.
- Ampère's Law is a discovery André-Marie Ampère made - it is as an example of Newton's Third Law in action, because it puts the "source" and "recipient" of a magnetic force on equal terms -they must be (as action and reaction are equal!).

Electomagnetism – time for some numbers!

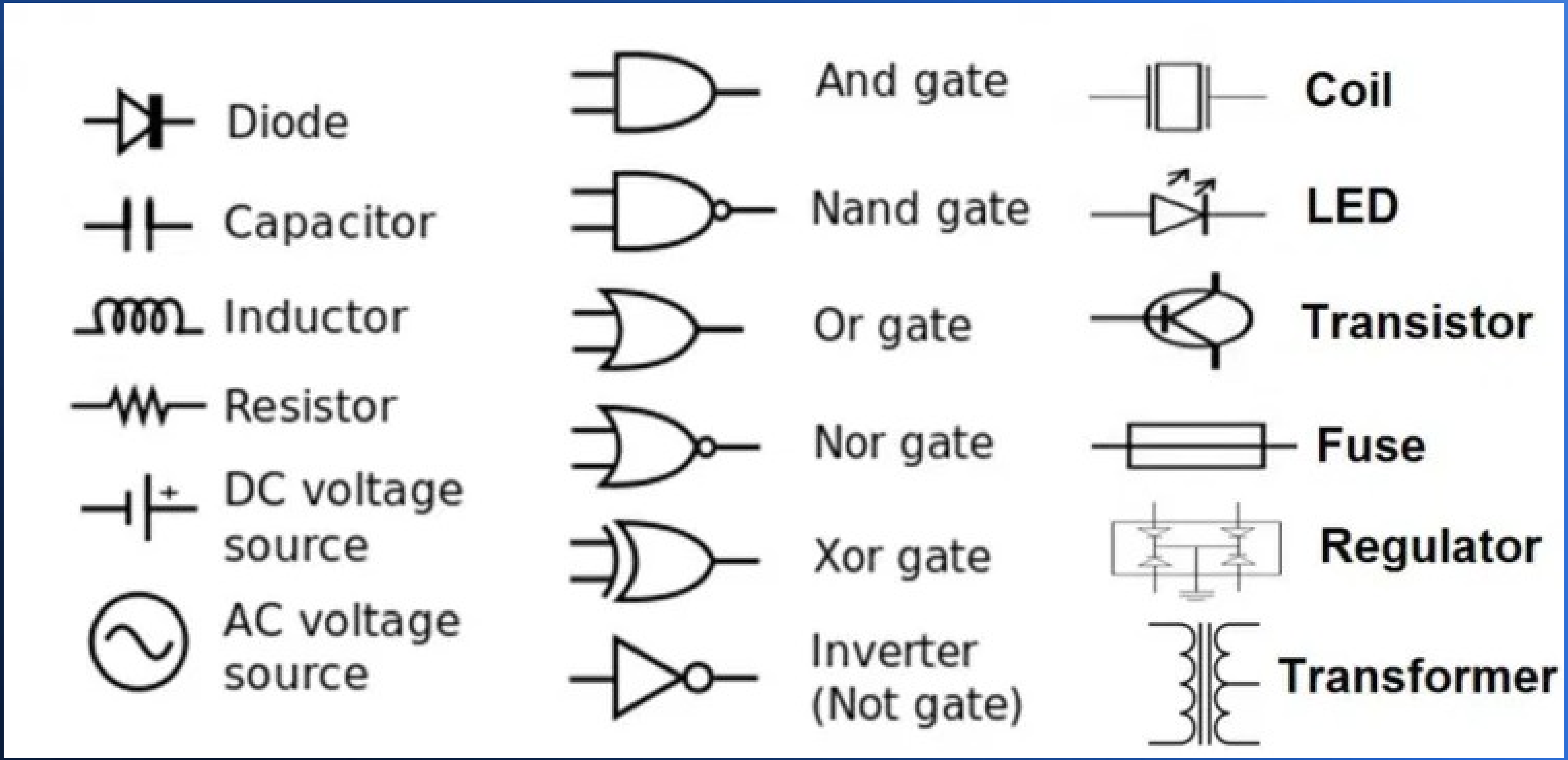
- A Tesla is equal to a Newton per meter and ampere. An exemplary example illustrates this: It corresponds exactly to the flux density of a Tesla, which exerts on a 1 meter long electrical conductor, which in turn conducts a current of 1 ampere, exactly 1 Newton attraction.
- The unit Tesla (T) in magnetism: The Tesla was named after the engineer and inventor Nikola Tesla. The definition of the magnetic flux density does not correspond directly to that of the magnetic field. However, it can ultimately be specified in the two quantities (units) Gauss and Tesla. The following relationship applies to convert the Tesla unit:
- 1 Tesla = 10,000 Gauss
- 1 T = 1000 mT)
- 1KG (outside) = 0.1T

Some Basic Components

- Function of the most Basic Electronic Components
- Terminals and Connectors: Components to make electrical connection.
- Resistors: Components used to resist current.
- Switches: Components that may be made to either conduct (closed) or not (open).
- Capacitors: Components that store electrical charge in an electrical field.
- Magnetic or Inductive Components: These are Electrical components that use magnetism such as inductors..
- Network Components: Components that use more than 1 type of Passive Component.
- Piezoelectric devices, crystals, resonators: Passive components that use piezoelectric. effect.
- Semiconductors: Electronic control parts with no moving parts.
- Diodes: Components that conduct electricity in only one direction.
- Transistors: A semiconductor device capable of amplification.
- Integrated Circuits or ICs: A microelectronic computer circuit incorporated into a chip or semiconductor; a whole system rather than a single component.

Some Basic Components

Symbols of Common Components

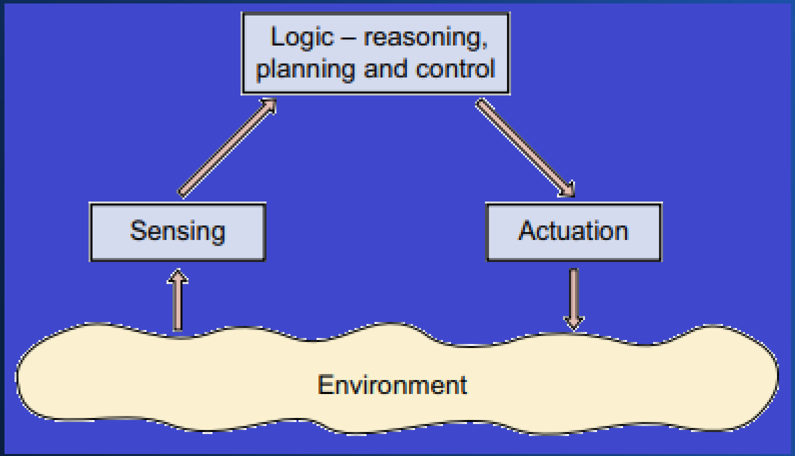


Opto-electronic (optical electronic) components

- There are various components that can turn light into electricity or vice-versa. Photovoltaic cells (also known as photoelectric cells) generate electric currents when light falls on them and they're used in various types of sensing equipment, including some type of smoke detector. Light-emitting diodes (LEDs) work in the opposite direction, converting small electric currents into light, and are typically used on the instrument panels of audio equipment. Liquid crystal displays (LCDs), such as those used in flatscreen televisions and laptop computers, are more sophisticated examples of opto-electronics.
- On the large scale, photovoltaic cells convert solar energy into electrical energy. I plan to establish a solar powered electronic business in Western Africa.
- And as we are about to see, the sensing/logic/actuation cycle depends heavily on sensors, including optoelectronic devices.

What is the sensing-logic-actuation cycle?

- Electronic devices can sense the world around them, converting a wide variety of physical phenomena into electrical signals that communicate useful information. Such devices (called transducers) have capabilities similar to our own human senses: hearing (microphones), seeing (cameras, including visible light and infrared, and proximity sensors), touch (piezoelectric transducers), and smell and/or taste (chemical sensors).



Sensing – a simple example

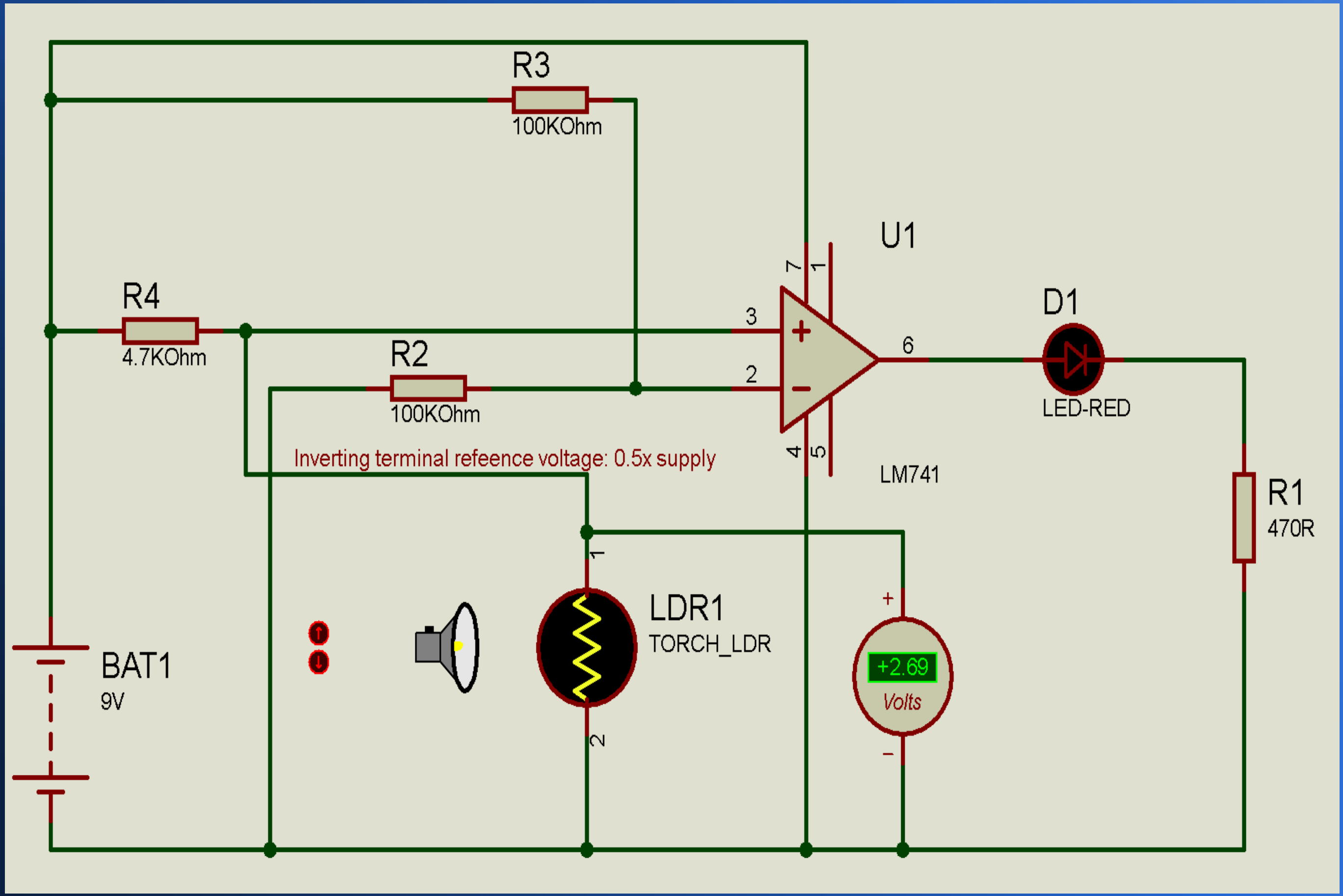
- A light dependent resistor used as a daylight sensor in a Proteus Simulation. As the sensor is virtual, the torch stands in for actual daylight. A UA741 is configured as a comparator in the following circuit.

This is a simple demonstration of how lighting can be controlled using a light dependent resistor. The LDR serves as a daylight detector.

The idea is that the lighting is triggered automatically when the light falls below a given level. The torch in this demonstration stands in for daylight in this demonstration circuit. And the UA741 operational amplifier here is used as a comparator. The reference voltage is half the supply voltage.

Sensing – Daytime (torch on) – note D1 light emitting diode is OFF.

Once light level reduces the voltage below the reference value of 4.5v, D1 turns off. An easy way to automate illumination.

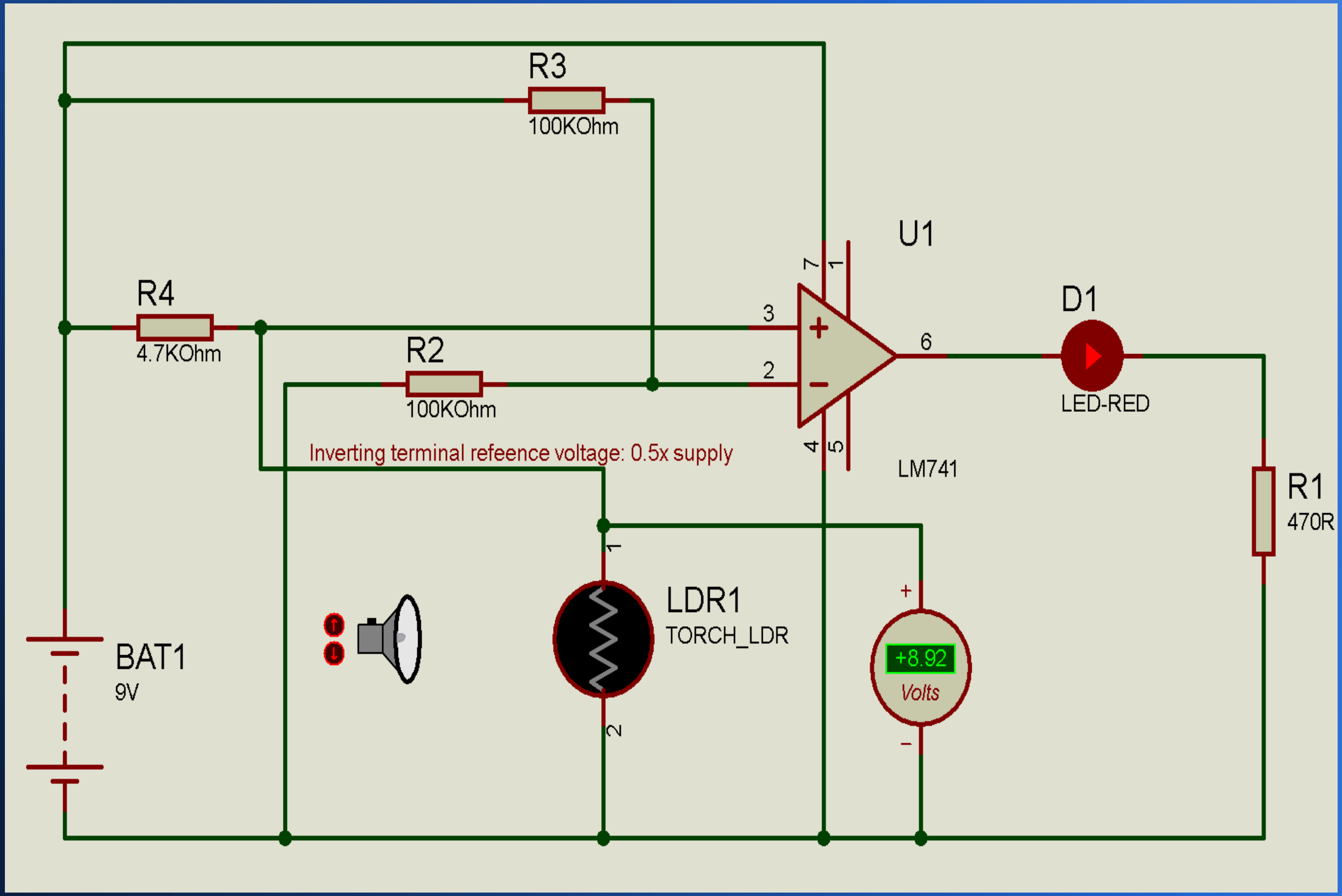


Sensing – Nighttime (torch off) – note D1 light emitting diode is ON.

8.92v is above

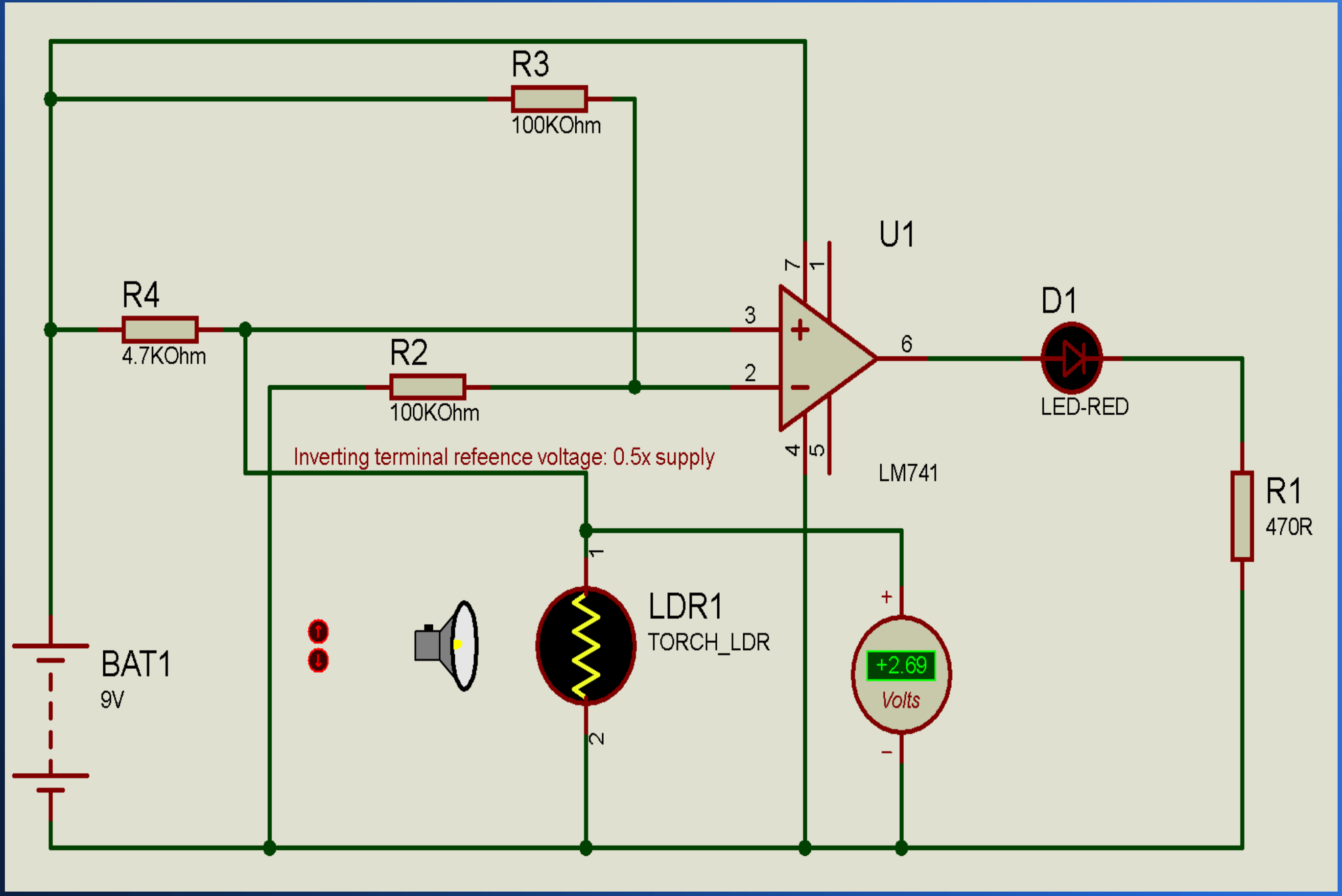
the Reference

Voltage.



Sensing – Daytime (torch on) – note D1 light emitting diode is OFF.

Once light level Increases, the voltage below the reference value of 4.5v, D1 turns off. An easy way to automate illumination.



Sensing beyond human senses

- It is possible to design electronic devices which can sense things which we cannot sense directly. For a few examples:
- Ultrasound allows us to ‘see’ inside our bodies;
- Infrared camera images allow us to ‘see’ pictures of radiated heat;
- terahertz (micrometric) images allow us to see through opaque materials
- Our eyes are only sensitive to a tiny 'window' in the electromagnetic specrum....

Sensing beyond human senses

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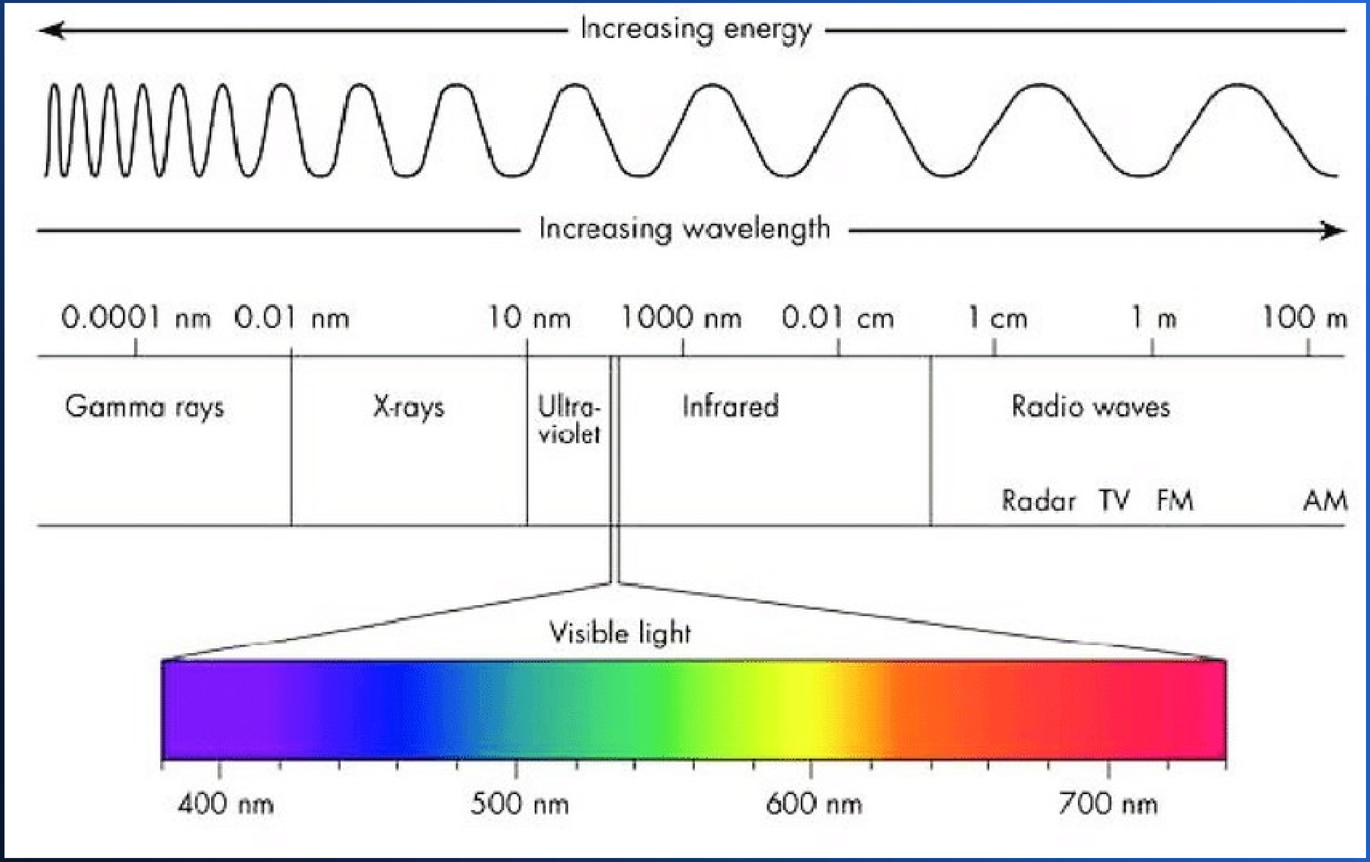
The electromagnitic spectrum

- Our eyes sense only wavelengths from 0.4 – 0.7μm.



Logic – what is it?

- Sometimes the information from sensors is fed directly to a human being to act on, as in a visual display. However, in many cases that information is used to control systems automatically. To do this requires the functions of logic, which are carried out by logic circuits or programmable microprocessors or microcontrollers.

Actuation

- Actuators are components that control the movement in an autonomous system. In many systems, actuators of various kinds are automatically controlled to give the desired behaviour. Examples include electric motors (including stepper motors), and pneumatic actuators.
- Robotics are possible through clever use of actuation!

Active and Passive Electronic Components

- Examples and Differences between active and passive electronic components:
- Active electronic components are those that can control the flow of electricity. Different types of common basic circuits generally have at least one active component.
- Some examples of active electronic components are transistors, vacuum tubes, and silicon-controlled rectifiers (SCRs).
- Passive electronic components are those that don’t have the ability to control electric current by means of another electrical signal. Examples of passive electronic components are capacitors, resistors, inductors, transformers, and some diodes.

List of Active Electronic Components

- Transistors, Diodes (All), Rectifier Diodes, Schottky Diodes
- Zener Diodes, Unipolar / Bipolar Diodes, Varicaps, Varactors
- Light-Emitting Diode (LED), Solar PV Cell, PV Panel
- Transistors (All), Photo Transistors, Darlington Transistors
- Compound Transistors, Field-Effect Transistors (FET).
- JFET (Junction Field-Effect Transistor)
- MOSFET (Metal Oxide Semiconductor FET)
- Thyristors
- Composit Transistors

List of Passive Electronic Components

- Resistors (All Types), Capacitors (All Types), Inductors / Coils
- Memristor / Networks, Sensors, Detectors, Transducers
- Antennas, Assembly Modules, Piezoelectric devices, Crystals
- Resonators, Terminals and Connectors, Cables, Switches
- Circuit Protection Devices (Such as fuses, earth leakage & other breakers)
- PCB's (The printed circuit board into which compont are mounted);
- Mechanical Devices such as a Fan, Lamp or Motor
- The above conduct as a function of a simple mathematical relationship.

Ideal and real world components: some basic differences

- Only real devices can be measured, and only ideal elements can be calculated or simulated! An equivalent electrical circuit model is an idealised electrical description of a real structure.
- This is why real world results may not agree with those from a circuit simulation package such as Proteus (my usual choice nowadays), Pspice, Electronics Workbench (used at university and since graduation).
- And as we will see, random and systematic errors can give misleading results in laborotory work – as addressed in this course!
- Even equipment used for measurements is not 'ideal' – for example, an ideal voltmeter would have an infinite resistance; an ideal ampmeter would have zero resistance. And in the real world, measuring equipment can be affected by factors such as the temperature in the laboritory!

Real circuits vs. simulated circuits: real components (R, L, C) vs. ideal components

- As a graduate and an experienced engineer, and a radio amateur licenced since 1995, I can speak from experience here.
- This course is an introductory course. So I will only give a basic outline here regarding the differences between ideal and 'real world' components.
- Once you gain experience, you come to a realisation that throughout what you have been taught about electronic components and circuit schematics as a beginner, there is a hidden side that is rarely talked about and not ever discussed in detail.
- There is a lot of difference between the ideal schematic that shows the intended current flow with ideal components that follow the rules, and where the connecting wires have no resistance, capacitance and inductance, and what we have to deal with in practice...

Difference between theory & practice

- The real world is completely different to a simulation. The 'hidden aspects' in a practical circuit tend to manifest themselves most at high frequencies.
- Resistors have frequency responses, capacitors have inductances, and inductors have resistances; PCB tracks have inductances and also capacitances between them.
- As an expeienced radio amateur and circuit designer, I am well aware that for example, a voltage controlled oscillator performs fairly adequately at HF frequencies (3-30MHz) if well designed, but is quite simply too unstable at VHF (30-300MHz), let alone UHF (300MHz – 3GHz)!
- Here are some brief examples of why the real world differs from the ideal. The list is NOT exhaustive.

Factors which cause circuit behaviour to deviate from a simulation

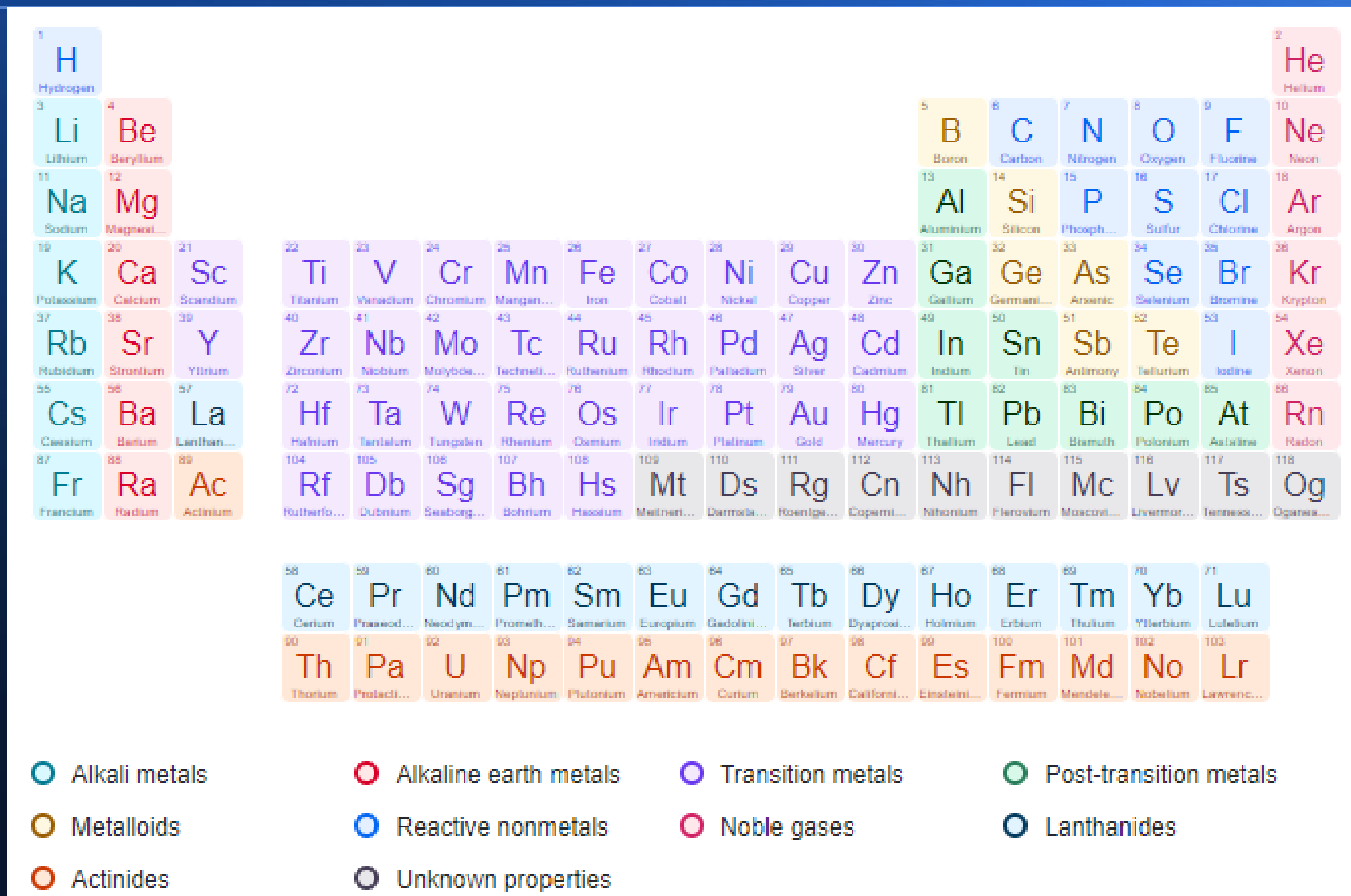
- As mentioned previously, even straight wires generate a magnetic field – hence they have inductance
- All conducting materials have some resistance, with the exception of a true superconductor very close to absolute zero (-273.16ºC, 0ºK)
- A wire wound resistor is a coil of wire – this is how an inductor is made, after all! Indeed all inductors, including the windings of motors and transformers, posess some resistance.
- Capacitor plates are seperated by an insulating gap – the dielectric. In practice the insulating material is not perfect! Capacitors, eventually, lose charge through their own shunt resistance, and a high enough voltage will ionize the air between and arc – spark! Across the gap!
- Spaces between wires act as a stray capacitor – hence 'stray capacitance'.

Factors which cause circuit behaviour to deviate from a simulation

- Capacitors made of a coil of aluminium foil with a dielectric seperator in a 'swiss roll' arrangement posess inductance as well as the intended capacitance. As will manifest itself at higher frequencies!
- Transformers have loss due to ohmic resistance in the windings, stray 'eddy' current in the core, capacitance between the primary and secondary coils
- Resistors (and in particular semiconductors) have a temperature co-efficient
- Even diodes have stray capacitance – as do transistor junctions
- <u>**And measured values can at best be only as good as the measurement technique employed!**</u>

Periodic Table of the elements

(For referrence, Silicon and Germanium are Matalloids).



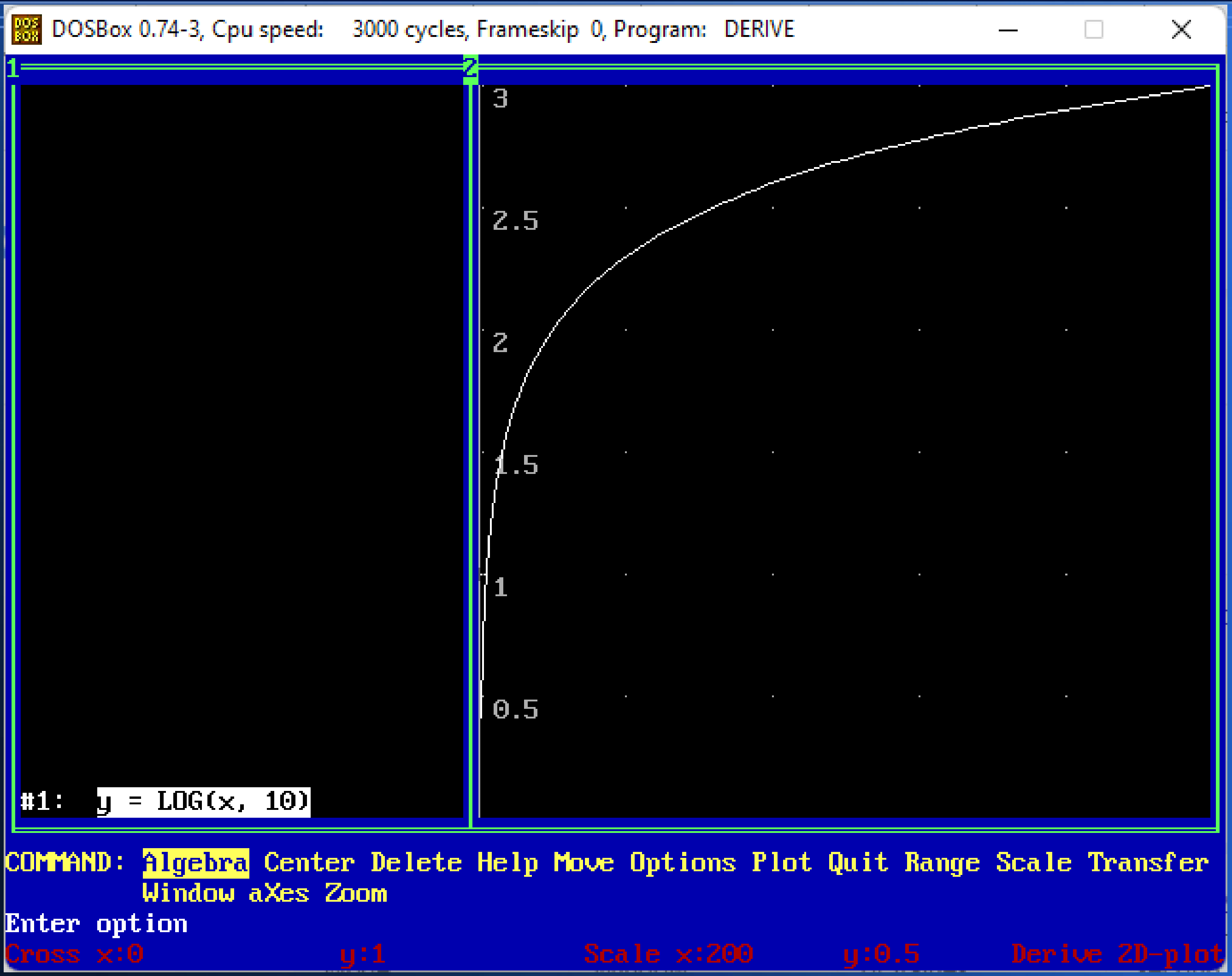
How to Create a linearised Logarithmic Graph and equation

- <u>How to Write the equation of a Linear Function whose Graph has a Line that is non linear – and to plot it in a linear form</u>
- Many functions have a lon-linear graph and equation.
- So -how to Write the equation of a Linear Function whose Graph has a Line with a straight gradient, and express the graph in linear form:
- Many functions have a non-linear relationship. Examples include: $y = x^2$, y =LN (X), y = log base n (x), y = ê^x, y = 1/x.
- A linear/log graph, formally known as a semi-logarithmic graph, is a graph that uses a linear scale on one axis and a logarithmic scale on the other axis. It's useful in science for plotting data points of two variables where one of the variables has a much larger range of values than the other variable. By plotting the data in this way, we can frequently observe relationships in the data that would not be as obvious if both variables were plotted linearly.

How to Create a Logarithmic Graph and equation

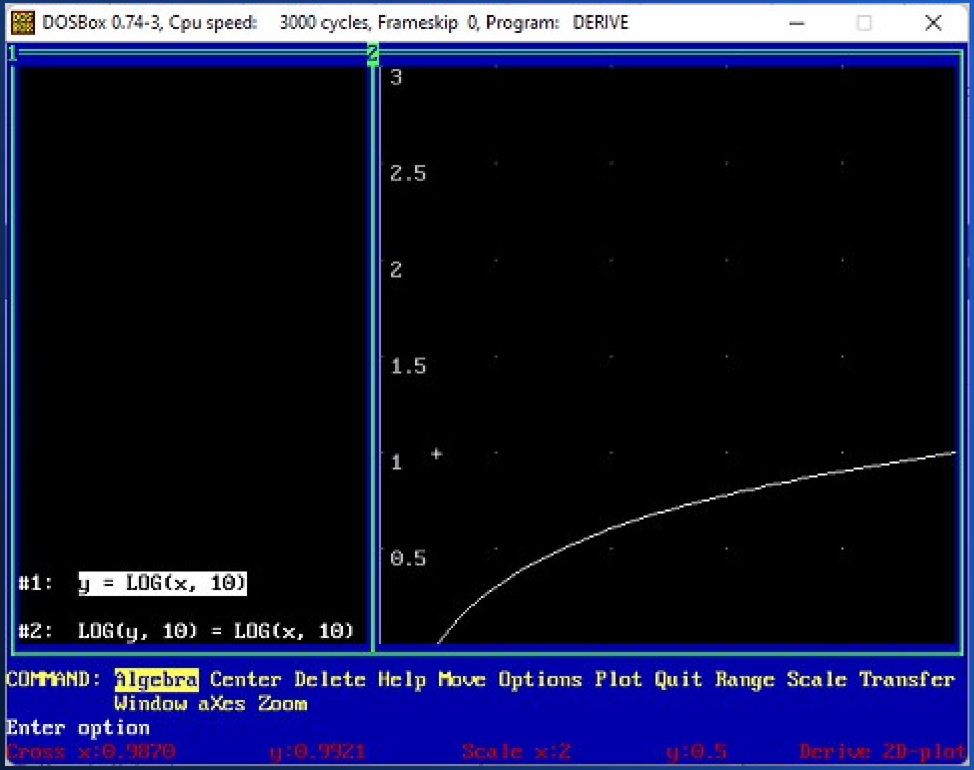
- Let us use the example of $y = \log_{10}(x)$. This is entered in Derive as y = log(x,10) x.
- The problem here is that relationships which would be obvious on a linear graph – of linear gradient – are hidden here.
- Also, many quantities which you encounter in engineering give very wide ranges of values. An example of such a quantity is sound intensity. Our ears detect sound logarithmicly – a sound you percieve as twice as loud may actually have a change in amplitude of a factor of ten.
- Simimarly, our eyes are also logarithmic in terms of light intensity. This is because they need to avoid damage in bright sunlight, but become very sensitive at night (they also sacrefice colour vision to improve sensitivity – which is why you only see in monochrome in the dark).

Let us use $y = \log_{10}(x)$ as an example.



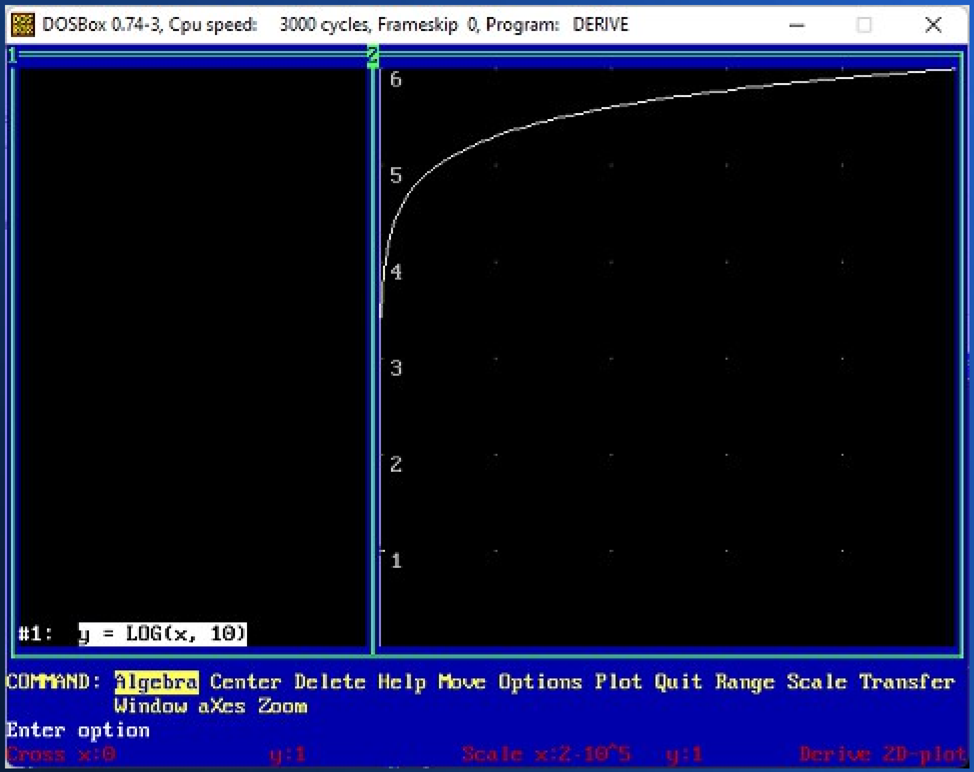
Let us view this on different scales...

- The range of the graph is shown at the bottom.
- Here x range is 0-10.



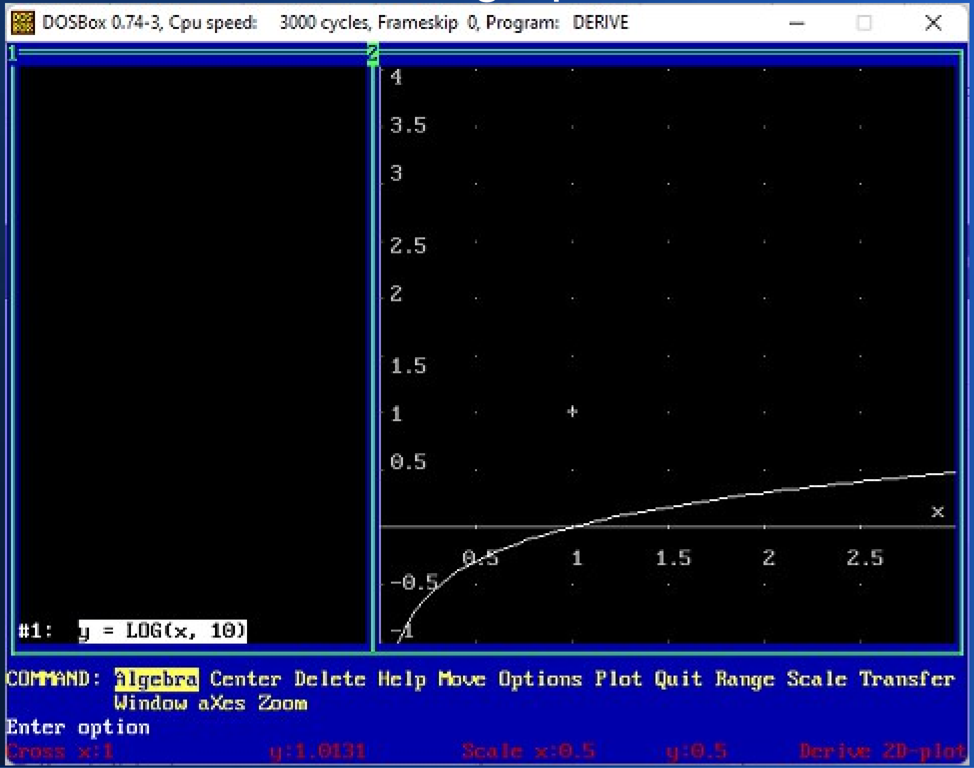
Let us view this on different scales...

- Here x range is 0 to 10^6 (on million).



Let us view this on different scales...

- And remember, y = log10 (x) – indeed all logarithmic graphs, start at – where
- Y = 0. So, how can we linearise the graph? Well, here comes the trick...

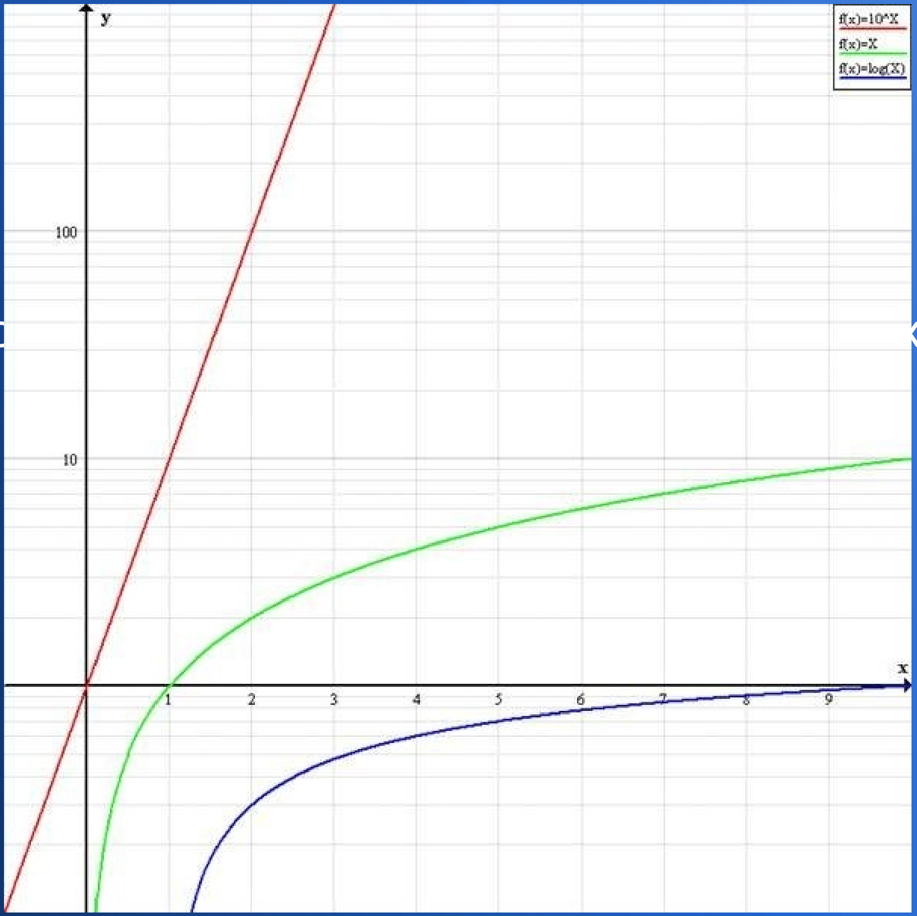


How to Create a linearised Logarithmic Graph and equation

- A log graph, formally known as a semi-logarithmic graph, is a graph that uses a linear scale on one axis and a logarithmic scale on the other axis. It's useful in science for plotting data points of two variables where one of the variables has a much larger range of values than the other variable. By plotting the data in this way, we can frequently observe relationships in the data that would not be as obvious if both variables were plotted linearly.
- It also has the advantage of allowing very wide ranges to actually fit onto a graph!

How to Create a linearised Logarithmic Graph and equation?

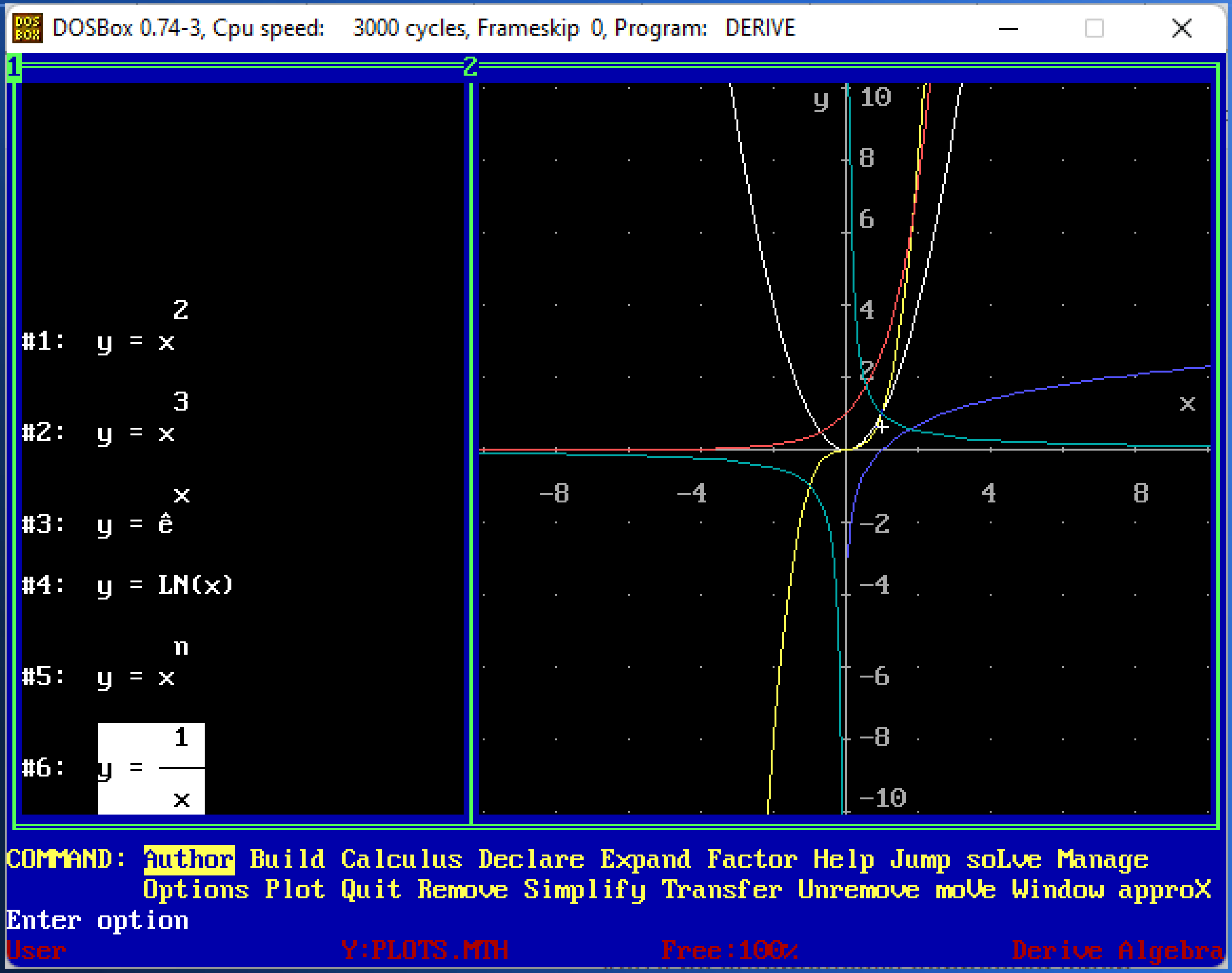
- Use a lin-log graph. This type of log graph has a y axis with a linear scale and an x axis with logarithmic scale. The scale of the x axis is therefore compressed by a factor of 10^x in relation to the y axis. In the illustration, y = lo x on the linear graph. Y = 10^x in red intersects the y axis at x = 10 and has a positive slope that approaches infinity. Y = x in green now looks like y = 10^x on the linear graph.



- This is known as a semi-logarithmic graph – it is linear on one axis and logarithmic on the other.

How to Create a linearised Logarithmic Graph and equation

- Examples of graphs which you might want to Linearise are here. The table on the next page shows how to plot them.

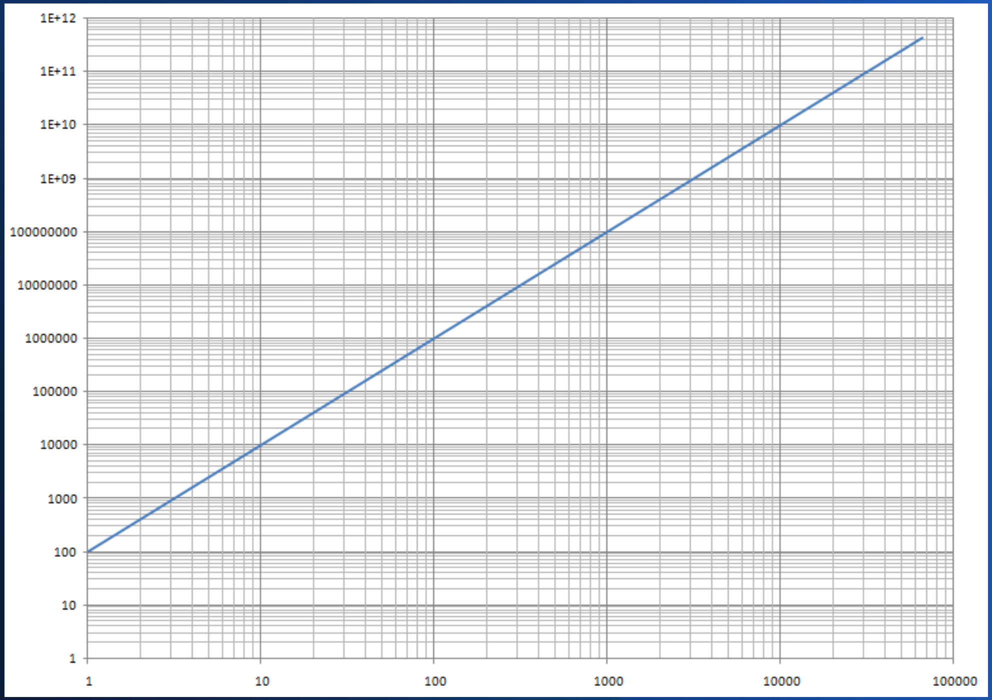


How to linearise these examples...

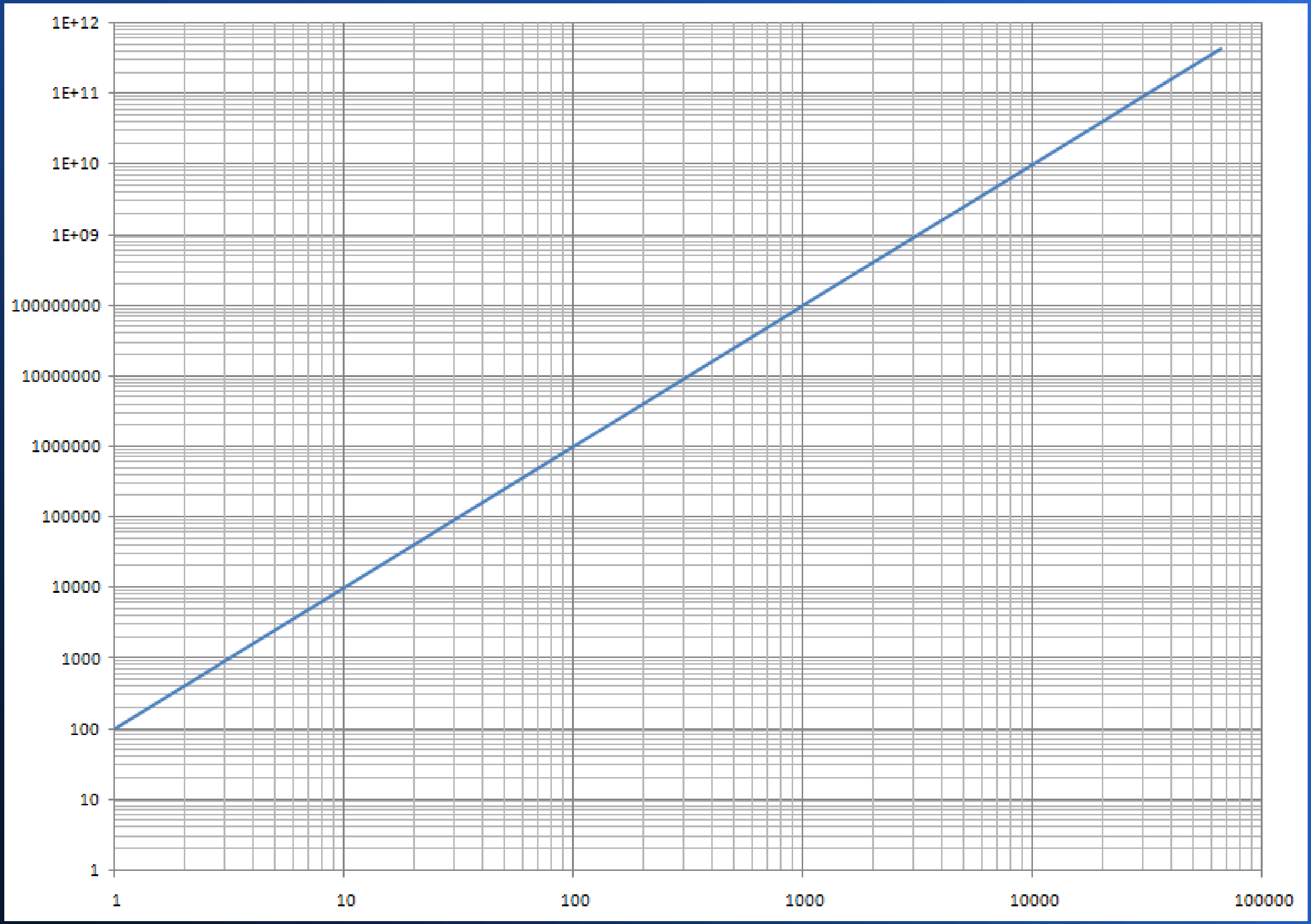
Expression	Linear plot – x axis	Linear plot - y axis
$Y = x^2$	x^2	Y
$Y = x^3$	x^3	y
Y = ê^x (Exponential)	ê^x	y
Y = ln(x) (Natural logarithm)	ln(x)	y
Y = x^n	x^n (if n value is known)	y
Y = 1/x (reciprocal).	1/x	y

Using a Log-Log Graph

- Occasionally, both values required have too wide a range to show on a linear graph. In this case, we can use a log/log graph so the scale of both axes is logarithmic.
- For example, here is a graph of $y=100x^2$ for x values from 1 to 100000, 10^5.



Using a Log-Log Graph: Notice how the plot is divided evenly into "blocks" by powers of ten. This allows a flat gradient to be visible on the graph. You can now easily see the relationship between the values.



In Microsoft Excel, how can I make a log or log-log graph?

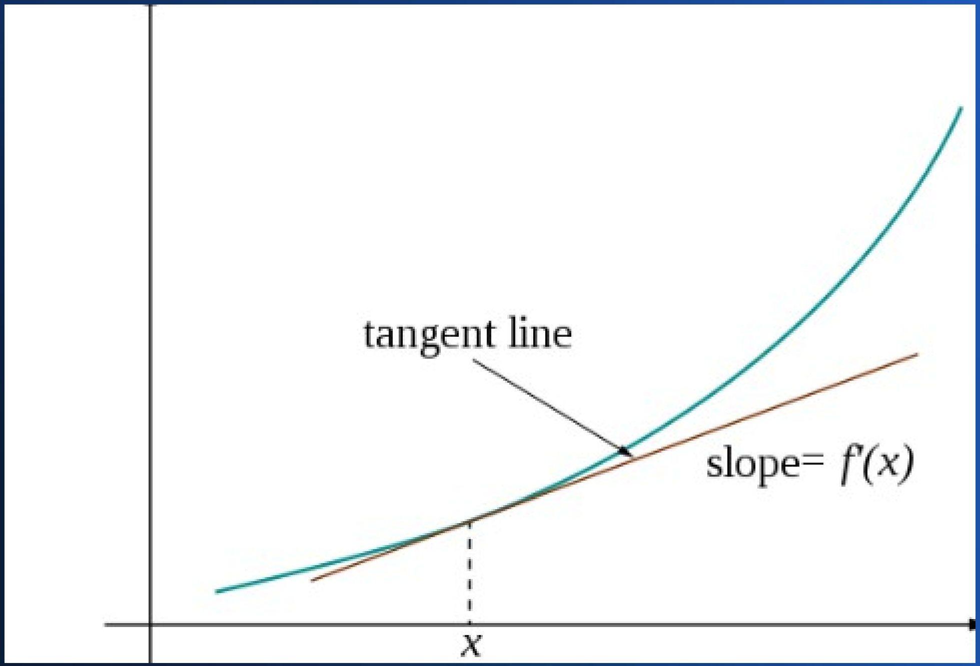
- To create a log-log graph in Microsoft Excel, you must first create an XY (scatter) graph. This is the only graph type that will work; other graph types permit logarithmic scales only on the Y axis. To create a log-log graph, follow the steps below for your version of Excel.
- For Excel 2010 or 2007:
- In your XY (scatter) graph, right-click the scale of each axis and select Format axis....
- In the Format Axis box, select the Axis Options tab, and then check Logarithmic scale.
- For older versions of Excel:
- In your XY (scatter) graph, double-click the scale of each axis.
- In the Format Axis box, select the Scale tab, and then check Logarithmic scale.
- I have not tried it in newer versions of Excel.

Linearising an Equation

- In mathematics, linearisation is finding the linear approximation to a function at a given point. The linear approximation of a function is the first order Taylor expansion around the point of interest. In the study of dynamical systems, linearisation is a method for assessing the local stability of an equilibrium point of a system of nonlinear differential equations or discrete dynamical systems.
- Linearisations of a function are simply straight lines—usually lines that can be used for purposes of calculation. Linearisation is an effective method for approximating the output of a function at any given point based on the value and slope of the function at this point. In essence, linearisation approximates the output of a function very close to a given value.
- Please be aware that this will only be of reasonable accuracy over a narrow range of X values.

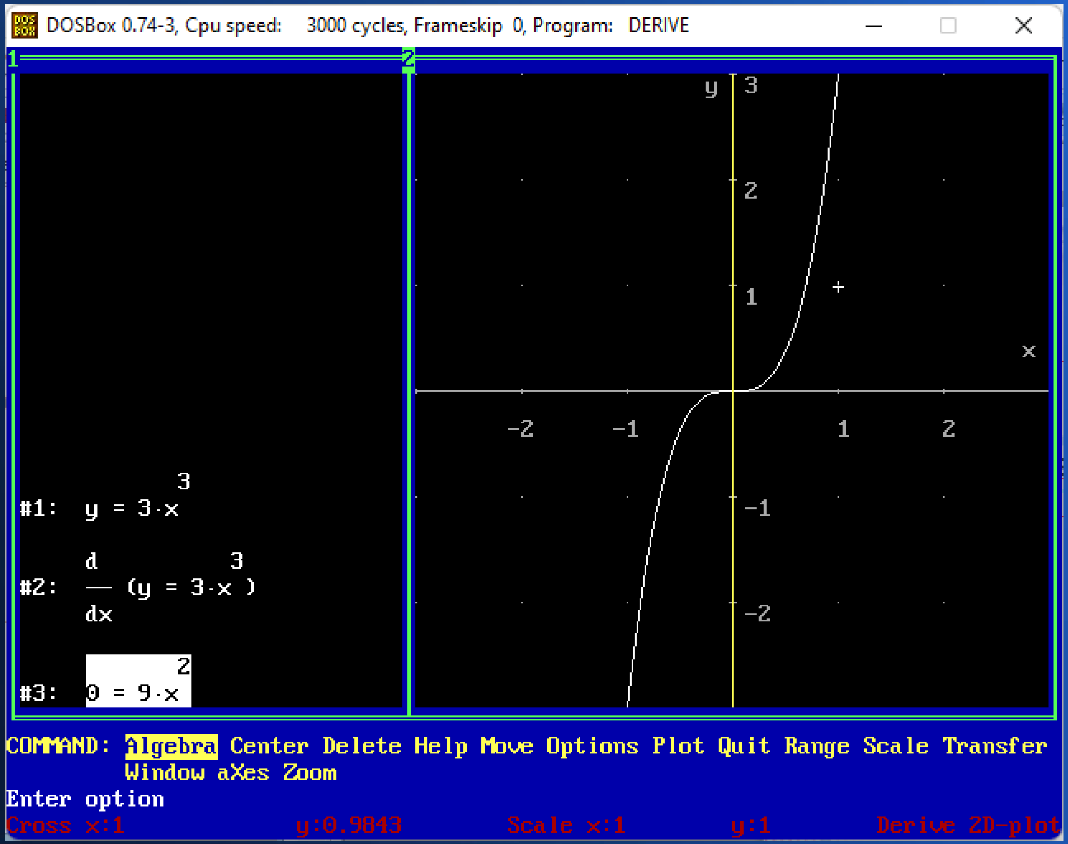
Linearising an Equation

- Because differentiable functions are only vary locally linear, the best slope to substitute in would be the slope of the line tangent to f(x) at x = a.
- For example, if f(X) = x² then drawing a tangent line to the graph will provide an approximation of the gradient for very narrow ranges of f(x) only.



Linearising an Equation

- The only way to find a general linearisation for a function is to find its derivitive. For example, the derivative of $y = 3x^3$ is $9x^2$.



Examples of common derivatives

- Please note that these are common derivatives. These find the general case of the gradient of a function.

for a very narrow range of f(x), drawing a tangent is an approximation.

$$\frac{d}{dx}(a)=0$$

$$\frac{d}{dx}(x)=1$$

$$\frac{d}{dx}(au)=a\frac{du}{dx}$$

$$\frac{d}{dx}(u+v-w)=\frac{du}{dx}+\frac{dv}{dx}-\frac{dw}{dx}$$

$$\frac{d}{dx}(uv)=u\frac{dv}{dx}+v\frac{du}{dx}$$

$$\frac{d}{dx}\left(\frac{u}{v}\right)=\frac{1}{v}\frac{du}{dx}-\frac{u}{v^2}\frac{dv}{dx}$$

$$\frac{d}{dx}(u^n)=nu^{n-1}\frac{du}{dx}$$

$$\frac{d}{dx}(\sqrt{u})=\frac{1}{2\sqrt{u}}\frac{du}{dx}$$

$$\frac{d}{dx}\left(\frac{1}{u}\right)=-\frac{1}{u^2}\frac{du}{dx}$$

$$\frac{d}{dx}\left(\frac{1}{u^n}\right)=-\frac{n}{u^{n+1}}\frac{du}{dx}$$

$$\frac{d}{dx}[f(u)]=\frac{d}{du}[f(u)]\frac{du}{dx}$$

$$\frac{d}{dx}[\ln u]=\frac{d}{dx}[\log_e u]=\frac{1}{u}\frac{du}{dx}$$

$$\frac{d}{dx}[\log_a u]=\log_a e\,\frac{1}{u}\frac{du}{dx}$$

$$\frac{d}{dx}e^u=e^u\frac{du}{dx}$$

$$\frac{d}{dx}a^u=a^u\ln a\frac{du}{dx}$$

$$\frac{d}{dx}(u^v)=vu^{v-1}\frac{du}{dx}+\ln u\;u^v\frac{dv}{dx}$$

$$\frac{d}{dx}\sin u=\cos u\frac{du}{dx}$$

$$\frac{d}{dx}\cos u=-\sin u\frac{du}{dx}$$

$$\frac{d}{dx}\tan u=\sec^2 u\frac{du}{dx}$$

$$\frac{d}{dx}\cot u=-\csc^2 u\frac{du}{dx}$$

$$\frac{d}{dx}\sec u=\sec u\tan u\frac{du}{dx}$$

$$\frac{d}{dx}\csc u=-\csc u\cot u\frac{du}{dx}$$

Errors and Uncertainties

- The difference between uncertainty and error:
- The main difference between errors and uncertainties is that an error is the difference between the actual value and the measured value, while an uncertainty is an estimate of the range between them, representing the reliability of the measurement. In this case, the absolute uncertainty will be the difference between the larger value and the smaller one.
- A simple example is the value of a constant. Let's say we measure the resistance of a material. The measured values will never be the same because the resistance measurements vary. We know there is an accepted value of 3.4 ohms, and by measuring the resistance twice, we obtain the results 3.35 and 3.41 ohms.
- Errors produced the values of 3.35 and 3.41, while the range between 3.35 to 3.41 is the uncertainty range.

Errors and Uncertainties

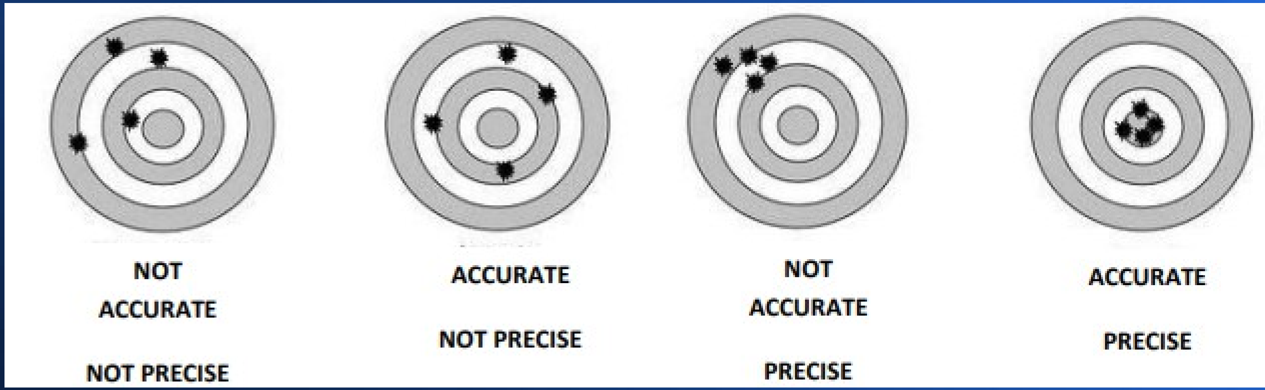
- What is the standard error in the mean?
- The standard error in the mean is the value that tells us how much error we have in our measurements against the mean value. To do this, we need to take the following steps:
- Calculate the mean of all measurements.
- Subtract the mean from each measured value and square the results.
- Add up all subtracted values.
- Divide the result by the √ of the total number of measurements taken.

Errors and Uncertainties

- PRECISION AND ACCURACY:
- Accuracy is the closeness of agreement between a measured value and a true or accepted value (measurement error reveals the amount of inaccuracy).
- Precision is a measure of the degree of consistency and agreement among independent measurements of the same quantity (also the reliability or reproducibility of the result).
- A voltmeter which gives readings of 10, 10, 10, 10 and 10 volts on five measurements of a known voltage of 10 volts is both precise and accurate;
- A meter registering 8,8,8,8, and 8 volts on five measurements at a known voltage of 10 volts is precise, but not accurate
- A meter which reads 11, 10, 8, 9 and 12 volts on five measurements of a known source of 10 volts is neither precise nor accurate.

Errors and Uncertainties

- For example, think of playing darts:



Errors and Uncertainties

- RANDOM AND SYSTEMATIC ERRORS
- There are 2 types of errors in measured data. It is important to understand which you are dealing with, and how to handle them.
- **RANDOM ERRORS** refer to random fluctuations in the measured data due to:
- The readability of the instrument
- The effects of something changing in the surroundings between measurements
- The observer being less than perfect (yes that's you!)
- Random errors can be reduced by averaging. A precise experiment has small random error.
- **SYSTEMATIC ERRORS** refer to reproducible fluctuations consistently in the same direction due to:
- An instrument being wrongly calibrated
- An instrument with zero error (it does not read zero when it should – to correct for this, the value should be subtracted from every reading)
- The observer being less than perfect in the same way during each measurement. Ξ
- Systematic errors cannot be detected or reduced by taking more measurements. Even an accurate experiment has small systematic error.
- When graphing experimental data, you can see immediately if you are dealing with random or systematic errors (if you can compare with theoretical or expected results).

Errors and Uncertainties

- REPORTING A SINGLE MEASUREMENT
- You would be surprised at how few people actually know how to take a proper reading of something!
- Most people try to report a measured value with a degree of certainty that is too generous – expressing more certainty in a reported value than really exists. You should avoid this! It is bad practice.
- Generally we report the measured value of something with the decimal place or precision going not beyond the smallest graduation (called the ‘least count’) on the instrument. In cases where the least counts are wide enough to estimate beyond them with certainty, you may do so. It is ultimately up to the experimenter to determine how to report a measured value, but be conservative and do not overestimate the precision of the instrument.
- Sometimes you hear that uncertainties should generally be reported as ½ the least count; this is technically correct. But since they should be reported with the same number of decimal places as the instrument, in practice this amounts to stating them as ± the least count.

Errors and Uncertainties

- *REPORTING YOUR BEST ESTIMATE OF A MEASUREMENT*
- The best way to come up with a good measurement of something is to take several measurements and average them all together. Each individual measurement has uncertainty, but the reported uncertainty in your average value is different than the uncertainty in your instrument. You do not use the instrument uncertainty in your final stated uncertainty – the precision of the instrument is not the same as the uncertainty in the measurement.
- If you take several measurements of something, you will get a range of values. The ‘real’ value should be within this range, and the uncertainty is determined by dividing the range of values by two. Always round your stated uncertainty up to match the number of decimal places of your measurement, if necessary.
- Your stated uncertainty should have only one significant figure if possible. In a physics laboratory, you should take 3 to 5 measurements of everything. Five is always best if you can manage it!

Errors and Uncertainties

- As an example:
- Six students measure the resistance of a lamp. Their answers in Ω are: 609; 666; 639; 661; 654; 628. What should the students reports as the resistance of the lamp?
- Average resistance = 643 Ω
- Largest - smallest resistance 666 - 609 = 57 Ω
- Dividing the range by 2 = 29 Ω
- So, the resistance of the lamp should be reported as: 643 ± 30 Ω

Errors and Uncertainties

- When taking several measurements, it should be clear if you have a value with a large error. Do not be afraid to discard any measurement that is clearly a mistake. You should never be penalised for this if you explain your rationale for doing so. In fact, it is permissible, if you have many measurements, to throw out the maximum and minimum values.
- When taking time measurements, the stated uncertainty cannot be unreasonably small – I would definitely say not smaller than 0.3 s, no matter what the range. When taking time measurements (such as the period of a pendulum), we can improve the accuracy of our data by measuring the time taken for 20 oscillations for example (20T). In this case, you can divide the uncertainty for 20T by 20 to get the uncertainty in T.
- For a pendulum, 20 oscillations (20T) are timed (in seconds) at 14.73; 14.69; 14.75 . What is T? The range in values is 0.06, but you cannot report the value for 20T as 14.72 ± 0.03 s. You must report it as 14.72 ± 0.30 s but T = 0.7360 ± 0.0150 s or better stated as T = 0.7360 ± 0.0200 s.

Errors and Uncertainties

- Uncertainty in calculated results: absolute and percentage uncertainties
- Absolute uncertainties are expressed as ± the number of units in the measurement (± Δx).
- Length = 234 ± 2 mm Period = 1.6 ± 0.3 s
- This tells you immediately the maximum and minimum experimental values of a measurement.
- Absolute uncertainties have the same units as the stated measurement. All uncertainties begin as an absolute uncertainty, stated according to the uncertainty in the precision of the instrument.
- Percentage uncertainties are expressed as ± [the fractional uncertainty in the measurement x 100] (± [(Δx/x)100]%).
- Length = 234 ± 2 mm or 234 ± (2/234)x100 = 234 (± 8.5 %) mm
- Period = 1.6 ± 0.3 s or 1.6 ± (0.3/1.6)x100 = 1.6 (± 18.8 %) mm
- Percentage uncertainties are unitless and can save lots of time when making calculations, even though it seems cumbersome to express uncertainty this way.

Errors and Uncertainties

- It is good form to leave all final calculated answers with an absolute uncertainty. Therefore, you need to be able to convert from absolute uncertainties to percentage and back again. Constants such as π do not affect the uncertainty calculation.
- When doing calculations involving percentage uncertainties, it is easier to leave out the (x 100) step and simply multiply using the decimal form.
- Uncertainties when making graphs:
- In many cases, the best way to present and analyse data is to make a graph. A graph is a visual representation of 2 things and shows nicely how they are related. A graph is the visual display of quantitative information and allows us to recognise trends in data. Graphs also let you display uncertainties nicely.

Errors and Uncertainties

- Making Graphs – please note
- You need to be able to make graphs by hand in the laboritory **(and always on graph paper please!)**, even though in many cases in the writing up you will be using computer software to create graphs using spreadsheets of data.
- It is generally best where possible to use specialist software for graphing purposes. Excel, for example, produces graphs which are of adequate accuracy for business use (for which it is chiefly intended) but it is not best suited to scientific use. For example, it may employ inadequate numerical or statistical techniques, its shortcomings may be poorly documented if at all, and in particular its plotting of logarithemic graphs leaves a lot to be desired (the points are not always the correct distance apart on a logarithmic scale).

Errors and Uncertainties

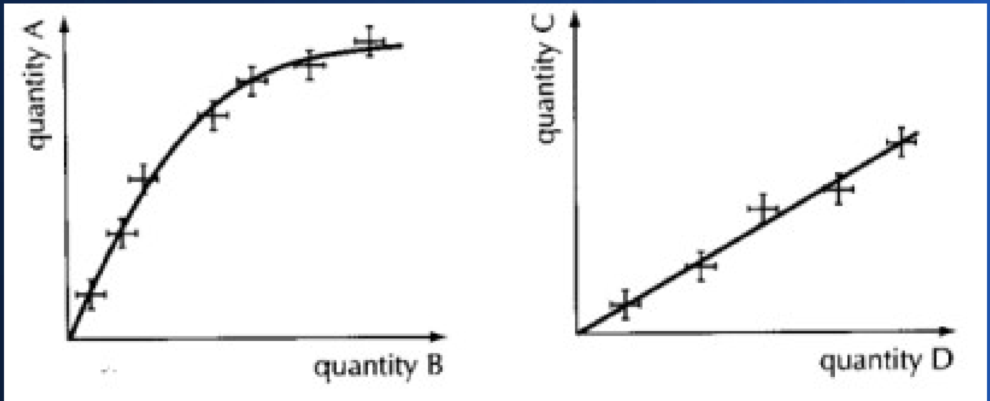
- Good practice when making graphs:
- 1. The independent variable is on the x-axis and the dependent variable is on the y-axis.
- 2. Every graph should have a title that this concise but descriptive, in the form ‘Graph of (dependent variable) vs. (independent variable)’.
- 3. The scales of the axes should suit the data ranges.
- 4. The axes should be labeled with the variable, units, and uncertainties.
- 5. Ample paper area should be used.
- 6. The data points should be clear.
- 7. Error bars should be shown correctly (using a straight-edge)
- 8. Data points should not be connected dot-to-dot fashion. A line of best fit should be drawn instead.
- 9. Each point that does not fit with the best fit line should be identified.
- 10. Think about whether the origin should be included in your graph (what is the physical significance of that point?)

Errors and Uncertainties

- A nice way to show uncertainty in data is with error bars. These are bars in the x and y directions around each data point that show immediately how big or small the uncertainty is for that value. Uncertainties can be constant values for every data point or percentage values (in which case the length will vary).
- Either way, an error box is created when there are error bars in both x and y directions around a data point. It is usually a rectangle and often varies in size around every point.

Errors and Uncertainties

- The line of best fit (which can be a curve OR a straight line) represents the trend shown on a graph. If you are doing this by hand, it is an estimation based on what the trend appears to be. For example, the data might suggest a linear relationship – if so, your job is to draw a line (with a straightedge) that goes through as many data points as possible.
- Approximately the same number of data points should be above your line as below it. That said, I would automatically be suspicious of every data point lay neatly on the line of best fit.



Exam Revision Techniques

- Take some time to understand your learning style
- When it comes to finding the best revision techniques for students, it all begins with understanding how you learn best, e.g. what your learning style is. There are lots of different learning styles out there, with many turning to the VARK theory to understand their preferred learning style. In essence, the VARK theory identifies us as being one of the following learners: visual, aural, read (or write), or kinaesthetic – take the test below to find out which type of student you are!
- Once you know the method of learning that suits you best, simply tailor each of your revision sessions by choosing the techniques that will make remembering the information much easier for you. You'll find that your revision becomes far easier, engaging, and effective on the whole.
- https://vark-learn.com/the-vark-questionnaire/

Exam Revision Techniques

- Organise your notes ahead of time
- To ensure you can kick-start your revision in the most efficient way possible, it's a good idea to (if they aren't already) organise, label, and clearly order your subject notes so that they are easy to read through and use as part of the revision process.
- When you sit down on your first few days of revision, the last thing you want to have to do is waste time finding and filing all your class notes together for you to then begin your revision. Taking the time outside of class to condense and organise your notes into a formulated system will have endless benefits, both at helping you to reconfirm your understanding of the content after class, but also making your revision far more manageable.

Exam Revision Techniques

- Use mind maps to connect ideas
- When it comes to your revision, do you find yourself struggling with remembering lots of new information? Or understanding how different topics relate to each other? Well, mind maps may be key to helping you succeed!
- In essence, the theory behind using mind maps is that making associations between related ideas can help us to memorise information quicker and faster – making it a very effective revision technique.
- Mind maps begin with one central theme or topic. From here, you can then create branches from this central idea with other related ideas that you want to develop or visualise. From these branches, you can add further detail and information, with keywords helping you to summarise information, include key terminology, and visually connect ideas between one another.
- Having a topic summarised into a mind map on one big sheet of A3 paper can be hugely beneficial to information retention, especially if you also use visual aids to help summarise processes or definitions.

Exam Revision Techniques

- Complete as many past papers as possible!
- Another highly effective revision technique to help you prepare for your exams is to get familiar with past papers. After all, there’s no point learning all that content if you don’t know how to apply it to the exams.
- Past papers can be great at helping you become familiar with the format of exams, including the different types of question styles and time restraints. Then, when it comes to the real thing, you’ll know exactly what to expect.
- But aside from this, completing past pacers can also be a good way to test your current understanding of a subject and identify any gaps of knowledge or areas that you’re struggling with.

Exam Revision Techniques

- Lastly, mix your study habits up to keep it engaging
- For some ideas on how to keep your revision engaging, try using one or some of the following techniques:
- Watch video demonstrations or documentaries
- Listen to podcasts
- Organise a group study session
- Mix your study time between at-home and at a library or local café
- Write about your topic as if you were telling a story
- Try teaching a topic to a friend or family member who has little to no knowledge of it
- And finally, do some revision with other members of the class!

Important notice!
重要通知！

When I taught the previous Engineering course in May, the results were delayed. When you take the exam, please ensure that you clearly mark your English name, Chinese name and student number on the exam paper.

This will expedite marking (and hence results!) for all.

Many thanks.

当我在五月份教授之前的工程课程时，结果延迟了。 参加考试时，请确保在试卷上清楚地标明自己的英文姓名、中文姓名和学号。

这将加快所有人的标记（以及结果！）。

非常感谢。

New information

- Any new announcements which I become aware of during the progress of the course will be published here.

New information