Module

8

AC Theory

Filters & Wave Shaping

Passive Filters & Wave Shaping

What you'll learn in Module 8.

Module 8 Introduction

Recognise passive filters with reference to their response curves.

- High pass, Low pass, Band pass, Band stop.

Section 8.1 Differentiators.

Recognise typical filter circuits.

- RC, LC and LR filters.
- Uses for passive filters

Recognise packaged filters.

- Ceramic filters, SAW filter, Three−wire encapsulated filters.

Section 8.2 How Filters Work

Passive filters, frequency selective attenuation, phase change with reference to phasor diagrams.

- High pass and Low pass filters.

Section 8.3 Bode Plots

Bode Plots, the use of Bode plots to describe:

- Attenuation.
- Phase Change

Section 8.4 Differentiators

The use of RC filters in waveshaping on non−sinusoidal waveforms.

- Differentiation.

Section 8.5 Integrators

The use of RC filters in waveshaping on non−sinusoidal waveforms.

- Integration.

Section 8.6 Filter Quiz

Introduction to Passive Filters

Passive filters, often consisting of only two or three components, are used to reduce (ATTENUATE) the amplitude of signals. They are frequency selective, so they can reduce the signal amplitude at some frequencies, without affecting others. Filter circuits are named to show which frequencies they affect.

Fig 8.0.1 shows the symbols used in block (system) diagrams for some filters, and beside them a diagram representing the frequency response of that filter. The block diagrams indicate the frequency that is attenuated by showing three sine waves with one or two crossed out, the vertical position of the wave indicating high medium or low frequencies.

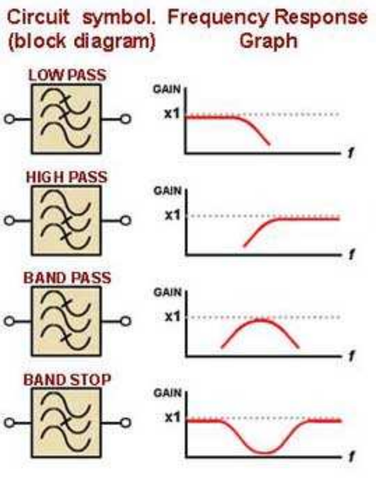


Fig 8.0.1 Filters

To indicate the effect a filter has on wave amplitude at different frequencies, a frequency response graph is used. This graph plots gain (on the vertical axis) against frequency, and shows the relative output levels over a band of different frequencies.

Passive filters only contain components such as resistors, capacitors, and inductors. This means that, the signal amplitude at a filter output cannot be larger than the input. The maximum gain on any of the frequency response graphs is therefore slightly less than 1.

The main difference between passive filters and active filters (apart from the active filter's ability to amplify signals) is that active filters can produce much steeper cut off slopes. However, passive filters do not require any external power supply and are adequate for a great many uses.

Module 8.1 Passive Filters

Uses for passive filters.

Filters are widely used to give circuits such as amplifiers, oscillators and power supply circuits the required frequency characteristic. Some examples are given below. They use combinations of R, L and C

As described in Module 6, Inductors and Capacitors react to changes in frequency in opposite ways. Looking at the circuits for low pass filters, both the LR and CR combinations shown have a similar effect, but notice how the positions of L and C change place compared with R to achieve the same result. The reasons for this, and how these circuits work will be explained in Section 8.2 of this module.

Low Pass Filters

Low pass filters are used to remove or attenuate the higher frequencies in circuits such as audio amplifiers; they give the required frequency response to the amplifier circuit. The frequency at which the low pass filter starts to reduce the amplitude of a signal can be made adjustable. This technique can be used in an audio amplifier as a "TONE" or "TREBLE CUT" control. LR low pass filters and CR high pass filters are also used in speaker systems to route appropriate bands of frequencies to different designs of speakers (i.e. ´ Woofers´ for low frequency, and ´Tweeters´ for high frequency reproduction). In this application the combination of high and low pass filters is called a "crossover filter".

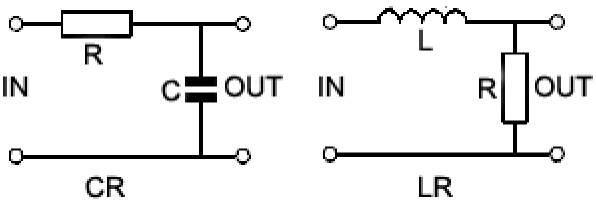


Fig. 8.1 1 Low pass filters.

Both CR and LC Low pass filters that remove practically ALL frequencies above just a few Hz are used in power supply circuits, where only DC (zero Hz) is required at the output.

High Pass Filters

High pass filters are used to remove or attenuate the lower frequencies in amplifiers, especially audio amplifiers where it may be called a "BASS CUT" circuit. In some cases this also may be made adjustable.

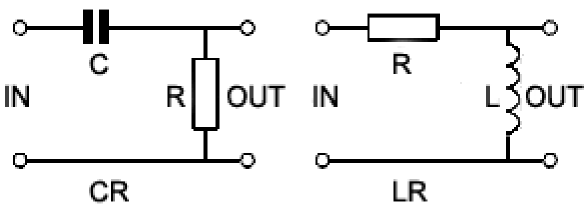


Fig. 8.1.2 High pass filters.

Band pass filters.

Band pass filters allow only a required band of frequencies to pass, while rejecting signals at all frequencies above and below this band. This particular design is called a T filter because of the way the components are drawn in a schematic diagram. The T filter consists of three elements, two series−connected LC circuits between input and output, which form a low impedance path to signals of the required frequency, but have a high impedance to all other frequencies.

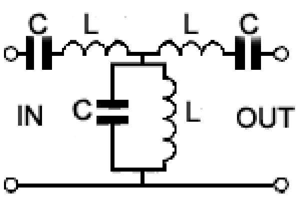


Fig.8.1.3 Band pass filters.

Additionally, a parallel LC circuit is connected between the signal path (at the junction of the two series circuits) and ground to form a high impedance at the required frequency, and a low impedance at all others. Because this basic design forms only one stage of filtering it is also called a ´first order´ filter. Although it can have a reasonably narrow pass band, if sharper cut off is required, a second filter may be added at the output of the first filter, to form a ´second order´ filter.

Band stop filters.

These filters have the opposite effect to band pass filters, there are two parallel LC circuits in the signal path to form a high impedance at the unwanted signal frequency, and a series circuit forming a low impedance path to ground at the same frequency, to add to the rejection. Band stop filters may be found (often in combination with band pass filters) in the intermediate frequency (IF) amplifiers of older radio and TV receivers, where they help produce the frequency response curves of quite complex shapes needed for the correct reception of both sound and picture signals. Combinations of band stop and band pass filters, as well as tuned transformers in these circuits, require careful frequency adjustment.

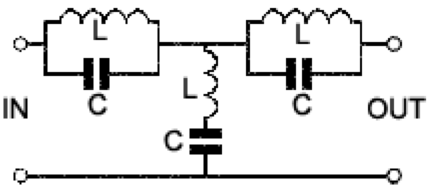


Fig.8.1.4 Band stop filters.

I.F. Transformers.

These are small transformers, found in older in radio and TV equipment to pass a band of radio frequencies from one stage of the intermediate frequency (IF) amplifiers, to the next. They have an adjustable core of compressed iron dust (Ferrite). The core is screwed into, or out of the windings forming a variable inductor.

This variable inductor, together with a fixed capacitor ´tunes´ the transformer to the correct frequency. In older TV receivers a number of individually tuned IF transformers and adjustable filter circuits were used to obtain a special shape of pass band to pass both the sound and vision signals. This practice has largely been replaced in modern receivers by packaged filters and SAW Filters.

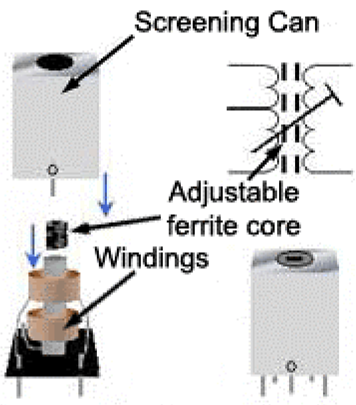


Fig.8.1.5 I.F. Transformer

Packaged Filters.

There are thousands of filters listed in component catalogues, some using combinations of L C and R, but many making use of ceramic and crystal piezo-electric materials. These produce an a.c. electric voltage when they are mechanically vibrated, and they also vibrate when an a.c. voltage is applied to them. They are manufactured to resonate (vibrate) only at one particular, and very accurately controlled frequency and are used in applications such as band pass and band stop filters where a very narrow pass band is required. Similar designs (crystal resonators) are used in oscillators to control the frequency they produce, with great accuracy. One packaged filter in TV receivers can replace several conventional IF transformers and LC filters. Because they require no adjustment, the manufacture of RF (radio frequency) products such as radio, TV, mobile phones etc. is simplified and consequently lower in price. Sometimes however, packaged filters will be found to have an accompanying LC filter to reject frequencies at harmonics of their design frequency, which ceramic and crystal filters may fail to eliminate.

SAW Filters

Fig.8.1.6 SAW Filters.

The illustration (right) shows a Surface Acoustic Wave (SAW) IF (intermediate frequency) filter ON A CIRCUIT BOARD FROM A PAL TV. SAW filters can be manufactured to either a very narrow pass band, or a very wide band with a complex (pass and stop) response to several different frequencies. They can produce several different signals of specific amplitudes at their output. Special TV types replace several LC tuned filters in both analogue and digital TVs with a single filter. They work by creating acoustic waves on the surface of a crystal or tantalum substrate, produced by a pattern of electrodes arranged as parallel lines on the surface of the chip.

The waves created by one set of transducers is sensed by another set of transducers designed to accept certain wavelengths and reject others. Saw filters may be found in many types of electronic equipment and have response curves tailored to the requirements of specific types of product, including communications devices, auotmotive and industrial applications, where they are used for selecting or rejecting particular bands of frequencies.

Typical Saw Filters by Epcos

Ceramic Filters

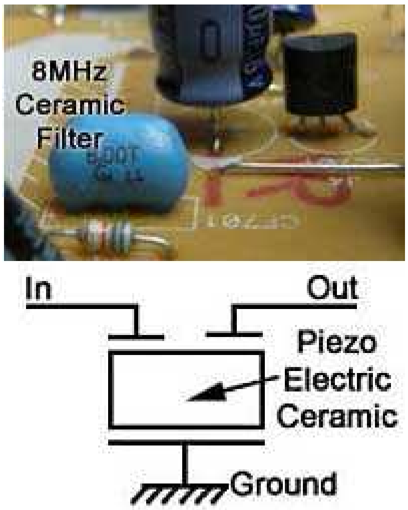


Fig.8.1.6 Ceramic Filter & its Circuit Symbol

Ceramic filters are available in a number of specific frequencies, and use a tiny block of piezo electric ceramic material that will mechanically vibrate when an a.c. signal of the correct frequency is applied to an input transducer attached to the block. This vibration is converted back into an electrical signal by an output transducer, so only signals of a limited range around the natural resonating frequency of the piezo electric block will pass through the filter. Ceramic filters tend to be cheaper, more robust and more accurate than traditional LC filters for applications at radio frequencies. They are supplied in different forms including surface mount types, and the encapsulated three pin package shown here.

Module 8.2 How Filters Work.

CR Filter Operation.

Figs 8.2.1 and 8.2.2 show two common methods of using C and R together to achieve alterations in AC signals. These CR combinations are used for many purposes in a wide variety of circuits. This section describes their effects when used as filters with sine wave signals of varying frequencies. The same circuits are also used to change the shape of non-sinusoidal waves and this topic "Differentiation and Integration" is described in Section 8.4 and 8.5 of this module.

The High Pass CR Filter

The CR circuit illustrated in Fig 8.2.1, when used with sinusoidal signals is called the HIGH PASS FILTER. Its purpose is to allow high frequency sine waves to pass unhindered from its input to its output, but to reduce the amplitude of, (to attenuate) lower frequency signals. A typical application of this circuit would be the correction of frequency response (tone correction) in an audio amplifier.

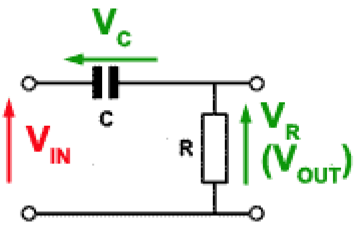


Fig 8.2.1 High Pass CR Filter

As described in Module 6, resistance is constant at any frequency, but the opposition to current flow offered by the capacitor (C) however, is due to capacitive reactance X_C, which is greater at low frequencies than at high frequencies.

The reactance of the capacitor (X_C) and the resistance of the resistor (R) in fig 8.2.1 act as a potential divider placed across the input, with the output signal taken from the centre of the two components. At low frequencies where X_C is much greater than R, the share of the signal voltage across R will be less than that across C and so the output will be attenuated. At higher frequencies, it is arranged, by suitable choice of component values, that the resistance of R will be much greater than the (now low) reactance X_C, so the majority of the signal is developed across R, and little or no attenuation will occur.

The Low Pass CR Filter

In Fig 8.2.2 the positions of the resistor and capacitor are reversed, so that at low frequencies the high reactance offered by the capacitor allows all, or almost all of the input signal to be developed as an output voltage across X_C. At higher frequencies however, X_C becomes much less than R and little of the input signal is now developed across X_C. The circuit therefore attenuates the higher frequencies applied to the input and acts as a LOW PASS FILTER.

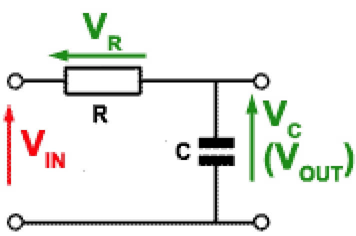


Fig 8.2.2 Low Pass CR Filter

The band of frequencies attenuated by high and low pass filters depends on the values of the components. The frequency at which attenuation begins or ends can be selected by suitable component choices. In cases of audio tone correction, the resistor may be made variable, allowing a variable amount of bass or treble (low or high frequency) cut. This is the basis of most inexpensive tone controls

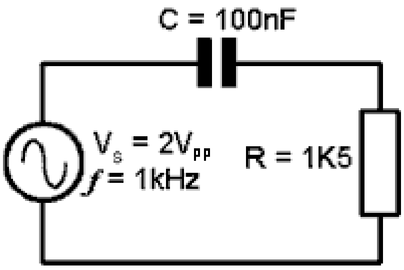
High and low pass filters can also be constructed from L and R. In this case the action is the same as for the CR circuit except that the action of X_L is the reverse of X_C. Therefore in LR filters the position of the components is reversed.

Phase Change in Filters

The above description of high and low pass filters explains how they operate in terms of resistance and reactance. It shows how gain (Vout/Vin) is different at high and low frequencies due to the relative values of X_C and R. However this simple explanation does not take the phase relationships between capacitors or inductors, and resistors into account. To accurately calculate voltage values across the components of a filter it is necessary to take phase angles into consideration as well as resistance and reactance. This can be done by using phasor diagrams to calculate the values graphically, or by a branch of algebra using ´complex numbers´ and ´j Notation´. However these calculations can also be done using little more than the Reactance calculations learned in Module 6 and the Impedance Triangle calculations from Module 7.

Problem:

Calculate the peak to peak voltages V_r appearing across R and V_c appearing across C when an AC supply voltage of 2Vpp at 1kHz is applied to the circuit as shown.



Although C and R form a potential divider across V_s it is not possible to calculate these values using the potential divider equation(because phase angles must also be taken into account):

$$V_r = V_s \frac{R}{X_c + R}$$

Follow these steps:

1. Find the value of capacitive reactance X_C using:

$$X_C = \frac{1}{2\pi f C}$$

$$X_C = \frac{1}{2\pi f C} = \frac{1}{2\pi \times 1\text{x}10^3 \times 100\text{x}10^{-9}} = 1591.5\Omega$$

2. Use the Impedance Triangle to find Z (the impedance of the whole circuit).

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

$$Z = \sqrt{(R^2 + X_C^2)} \equiv Z = \sqrt{(1500^2 + 1591.5^2)} \equiv 2187\Omega$$

3. Knowing that the supply voltage V_s is developed across Z, the next step is to calculate the volts per ohm (V/Ω),

$$\frac{V_s}{Z} = \frac{2}{2187} = 914.5\mu V/\Omega$$

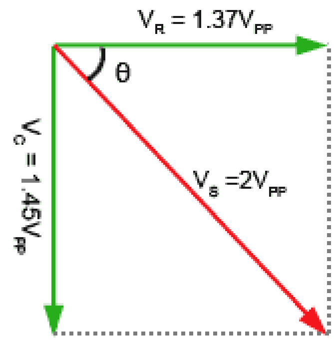
Because the volts per ohm will be the same for each component as it is for the circuit impedance, the result from step 3 can now be used to find the voltages across C, and across R.

$$V_R = 1500 \times 914.5\mu V/\Omega = 1.37V$$

$$V_c = 1591.5 \times 914.5\mu V/\Omega = 1.45V$$

If required, the Phase angle θ could also be found using trigonometry as described in Phasor Calculations, Module 5.4 (Method 3). To find the angle θ (the phase difference between the supply voltage V_S and the supply current, which would be in the same phase as V_R) the two voltages already found could be used.

$$\text{Angle } \theta = \tan^{-1}\frac{\text{Opposite}}{\text{Adjacent}} = \tan^{-1}\frac{V_C}{V_R} = \tan^{-1}\frac{1.45}{1.37} = 46.6°$$



Because circuits containing capacitance (or inductance) in addition to resistance affect the phase relationships of sine wave signals. This allows these same circuits to be used to change the PHASE of signals instead of, or as well as the amplitude.

CR Filter Operation

Fig 8.2.3 and Fig. 8.2.4 on pages 7 and 8 demonstrate how phasor diagrams can explain both the amplitude and phase effects of a CR filters. Adjust the frequency slider and notice that it is the input voltage that apparently changes phase, but this is just because the circuit current phasor (and theefore the V_R phasor, which is always in phase with the current) is used as the static reference phasor. The thing to remember is that there is a phase change of between 0° and 90° happening between V_{IN} and V_{OUT}, which depends on the frequency of the signal.

Using phasor diagrams to explain the high pass filter shows that:

• In The High pass Filter (Fig. 8.2.3) at low frequencies the output V_{OUT} (V_R) is much smaller than V_{IN} (V_C) and a phase shift of up to 90° occurs with the output phase leading the input phase.

• At high frequencies there is little or no difference between the relative amplitudes of V_{OUT} (V_R) and V_{IN} , and little or no phase shift is taking place. At the corner frequency the phase shift is 45° and below that the Bode plot shows that frequency gain falls off at a steady rate of -20dB per decade.

Fig 8.2.3

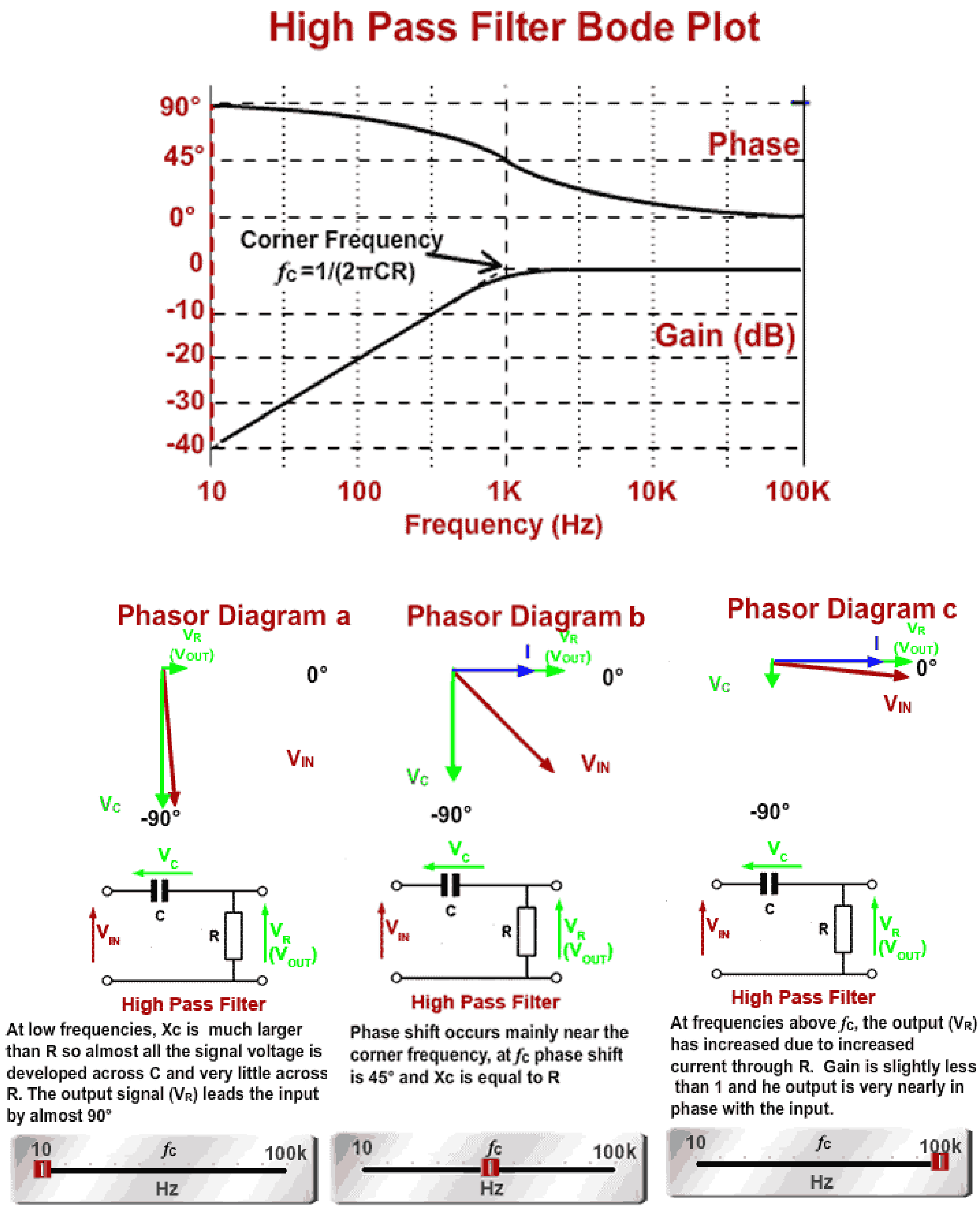
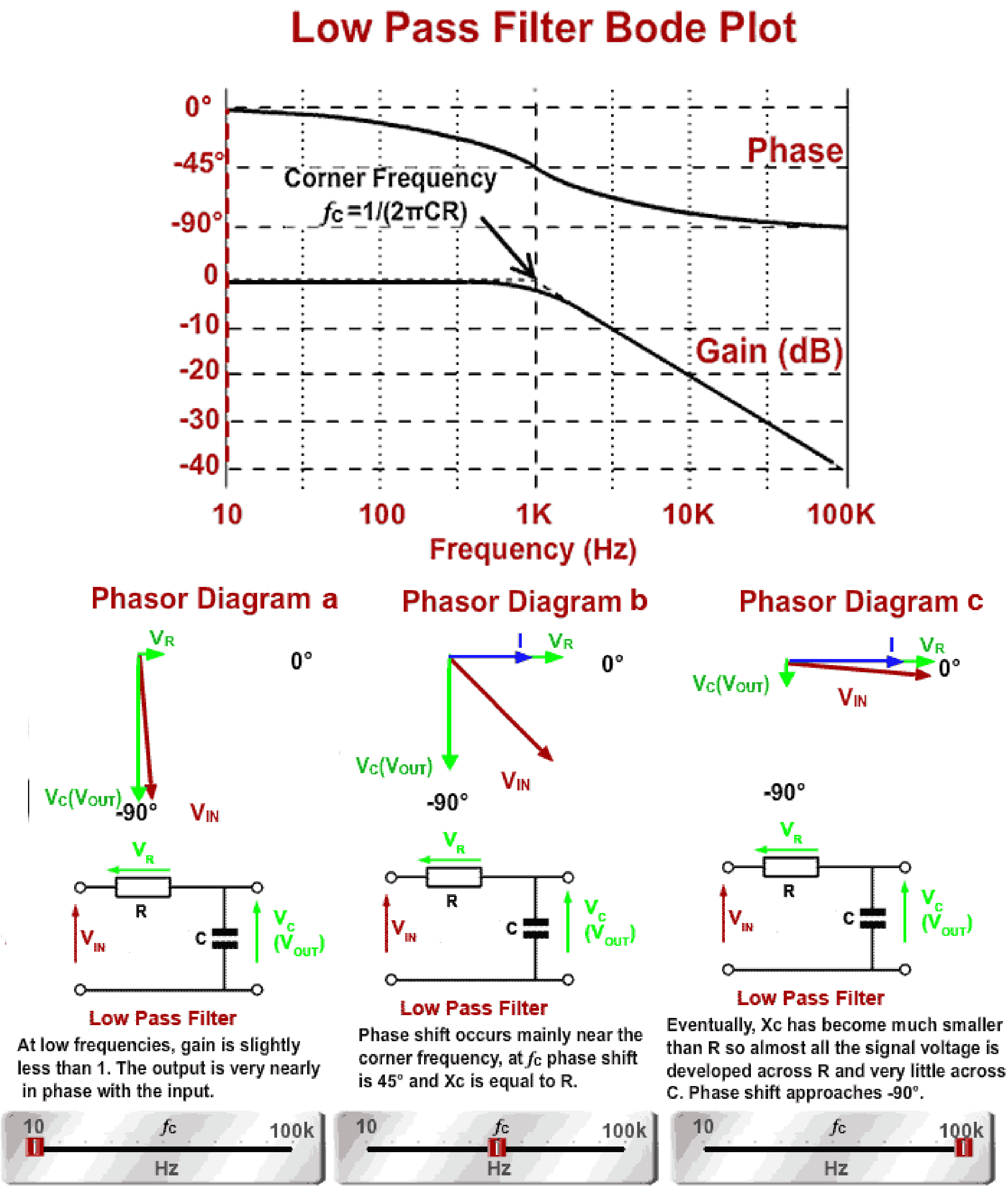

High Pass Filter Bode Plot
90°
45°
0°
Phase
Corner Frequency
fc=1/(2πCR)
0
-10
-20
-30
-40
Gain (dB)
10
100
1K
10K
100K
Frequency (Hz)
Phasor Diagram a
VR (VOUT)
0°
VIN
VC
-90°
Phasor Diagram b
I
VR (VOUT)
0°
VC
VIN
-90°
Phasor Diagram c
VR (VOUT)
I
0°
VC
VIN
-90°
High Pass Filter
At low frequencies, Xc is much larger than R so almost all the signal voltage is developed across C and very little across R. The output signal (VR) leads the input by almost 90°
High Pass Filter
Phase shift occurs mainly near the corner frequency, at fc phase shift is 45° and Xc is equal to R
High Pass Filter
At frequencies above fc, the output (VR) has increased due to increased current through R. Gain is slightly less than 1 and he output is very nearly in phase with the input.
10
fc
100k
Hz

Fig 8.2.4 similarly demonstrates the action of a CR Low Pass Filter. In this circuit note that $V_{OUT} = V_C$ so the output phase lags the input phase by up to about -90°, depending on the input frequency

Fig 8.2.4



Module 8.3 Bode Plots

Showing Phase Shift and Attenuation

When considering the operation of filters, the two most important characteristics are:

- The FREQUENCY RESPONSE, which illustrates those frequencies that will, and will not be attenuated.
- The PHASE SHIFT created by the filter over its operating range of frequencies.

Bode Plots show both of these characteristics on a shared frequency scale making a comparison between the gain of the filter and the phase shift simple and accurate.

Frequency is plotted on the horizontal axis using a logarithmic scale, on which every equal division represents ten times the frequency scale of the previous division, this allows for a much wider range of frequency to be displayed on the graph than would be possible using a simple linear scale. Because the frequency scale increases in "Decades" (multiples of x10) it is also a convenient way to show the slope of the gain graph, which can be said to fall at 20dB per decade.

The vertical axis of the gain graph is marked off in equal divisions, but uses a logarithmic unit, the decibel (dB) to show the gain, which with simple passive filters is always unity (1) or less. The dB units therefore have negative values indicating that the output of the filter is always less than the input, (a gain of less than 1). The upper section of the vertical axis is plotted in degrees of phase change, varying between 0 and 90° or sometimes between −90° and +90°

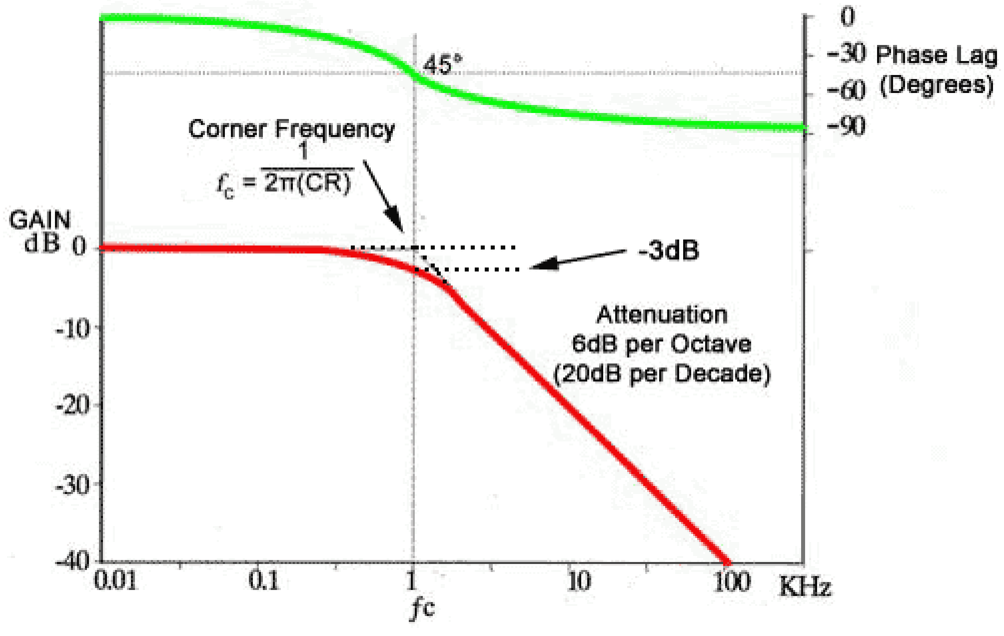


Fig 8.3.1 Bode Plot for a Low Pass Filter.

A Bode plot for a low pass filter is shown in Fig 8.3.1. Note the point called the corner frequency. This is the approximate point at which the filter becomes effective. Frequencies below this point are unaffected by the filter, while above the corner frequency, attenuation of the signal increases at a constant rate of 6dB per octave. This means that the signal output voltage is halved (−6dB) for each doubling (an octave) of the input frequency.

Alternatively the same fall off in gain may be labelled as −20dB per decade, which means that voltage gain falls by ten times (to 1/10 of its previous value) for every decade (tenfold) increase in frequency. The fall off in gain of a filter is quite linear beginning from the corner frequency (also called the cut off frequency). I.e. if the voltage gain of a low pass filter is 1 at a frequency of 1kHz, it will be 0.1 at 10kHz. The linear fall off in gain is common to both high and low pass filters, it is just the direction of the fall, increasing or decreasing with frequency, that is different.

The corner (or cut off) frequency (f_C) is where the active part of the gain plot begins, and the gain has fallen by −3dB. The phase lag of the output signal in a low pass filter (or phase lead in a high pass filter) is at 45°, exactly half way between its two possible extremes of 0° and 90° The corner frequency may be calculated for any two values of C and R using the formula:

$$f_C = \frac{1}{(2\pi CR)}$$

For LR filters the formula is:

$$f_C = \frac{1}{(2\pi L/R)}$$

Note that the corner frequency is that point where two straight lines representing the two sections of the graph either side of f_C would intersect. The actual curve makes a smooth transition between the horizontal and sloping sections of the graph and the gain of the filter is therefore -3dB at f_C

Module 8.4 Differentiators

A simple RC network such as the high pass filter illustrated in Fig. 8.4.1, when used with non sinusoidal waves produces a change in wave shape at its output. With a sine wave input, only the amplitude and phase of the wave change. However, if the input wave is a complex wave, such as a square or triangular wave, the effect of these simple circuits appears to be quite different.

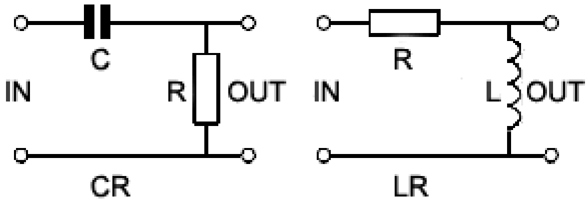


Fig.8.4.1 The Differentiator Circuit

Differentiation

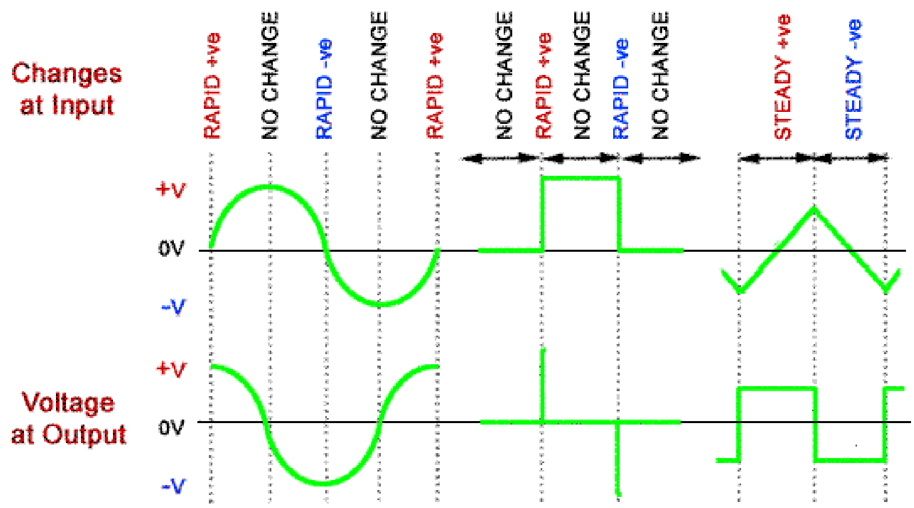


Fig 8.4.2 Differentiation.

The circuit is called a DIFFERENTIATOR because its effect is very similar to the mathematical function of differentiation, which means (mathematically) finding a value that depends on the RATE OF CHANGE of some quantity. The output wave of a DIFFERENTIATOR CIRCUIT is ideally a graph of the rate of change of the voltage at its input. Fig. 8.4.2 shows how the output of a differentiator relates to the rate of change of its input, and that actually the actions of the high pass filter and the differentiator are the same.

Because the differentiator output is effectively a graph showing the rate of change of the input, whenever the input is changing rapidly, a large voltage is produced at the output. The polarity of the output voltage depends on whether the input is changing in a positive or a negative DIRECTION.

Sine Waves

A graph of the rate of change of a sine wave is another sine wave that has undergone a 90° phase shift (with the output wave leading the input wave).

Square Waves

The square wave input and output in Fig 8.4.2 shows the ideal differentiator action of a high pass filter. The output wave is now nothing like the input wave, but consists of narrow positive and negative spikes. The positive spike coincides in time with the rising edge of the input square wave.

The negative spike of the output wave coincides with the falling, or negative going (towards zero volts) edge of the square wave. Notice that the DC level of the wave is also changed by the differentiator. The output wave now has both positive and negative half cycles above and below a centre line of zero volts, due to the dc blocking effect of the capacitor.

Triangular Waves

A triangular wave has a steady positive going rate of change as the input voltage rises, so produces a steady positive voltage at the output. As the input voltage falls at a steady rate of change, a steady negative voltage appears at the output. The graph of the rate of change of a triangular wave is therefore a square wave. Wave shaping using a simple high pass filter or differentiator is a very widely used technique, used in many different electronic circuits.

Practical Differentiation

Although the ideal situation is shown in Fig. 8.4.2, how closely the output resembles perfect differentiation depends on the frequency (and therefore periodic time) of the input wave and the time constant of the components used, as shown in Fig. 8.4.3. The high pass filter works as a differentiator when the input is:

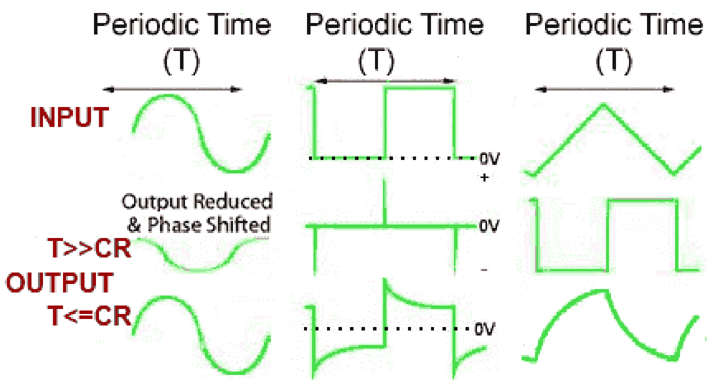


Fig 8.4.3 Practical Differentiation.

a. A non-sinusoidal wave.

b. The time constant(T) of the input wave is much greater (longer duration) than the time constant(CR) of the circuit (T>>CR), i.e. at relatively low frequencies.

When T is less than or equal to CR (T<=CR) the output wave shape will be less than an ideally differentiated wave shape, being more or less like the waveforms shown in the bottom row of Fig. 8.4.3.

Although passive (with no amplification) differentiators are cheap and efficient, where it is necessary to control the amplitude of the output, active differentiators using op-amps, as described in Amplifiers Module 6.6 are often used.

Module 8.5 Integrators

The Integrator Circuit

Integration is used extensively in electronics to convert square waves into triangular waveforms; in doing this it has the opposite effect to differentiation (described in Filters & Wave shaping Module 8.4). The shape of the input wave of an integrator circuit in this case will be a graph of the rate of change of the output wave as can be seen by comparing the square wave input and output waveforms in Fig. 8.5.2. Notice that the integrator circuit (shown in Fig. 8.5.1) is that of the CR low pass filter described in Filters & Wave shaping Module 8.2.

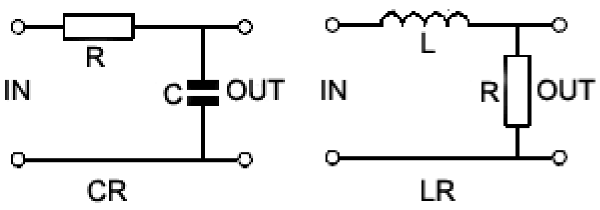


Fig. 8.5.1 The Integrator (also low-pass filter) Circuit.

Integrator Action with a Sine Wave Input

If the input is a sine wave, the circuit does not act as an integrator, but as a simple low pass filter (LPF) where the amplitude of the output wave is reduced, and its phase relative to the input wave is shifted so that it lags by up to -90° dependant on the frequency of the wave and the CR time constant of the circuit, as described in Filters & Wave shaping Module 8.2

The low pass filter circuit is therefore called an integrator only when:

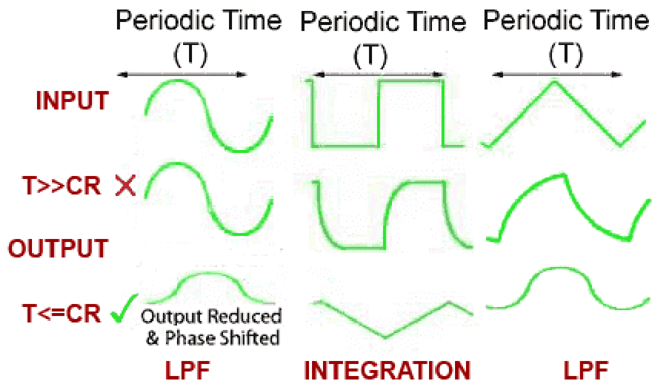


Fig. 8.5.2 Integrator Action

a. The input wave is a square wave.

b. The periodic time(T) of the input wave is much shorter than the circuit time constant(CR) i.e.(T<=CR).

Provided that these conditions are met, then the action of the integrator is opposite to that of the differentiator circuit described in Filters and Wave shaping Module 8.4.

Integration of a Square Wave

With a square wave input and the correct relationship between the periodic time of the wave and the time constant of the circuit, Fig 8.5.2 shows that integration takes place. The output is now (considering the waveforms as simple graphs), a graph of the changing area beneath the input wave. The integrator has converted the square wave input to a triangular wave at the output, the slope of this wave describes the increase in area beneath the square wave (moving from left to right). For the circuit to act effectively as an integrator, the periodic time of the wave must be similar to, or shorter than the circuit time constant i.e. (T<=CR). The higher the frequency of the input wave for a particular time constant, the better the shape of the output wave will be, but the smaller its amplitude. Also notice that, unlike the differentiator, the integrator does not block any DC component of the input wave. Therefore the reduced amplitude output wave will have a DC component, which (ignoring the resistance of any load placed on the output) will be the same as the average DC level of the input wave.

At lower frequencies, where the periodic time T of the wave is much longer than the time constant of the circuit CR (T>>CR), some change in wave shape does occur, but the output does not conform to the definition of an integrator; the circuit has just rounded the rapid vertical transitions of the square wave. The output at these low frequencies is not a graph of the changing area beneath the input wave, the circuit is acting as a low pass filter and removing the high frequency components of the square wave that were responsible for the rapid vertical changes at each half cycle.

Action on a Triangular Wave

When the input to the integrator circuit is a triangular wave, the output seems to become a sine wave. Remember however, that the integrator circuit is also a low pass filter that has the effect of removing the higher frequency harmonics present in the complex (triangular) wave at its input, leaving just the fundamental (sine wave) and possibly a few lower frequency harmonics. At low frequencies, the output from the integrator circuit is therefore a rounded form of the triangular input wave.

The main purpose of a passive CR integrator is to produce a good triangular wave shape from a square wave input, which it can do very well and at very low cost (only two components are needed) although the output will be reduced in amplitude. Any lack of amplitude may be overcome by combining the passive CR circuits described in this module with an op-amp to create an active filter, differentiator or integrator as described in Amplifiers Module 6.6.

Module 8.6 Filter Quiz

What you should know.

After studying Module 8, you should:

Be able to recognise typical filter circuits.

Be able to describe how passive filters work, and relate this to phasor diagrams.

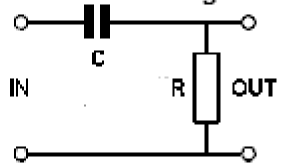
Be able to describe Bode Plots and their uses.

Be able to describe the use of RC filters in wave shaping.

1.

Refer to Fig 8.6.1. What is this circuit called when used with sinusoidal signals?

Fig 8.6.1



a) A high pass filter.

b) A differentiator.

c) A low pass filter.

d) An integrator.

2.

With reference to Fig 8.6.2, which of the formulae would be used to find the corner frequency of a low pass filter?

Fig 8.6.2

$$f_c = \frac{1}{2\pi CL} \quad \text{a} \qquad f_c = \frac{1}{2\pi CR} \quad \text{b}$$

$$f_c = 2\pi CL^2 \quad \text{c} \qquad f_c = 2\pi CR^2 \quad \text{d}$$

a) Formula a b) Formula b c) Formula c d) Formula d

3.

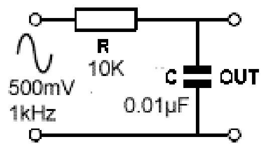
Which of the following labels would most appropriately describe a High pass filter when used in an audio amplifier?

a) Bass boost b) Bass cut c) Treble boost d) Treble cut

4.

With reference to Fig 8.6.3 what would be the approximate amplitude of the signal at the output?

Fig 8.6.3



a) 1V b) 500mV c) 250mV d) 125mV

Continued

5.

Which of the following describes the circuit in Fig 8.6.4?

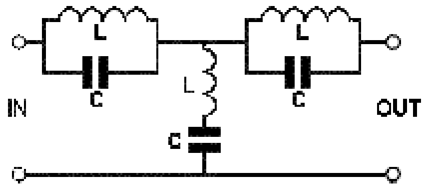


Fig 8.6.4

a) Band stop filter

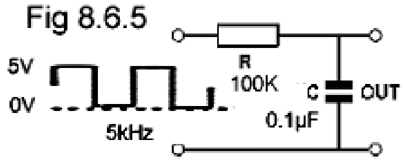
b) Band pass filter

c) High pass filter

d) Low pass filter

6.

What will be the waveform at the output of Fig 8.6.5?

Fig 8.6.5


a) A rounded square wave

b) Differentiated pulses

c) A triangular wave

d) A parabolic wave

7.

A square wave with a periodic time of 10µs is applied to the input of a differentiator circuit. For differentiated pulses to appear at the output, the time constant of the CR network should be approximately:

a) 1µs b) 2.5µs c) 5µs d) 10µs

8.

Which of the following networks can be used as a differentiator?

a) Notch filter b) High pass filter c) Band pass filter d) Band stop filter

9.

With reference to Fig 8.6.5, if a DC voltmeter is connected across the output terminals of the circuit with the input shown, what will be the voltmeter reading?

a) 5V b) 2.5V c) 1.25V d) 0V

10.

With reference to Fig 8.6.6, if a triangular wave having a long time constant is applied to the input, what waveform would be expected at the output?

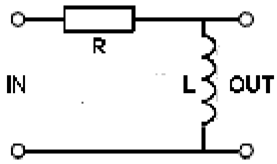


Fig 8.6.6

a) A Square wave

b) A triangular wave

c) Differentiated pulses

d) A sine wave