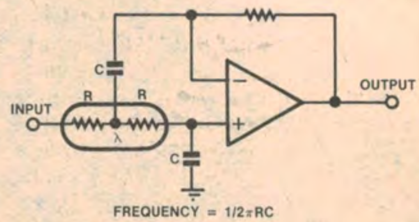
Add-on for improved sound

Dynamic noise reduction filter

This article describes a dynamic noise filter which is very similar in operation to the recently released National Semiconductor dynamic noise reduction IC (DNR). It can be used to reduce the noise content of any audio program source such as a cassette deck, VCR or FM tuner.



FREQUENCY = $1/2\pi RC$

Fig. 2

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There are a number of add on noise reducers available. However, the majority of these rely on encoding and decoding. This means that if the source has not been encoded, then decoding will not give the improvement expected. In order for these units to function correctly an accurate and stable means of control must be exercised over the signal on record and playback.

Dynamic noise filtering relies on two ed technique. That is, it is used on playback only. It can be used to improve the effective signal to noise ratio of tapes, tuner or disc. This means it is a versatile piece of equipment for the serious listener. In addition, it is inexpensive to build and easy to use.

Dynamic noise filter relies on two assumptions for its operation. First, if a program signal contains only low frequencies, a wide bandwidth is unnecessary. Second, if two high frequencies are present, the loudest of the two will tend to mask the other. In other words, a high frequency signal will tend to mask high frequency noise.

Given these two assumptions, the circuit was designed to sense the high frequency content of the music being processed, and to increase the bandwidth as the high frequencies increase. By examining Fig. 1 it can be understood how this is accomplished. Each channel is buffered to give a high input impedance, and a low output impedance to drive the voltage-controlled filter. The VCFs are controlled by varying the resistance of a light dependent resistor via a LED.

The two channels are mixed prior to the filter, and the signal is amplified by three. The resultant of the two channels is attenuated by the sensitivity control and then passed through a high pass filter. This removes the unwanted low frequency content. The filter has a cut-off point at 4kHz.

After removing the low frequencies, the signal is converted to DC by a precision rectifier and filter capacitor. The rectifier also provides the remaining gain of 10. The DC signal is then summed with a bias voltage, which allows the VCF's cut-off frequency to be established in the absence of any signal. The resultant control voltage is converted to current in order to drive the LEDs, and in turn, they control the filter. It can now be seen that only high frequencies will control the filter, and therefore a wide bandwidth will only be provided when it is needed.

We will return to the VCF for a moment as it is the heart of the dynamic noise filter. There are a number of ways of constructing a voltage-controlled low pass filter, but generally you must control one or more resistors or capacitors in order to vary the cut-off frequency.

The filter settled upon is a second-order Butterworth design. This gives an attenuation of minus 12dB per octave after the cut-off frequency, and a flat response in the pass band up to the cut-off point. Fig. 2 shows the VCF. During the investigation into the design, I considered using FETs in the ohmic region, this being a popular way of achieving a voltage-controlled resistor. However, to obtain four (for stereo) FETs that are matched would prove a difficult task for the amateur constructor. In addition to this FETs are inherently non-linear in this mode of operation and can cause excessive distortion unless signal levels are kept low.

The Butterworth filter lends itself well

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Fig. 1

Fig. 1: Block diagram shows how the dynamic noise filter works. The two input channels are separately buffered and passed through voltage controlled filters.

to a centre-tapped LDR. The matching and tracking of the two halves are excellent since they are manufactured together. Linearity is good, so distortion can be kept down to levels which are quite respectable (less than 0.1%).

The resistance verses incident light is essentially linear, which means a doubling of control voltage will double the filter's cut-off frequency. A decrease in resistance occurs very quickly and this helps in giving the unit a fast attack time. The return to high resistance is a little slower (approximately 100 milliseconds), but this is more than adequate for processing music.

The circuit diagram

Refer now to Fig. 3 to consider the complete circuit. IC1a and 1b are non-inverting buffers giving an input resistance of 47kΩ set by the input resistor. The associated coupling capacitor sets the low frequency minus 3dB point at 20Hz. The outputs of the buffers are directly coupled to each VCF and are also summed by IC1c which has a gain of 3.3.

The output of the VCF is coupled to the main output via a 100Ω resistor, and 100µF bipolar capacitor. This removes any DC offset and provides isolation from unhealthy loads. The 10kΩ pot following the summing amplifier provides control over varying signal levels, that will occur from one piece of equipment to another. It allows a variable amount of signal to be passed on to the high pass filter.

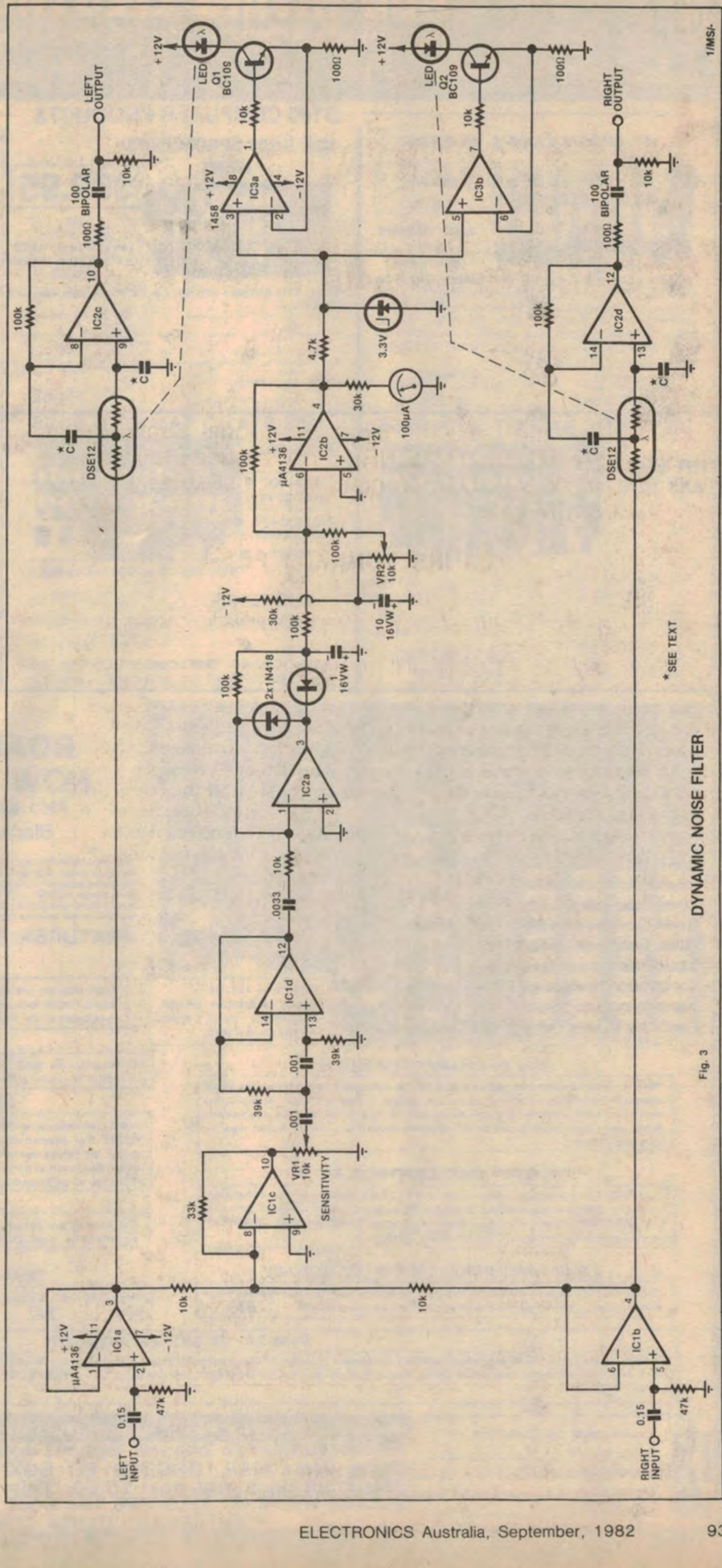
This provides a minus 18dB per octave attenuation after the cut-off point by using a second-order Butterworth filter (12dB/octave) and by selecting the coupling capacitor of the next stage, to give a 3dB point to coincide with the filter.

Following the filter is the half wave rectifier (IC2a) and filter capacitor. The filter capacitor is charged to the peak value of the waveform and this provides a fast attack time. The rectifier also provides the remaining gain of 10. A precision rectifier has the virtue of providing rectification down to very small signal levels by incorporating the diodes within the feedback loop of an op-amp.

The resultant DC is summed with a variable DC voltage in IC2b. The DC voltage is varied by VR2 which forms the bottom leg of a voltage divider and is used to set the VCF cut-off frequency in the absence of any signal.

The sum of the two DC voltages is the control voltage that is used to adjust the

Fig. 3: At right, the complete circuit diagram of the dynamic noise filter. Each voltage controlled filter uses a centre-tapped light dependant resistor.



filter's cut-off point. The meter M1 displays this DC voltage and is calibrated in kilohertz to give a visual display of the filter's instantaneous cut-off frequency.

IC3a, 3b, Q1, Q2, and the associated components, convert the control voltage to current in order to drive the LEDs. This ensures that voltage versus light output will be as linear as possible. The 3.3 volt zener diodes catch the control voltage and are used to limit the LED current to a non-destructive level.

The LEDs are coupled to the LDRs and the filter's cut-off frequency is thus proportional to the control voltage. Fig. 4 shows the filter's cut-off frequency verses voltage for my prototype. It can be seen that it is essentially linear.

Fig. 4

Fig. 4: Filter cut-off frequency vs voltage.

Construction

Any of the normal construction techniques may be used and the final choice will depend on the constructor's past experience. I prefer to use tag strip and wire wrap sockets for "one off" work, as any changes or developments can easily be incorporated. The LED and LDR are held together with "Heatshrink" tube as shown in Fig. 5. I used a hair dryer to shrink the tubing. This holds the assembly together neatly and will help exclude any unwanted light.

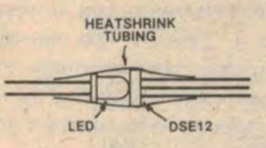


Fig. 5

Fig. 5: Construction of LED/LDR pair.

My prototype was built with wooden ends and aluminium plate as this complemented some existing equipment I already had. There are, however, an abundance of instrument cases available if you wish a good finish without having to spend too much time on the housing.

Calibration

Although I have not tested many LDRs, I believe that there is enough variation between them to necessitate calibration of the unit. I was able to align the filters easily since I possess a CRO. However, many readers will not. It is possible to calibrate the unit by varying one of a few components but the explanation of all these variables would take some time. In order to make calibration simple, the following steps should be followed.

1. Connect the LED to a voltage source via a resistor. The resistor is calculated to allow 20mA to pass through the LED by the formula

$$R = \frac{V - 1.7}{.02}$$

2. With LED illuminated, measure one half of the LDR with an ohmeter, and note the value.

3. Calculate the value of the capacitors to be used in the VCF by the formula

$$C = \frac{1}{2\pi R.\ 20{,}000}$$

where R is the resistance of the LDR.

The process is then repeated for the second channel. With the capacitor value chosen, the filter can be controlled to 20kHz by a control voltage of 2V RMS. The meter is calibrated to read 30kHz FSD by varying the series 30kΩ resistor.

Operation

If the DNF is to be used on tape or tuner, it is connected in series with the output of these. If it is to be used for disc it is connected in the tape loop of your amplifier.

The initial adjustment of the unit is as follows: First set the DC bias control to maximum and the sensitivity to minimum. The music should sound natural with ample high frequencies. Slowly reduce the DC bias until you can hear a considerable reduction in the background hiss.

Now the sensitivity control should be slowly increased and the music should now regain its high frequency content. The operating procedure is given as a guide. Correct adjustment will depend on the type of music being processed. I encourage you to experiment with different settings in order to obtain maximum noise reduction with minimum noticeable side effects.

After using the DNF for some months, I can say that I now have records in my collection that I had previously considered throwing away. I hope you will receive the same benefits and will continue to listen to old favourites that may have otherwise collected dust.