ELECTRONICS —it's easy! PART 21

All about electronic filters

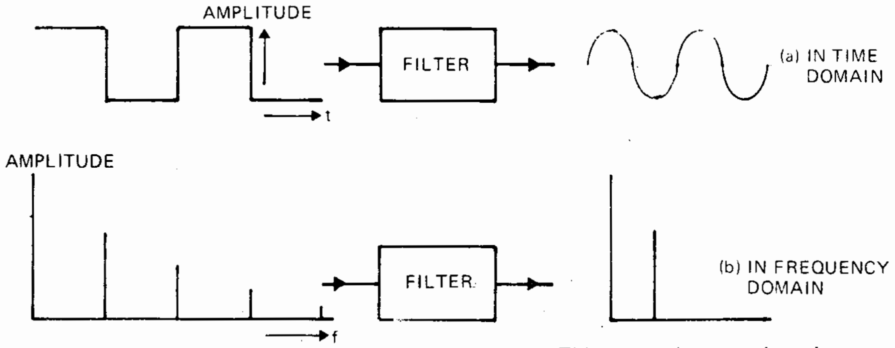


Fig. 1(a). A filter alters the frequency content of a signal. This means the wave shape is changed when displayed as an amplitude-time graph. (b). Using the frequency spectrum form of display the filter removes (or enhances) certain frequencies.

TO SEPARATE peas from boiling water, or dirt from engine oil, one must use an appropriate filter. When the term filter is used, in any discipline, the meaning is always the same — it is a device for separating or selecting something from an available mixture or range of things.

Filters are also extensively used in electronics where they are used to select a desired part of the range of frequencies which make up a particular signal. We have seen many examples of this throughout our course so far. For instance, in our discussion of multiplexed telephone systems, we saw how it is necessary to *separate* the various frequency channels and pass them to individual outlets. We also saw how an LC tuned circuit is used to *select* only one desired radio broadcast station from the many available.

Other examples of the use of filters are the crossover networks used in hi-fi speaker systems, to divide the audio bandwidth between two or more speaker drive units, the compensation stages in instrumentation control systems which improve performance by attenuating or enhancing relevant frequencies or the filters used to correct for the non-linear attenuation versus frequency which occurs with long-line telephone communications.

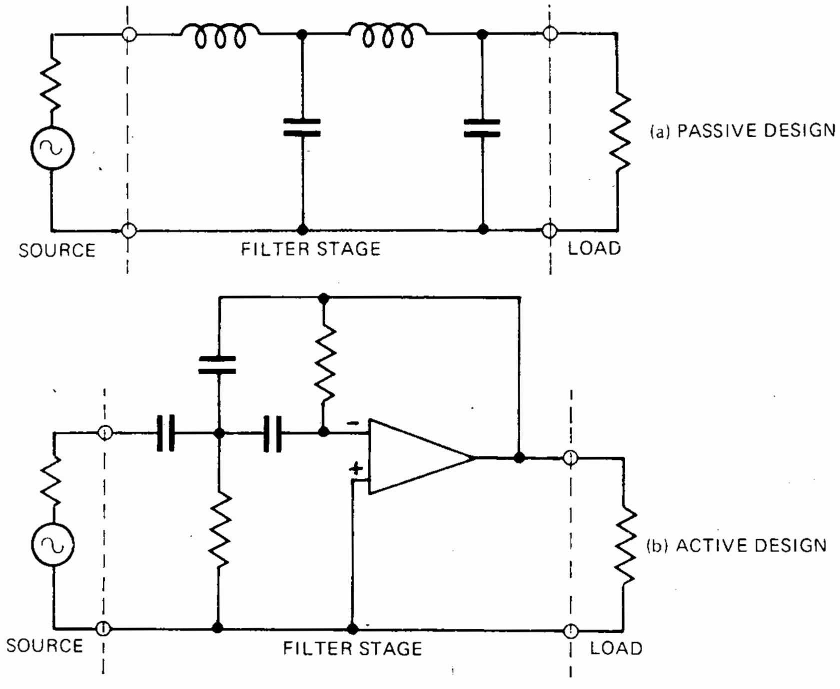


Fig. 2(a). Passive filters use R, L. and C components only. (b). Active filters incorporate active elements with passive elements to great effect.

ALTERING THE FREQUENCY RESPONSE

Electronic filters, in a general sense then, alter the frequency content of signals. Their action can be comprehended first by considering the stage as a unit that alters the amplitude/time shape of an input waveform. This concept is illustrated in Fig. 1a where a square-wave is filtered to remove all but its fundamental sine wave. Alternatively, filters may be thought of as devices that change the frequency spectrum. This is illustrated in Fig. 1b. Both concepts are correct, each finding use to suit different needs.

We generally think of filters as devices which change the amplitude of the signal with frequency. However, filters may also change the phase of the signal. In many applications the phase shift is undesirable and must be considered when making the selection of filter type.

Unlike other circuit blocks which are available as built up units, filters are generally made specifically for the task.

Many filters are extremely simple — varying from two components to (say) ten and the design procedures of most are easily found in texts. This is not, however, to say that filters are trivial and not worth learning about. Filter designs may be grouped into two main classes — those called passive filters (Fig. 2a) that use passive components only — such as resistors, capacitors and inductors; and those called active filters (Fig. 2b) that are based upon an op-amp using single or multiple path feedback loops. Design procedures can be quite complicated but because of the universal need for a few basic types of response, most design is now a matter of applying simple formulae or using graphs to arrive at the component values.

By way of interest the design philosophy of filters — or any network requiring a given frequency response — can proceed two ways. First, one can propose a network configuration and then mathematically analyse it to get the generalised formula. This is called network analysis. The alternative and more modern approach (in the last few decades, that is) is to start with a mathematical expression of the

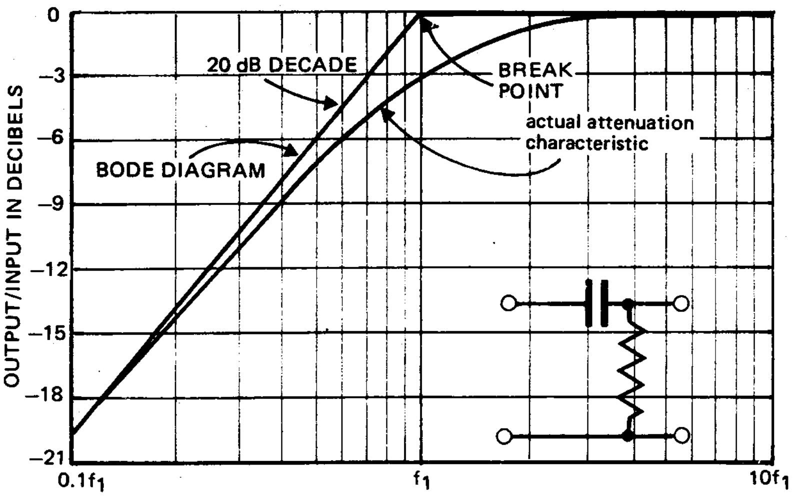


Fig. 3. Bode diagrams usually express amplitude (and phase) variation with frequency in terms of simplified responses consisting of straight lines turning at break points. The actual response will be more gradual near the breakpoint.

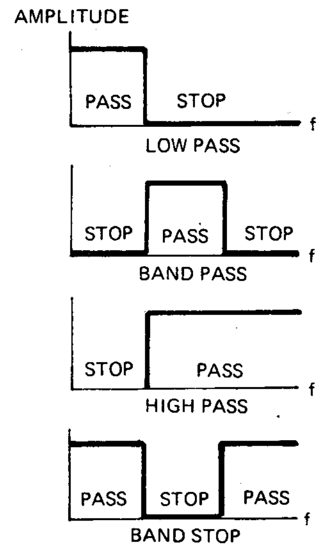


Fig. 4. Idealised responses of various categories of filter.

frequency response needed and, by using appropriate mathematical procedures, create on paper the circuit needed to provide such a response. This is called circuit synthesis. The latter method has a certain fascination because it provides the answer in a more logically direct manner than the cut and dried analysis process (although sometimes one ends up with a requirement for non-realisable circuit needs such as negative frequency!). On the other hand, however, synthesis requires mathematical ability and considerable experience.

In the following sections we will analyse a few of the more common filter stages.

THE BODE DIAGRAM

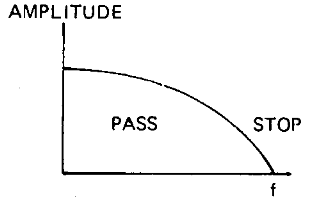
One of the, now classical, works on network analysis is a book "Network Analysis and Feedback Amplifier Design" by H. W. Bode published by Van Nostrand in 1945. Today Bode's work is mostly remembered by the graph which carries his name and relates the amplitude, or phase shift, to frequency for an amplifier, feedback system or a frequency modifying stage such as a filter. There is, at least, in principle, no distinction between the frequency response plots we have discussed to date and the Bode diagram. In practice, however, Bode diagrams are usually mathematical simplifications in that they are drawn with straight lines only, these lines changing direction at what are known as break-points and sloping at known rates.

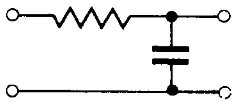
The Bode diagram exemplifies the behaviour of a circuit as a tool, and is derived from mathematical knowledge of the system, not from actual tests. In truth, the linearization simplification is usually not far from reality, and we will meet Bode diagrams in our study of filters. Fig. 3 shows the difference between a Bode diagram and an actual response plot for an RC filter. The Bode diagram plots signal amplitude in decibels on a linear scale against frequency on a logarithmic scale.

TYPES OF RESPONSE

As with amplifiers, filter frequency responses are grouped into low-pass, band-pass and high-pass. Theoretically, ideal filters would have responses as shown in Fig. 4. There is also a constant need in electronic systems for a band-stop stage.

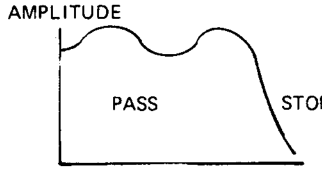
In reality it is impossible to obtain exactly square response curves. The response always rises or falls, within the transition region, with a rate of steepness that depends on the design used. A general rule is that the simpler the design (least number of components) the more gradual will be the transition. Also the more rapid the transition the more likely are effects of "ringing" encountered. Do not confuse these concepts of shape with amplitude-time wave shape graphs: these are amplitude (phase) — frequency curves. To illustrate this concept compare the two extremes given in Fig. 5. Figure 5a is for a most basic RC stage, Fig. 5b is for a response having rapid cutoff — a Chebyshev filter stage.





(a) SIMPLE R.C. LOW PASS FILTER STAGE

Fig. 5. As a general rule the more complex the filter circuit, the sharper the roll-off but the more variable the response in the passband region. (a) RC low pass stage. (b) Advanced Chebyshev stage.



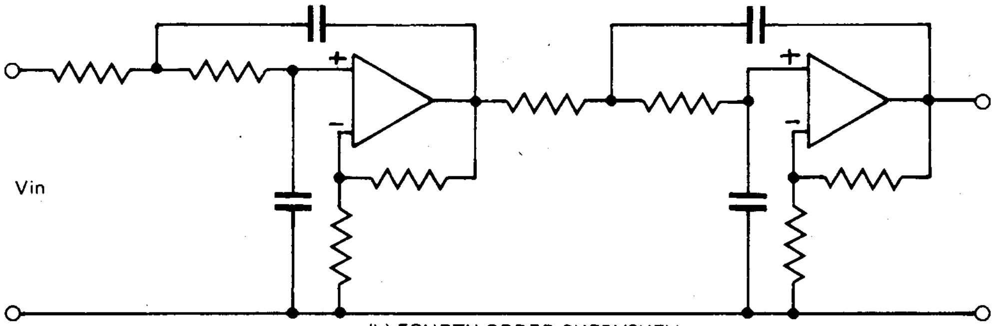


FIG. 5

(b) FOURTH ORDER CHEBYSHEV LOW PASS FILTER STAGE

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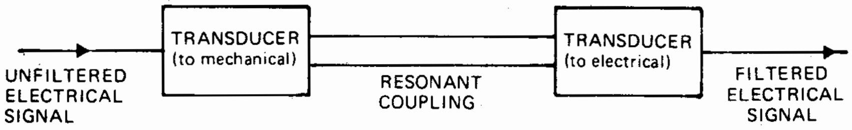

Fig. 6. By converting the electronic signals to mechano-acoustic form it is possible to make use of the extreme sharpness of mechanical resonant systems.

It is also worth noting that no filter is perfect, for frequencies are only attenuated relative to each other. If a signal appears at a high enough level at the input of a filter stage it will appear at a reduced level in the output and could be troublesome. Acknowledging this, the degree of attenuation chosen should be matched to the circumstances expected. It is pointless (and unnecessarily expensive) designing a stage to provide, say, 120 dB reduction of the unwanted frequency if it never reaches more than, say, 10 dB of the wanted frequency, apart from which an unwanted signal which is more than 60 dB down on the wanted one rarely causes problems.

DEFINING THE RESPONSE BANDWIDTH

As realistic filters fall short of being ideal there is no clear-cut point, where the response changes markedly enough, to use as the criteria for defining bandwidth. In some simple filters we could use the apparent position of the breakpoint but this would not hold for all filters.

The convention used is that the cut-off point is defined as where the response power falls to one-half of the passband value. Half power, expressed as a voltage change, is 0.707 of the passband voltage level which is −3 dB in decibel units. (Often called the '3 dB down' point.)

The bandwidth of bandpass (or bandstop) filters is, therefore, the frequency interval between the two cut-off points situated on each side of the bandpass (or stop) region. Bandwidth of a high-pass design has no real meaning as the frequency rises to infinity. Low-pass units have a band-width from zero frequency (dc) to the cut-off value.

In the case of complex designs the stated response often omits what happens at frequencies remote from the usual frequencies of interest. It is wise never to assume that, say, a bandpass filter only passes frequencies between the design points. It may well have "windows" much removed from that region. Additional stages are added in some system designs to exclude these effects.

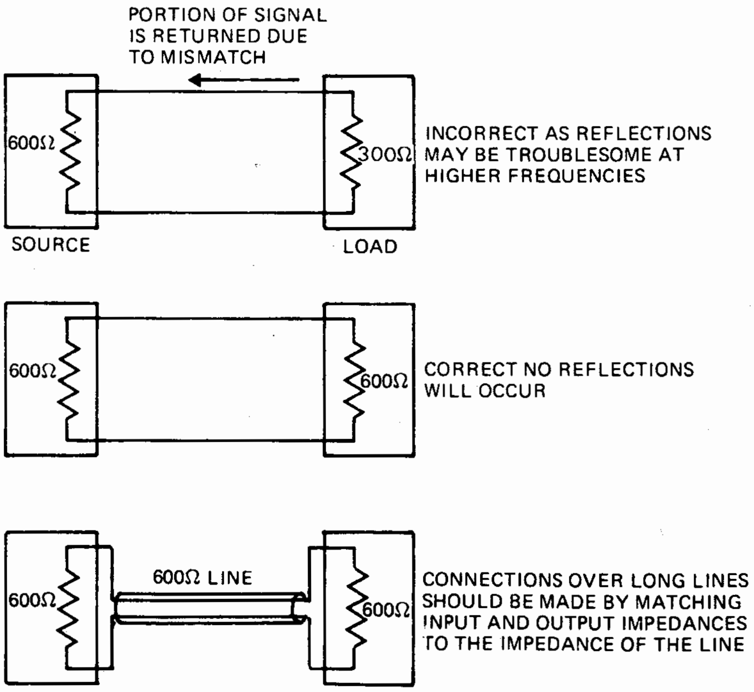

Fig. 7. Reflection of RF energy will arise if stages are not terminated into each other with the same impedance. Filter stages should observe this requirement.

Whereas the majority of filters used in electronic systems are made solely from electronic components there do exist circumstances where transduction to mechanical principles for filtering, and back again to electrical, are advantageous. One example is the use of tuned resonant reed filters, such as is depicted in Fig, 6, which exhibit extremely narrow band-pass characteristics.

Often the response of a bandpass is expressed in terms of its quality factor – that is the Q-factor of the peak. This definition was discussed when we dealt with resonant circuits earlier in the course.

THE EFFECT OF ADDING A FILTER

When the main purpose of adding a filter is to alter the frequency composition of signals it is not unexpected that the other effects brought about by its insertion might be overlooked.

As in any system changed by the addition of a cascaded 'box', the output of the preceding stage and the input of that following must be considered from the loading point of view. It is quite unrealistic to design a stage in isolation, unless the filter stage is adequately buffered, for the impedances connected to its input and output will alter the cut-off points – and hence different values will be required to achieve the designed characteristic.

The term 'Insertion Ratio' will often be encountered, it describes the ratio of output voltage with and without the filter, that is, the voltage Insertion

$$\text{Ratio} = \frac{\text{Vout (no filter)}}{\text{Vout (with filter)}}$$

Expressed in decibels of loss we arrive at the term Insertion Loss = 20 $\log_{10}$ (Voltage Insertion Ratio). In practical cases, however, one may well design a stage to provide insertion gain (especially in active filter stages).

When matching a filter into a system it may be important to conserve power, voltage or current. To ensure maximized power transfer the input impedance to the filter must be of the same value as the output impedance of the stage before. Similarly, its output must be terminated into the same value. If voltage levels are to be maximized then the filter input impedance must be much higher than the output impedance of the driving stage. Current maximization requires the reverse relationship.

When the frequency of operation is high another problem becomes significant – that of reflections. When energy is launched into a network containing storage elements – a filter stage is such – some of the energy may be returned to the source which, in turn, may reflect it again, the final situation being that the net sum of all of these travelling waves of energy cause excessive power losses in the line (and distortion). This effect is very pronounced in radio-frequency transmission lines.

The extent to which a reflection occurs is decided by the degree of difference in the impedances seen in both directions at a system block junction. If a filter is terminated into the source with the same impedance in both directions there is no mismatch and no reflection occurs. This concept is depicted in Fig. 7.

As the two impedances differ in magnitude so does the amount of signal reflected. A similar situation applies at the output of the filter.

Mismatch terminations begin to generate noticeable spurious signals this way from megahertz frequencies upwards. This is the reason why wide-bandwidth amplifiers, such as videoamps, must be designed with output impedances that match the feeder cable. Coaxial cable can be shown to have a characteristic impedance set by the ratio of size and spacing of its conductors. It is invariant with length of cable. Typical coaxial cables have impedances of 50 or 75 ohm. Alternatively another kind of cable having two wires with a fixed separation between them may be used. Such transmissions lines have typical impedances of 200, 300 or 600 ohms. Whilst on this subject, one way of locating open-circuit and short circuit faults in cables is to send a sonic pulse (these travel much slower than electromagnetic waves) down the cable — timing the arrival of reflected pulses produced by the gross mismatch that exists at the fault.

Filter stages, as said before, also introduce phase shifts. A sine-wave input will appear at the output shifted in time by some fraction of the electrical cycle. In the compensation networks of feedback controllers phase shift must be carefully controlled, for a wrong value of phase shift may cause the system to become unstable. That is, if the phase shift approaches 180°, the feedback becomes positive, instead of negative, and the system oscillates.

PASSIVE DESIGNS

THE RC FILTER

The simplest passive electronic filter is the RC network set to act as a low-pass or high-pass stage. The two alternatives are shown in Fig. 8. In Fig. 8a it is easy to see that at low frequencies the capacitive reactance is very high and the output is the same as the input, provided the load

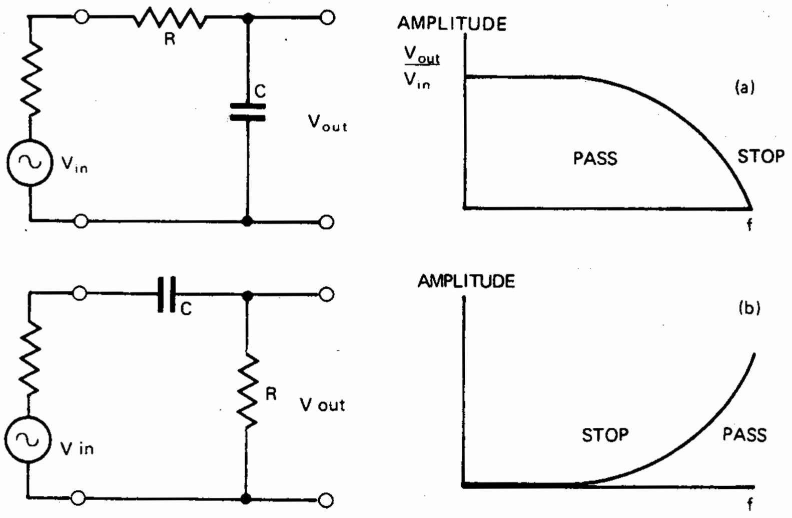

Fig. 8. Basic RC filter stages (a) low-pass (b) high-pass.

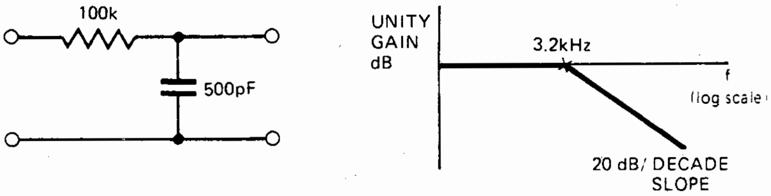

Fig. 9. Bode diagram for low-pass RC filter in which source and load are not significant.

impedance connected is significantly higher than the value R. As the frequency rises X_c decreases, lowering the output voltage. The reverse situation applies for the high-pass unit.

Mathematical analysis shows that the response plot — the Bode diagram — for these can be constructed by recognizing that there is just one break point and that the response falls away at 20 dB/decade change in frequency (ie 6 dB/octave). An octave change corresponds to 2 : 1 frequency ratio; a decade change is a 10 : 1 ratio. The jargon used is that the response rolls-off at the stated rate. Regardless of the values of RC chosen the roll-off rate stays the same. The break point occurs at $f_c = \dfrac{1}{2\pi RC}$.

To illustrate this consider the construction of the Bode diagram for a low-pass filter with R = 100 kilohms and C = 500 pico-farads. The break point occurs at

$$fc = \frac{1}{6.28 \times 100 \times 10^{3} \times 500 \times 10^{-12}}$$

and it slopes downward from there at 20 dB/decade to give the plot shown in Fig. 9.

This much may seem almost trivial and, indeed, it is over-simplified. In practice there will be a source and a load impedance connected to the filter terminals. Fig. 10 shows the practical case in general.

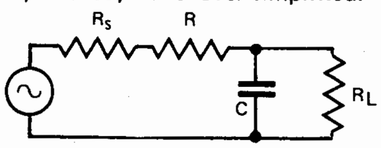

Fig. 10. Practical RC filter designs should allow for source and load resistances.

It is also not hard to reason out what happens when the source and load impedances are taken into account for Rs is in series with R and R_L is in parallel with C. By expanding our mathematics we find that the formula becomes

$$f_c = \frac{1}{2\pi\,[(R_S + R)\,//\,R_L]\,C}$$

Hence, if the stage is not buffered the breakpoint can be quite different from that arrived at from the time-constant of the filter alone. For example if load and source impedances are both 1 k in our previous example the breakpoint changes from 3.2 kHz to 2.66 kHz. Further, the stage will introduce attenuation: the gain in the passband becomes

$$\frac{Vout}{Vin} = \frac{R_L}{R_S + R + R_L}$$

$$\text{for our example} = \frac{1000}{1000 + 5000 + 1000}$$

$$= 0.4$$

By use of appropriate values of source and load resistance it is possible, therefore, to set the attenuation and draw an appropriate Bode diagram.

The high-pass RC filter is considered in the same way — to arrive at

$$f_c = \frac{1}{2\pi\,(R_S + R\,//\,R_L)\,C} \text{ and}$$

$$V_{out} = \frac{R\,//\,R_L}{R_S + R\,//\,R_L}$$

for the practical case where source and load impedances cannot be ignored.

The observant reader will probably have realised that an amplifier stage with capacitive coupling has an equivalent circuit that is a combined highpass and lowpass filter with gain added between. The high-pass response arises from the coupling capacitor and the stage input impedance, the low-pass response from the output impedance and the stray capacitance existing to ground.

It is possible to combine a low-pass RC stage with a high-pass stage to arrive at a bandpass filter. These, however, are not particularly selective bandpass filters because of the relatively poor roll-off slopes (20 dB/decade). Further, if the bandwidth required is small, the two stages interact producing a non-constant passband gain. To obtain a satisfactory design it is important to ensure that the second stage resistance (the shunt of the high-pass stage) is at least ten times that of the first (the series resistance of the low-pass stage). Also the two break points should be at least a decade apart.

RC NOTCH FILTERS

Some applications call for rejection of a narrow band of frequencies, the reduction of 50 Hz or 100 Hz noise, for example. A very effective, yet, inexpensive technique makes use of a type of Wheatstone bridge which requires only resistors and capacitors and yet provides very sharp roll-off.

The Twin-T or parallel-T notch filter is such a circuit and is shown in Fig. 11. (It can be redrawn as a more-obvious bridge circuit and comprises two T circuits connected in parallel). At high or low frequencies it is easy to see that the capacitances either go to low or high reactances providing in both instances a virtually unaltered signal level through the stage. At the balance point, of a twin-T bridge, there exists a frequency — the so-called notch — at which the output falls very nearly to zero. This occurs for the circuit of Fig. 11 at

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

Loading will reduce the depth of the notch.

In some applications it is desirable to be able to tune the notch to varying frequency values. In the Twin-T design this requires that all three resistors (or capacitors) be varied simultaneously. A ganged multi-unit potentiometer or capacitance bank is used.

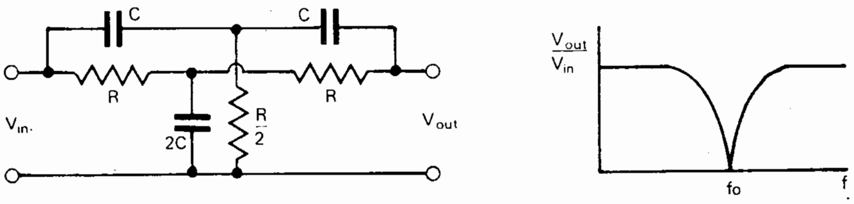

Fig. 11. The Twin-T notch filter provides very narrow rejection of a particular frequency.

Other forms of bridge filter exist, each having its own particular feature. No simple RC circuits exist that exhibit the reverse characteristic of the notch filter — that is spike acceptance of a particular frequency. This reponse however, can be provided by using a notch-filter as the feedback impedance in an op-amp that is set up as a simple inverter. This is shown in Fig.12. In this way the gain of the stage rises rapidly with increase in effective feedback resistance at the notch frequency.

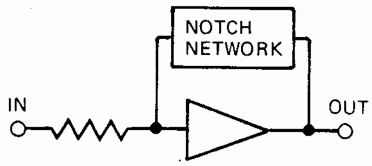

Fig. 12. Notch acceptance can be provided by using a notch-rejection circuit in the feedback of an op-amp.

IMPROVING THE ROLL-OFF

RC filters, apart from notch circuits, cannot provide much selectivity between signals due to their poor 20 dB/decade rolloff. This slope can be improved by cascading stages but this is not a preferred method for there exist other more economical designs.

The next stage of complexity is to use designs combining inductors and capacitors: no resistors are needed. That these provide improved roll-off is to be expected for we have seen earlier in this course that a resonant circuit can provide very sharp responses. By way of example a single stage LC filter can provide at least 12 dB and up to 25 dB/octave rolloff compared with only 6 dB/octave for an RC stage, and furthermore methods have been established (discussed in next part) that enable these to be cascaded without difficulty — a four stage unit can achieve 100 dB/octave rolloff! It is even possible to 'peak up' a specific frequency in the passband. In the next part we will also explain the virtues of adding amplifiers to form active filter circuits. ●