AUDIO TALK

by LEO SIMPSON

Noise Weighting Filters

Elsewhere in the this issue, readers will find an article on the construction of weighting filters to the CCIR and DIN standards. We decided to make these up so that we could confirm the signal-to-noise ratios of equipment which were specified according to one of the above standards. But we did not do this because of a belief that weighted noise measurements are in any way valid. Far from it!

I have always been of the opinion that unweighted noise measurements are the most consistent and fairest method of objectively assessing the signal-to-noise ratio of audio equipment. At the same time I acknowledge that an objective measurement often does not bear a good relationship to the subjective signal-to-noise ratio.

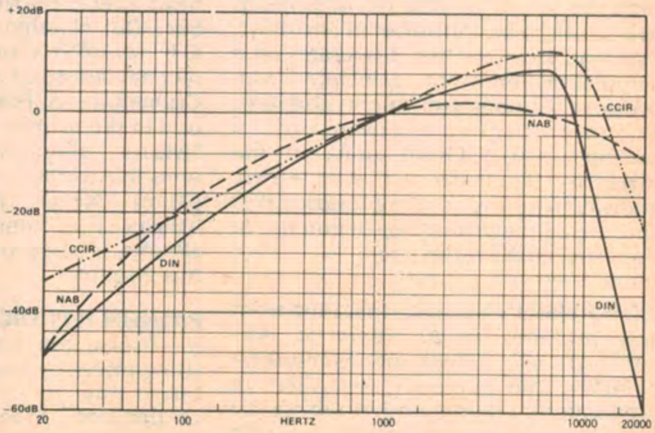
This means that sometimes we have to qualify a signal-to-noise ratio measurement with some subjective comment, eg, the S/N ratio may only have been 65dB but subjectively "the amplifier was quiet at all times" or perhaps "the amplifier was plagued with low level hum" or some other comment which attempts to overcome the shortcomings of the objective measurement.

The thinking behind the concept of weighting filters is similar to that advanced in favour of Loudness controls which we discussed in detail last month. Or more correctly: I preached and, I hope, you accepted. Well, the starting point for this month's homily is the same, the Fletcher and Munson graph of equal loudness contours. An updated version of these contours was featured in last month's column.

I recount part of the discussion on the equal loudness contours: at very intense sound levels, the ear shows a pronounced peak in its response in the region of 3 to 4kHz, of about 15dB. At much lower levels, the peak is less pronounced (to about 8 or 9dB) but the response below about 200Hz is markedly reduced. At these lower levels, the high frequency response is also slightly reduced but some parts of the high frequency response are actually slightly boosted.

Reference to the curves of weighting filters shown on this page will show how the framers of the CCIR and DIN standards have attempted to compensate for the characteristic of the ear at very low sound levels. The DIN curve, for example, shows quite a drastic approach. Noise in the region of 5 to 7kHz is boosted by more than 8dB, while below 1kHz the response tapers at a rate beginning at 6dB/octave but increasing to 12dB/octave at low frequencies. But above 9kHz, the DIN curve plunges at the rate of 48dB/octave. This means that the DIN people have made an assumption which really does not follow from the equal loudness contours.

At right are plotted three oft-used weighting filter curves employed in noise measurements. The NAB curve is less drastic than the CCIR and DIN curves.



The assumption appears to be that nobody can hear noise and hiss above 10kHz. Well, hands up all those who can hear frequencies above 10kHz. Yes, I should think that many of you can.

By contrast, the CCIR curve is less drastic in its attenuation of the high and low frequencies, but boosts the frequencies in the range 5 to 8kHz by 12dB or more.

I regard the amount of boost provided by the CCIR curve as exaggerated and so, apparently, does the American Institute of High Fidelity. When they use this curve for noise measurement, they set the unity gain point at 2kHz rather than 1kHz.

This latter approach makes the CCIR curve more realistic. But I feel the DIN curve has been designed more to produce good signal-to-noise ratio figures from mediocre equipment, rather than to reflect the subjective effect of various noise components.

A good illustration of the unrealistic measurements produced by the CCIR and DIN weighting curves concerns cassette decks with Dolby noise reduction Here, a DIN weighted measurement may produce a result of about 60dB with respect to OVU with Dolby switched out. But switching the Dolby in may give little or no improvement in reading. This is in spite of the quite audible and worthwhile reduction in hiss afforded by the Dolby system.

But there is another reason for my dissatisfaction with these weighting curves, and that is the apparent assumption that all noise originating from audio equipment is random or "white". This is not the case. For example, noise from turntables and cassette decks may be partly cyclic in nature, eg, noise from bearings, belts and motors.

Similarly, noise in amplifiers and tuners may be predominantly hum or "frame buzz". With these non-random types of noise, the ear no longer follows the equal loudness contours (inasmuch as any one person's ears follow those contours). Rather, the ear "tunes" these noises in and filters them out from the surrounding ambient "rubbish".

In fact, most high fidelity enthusiasts have at one time or another, found themselves consciously listening out for these defects, rather than listening to the music. This "fault-seeking" characteristic of the human ear, as far as the appreciation of audio equipment is concerned, makes nonsense of weighting curves.

Hence, I am sticking to my opinion that, for the time being, the unweighted signal-to-noise ratio is still the best objective test. All that seems to me necessary is that it should be qualified, where necessary, by subjective comment.

DIN/CCIR weighting filters

Recently, we decided to produce a set of filters to enable us to perform weighted measurements of signal-to-noise ratios of audio equipment. This article gives brief details of the filters, which give weighting according to the DIN and CCIR standards.

by RON DE JONG

Signal-to-noise ratios of audio equipment can be quoted in two forms, either "weighted" or "unweighted". The term "weight" in this context refers to the method by which noise in the midrange is given emphasis and noise at the upper and lower ends of the audible frequency range are attenuated.

An unweighted signal-to-noise ratio measurement is made by simply using an AC millivoltmeter to measure the residual noise of the equipment under test. The millivoltmeter will have a frequency response which is flat, ie, within say ±0.5dB from 20Hz to 20kHz. The value of the noise voltage measured is then referred to a reference level (say 1 volt RMS) to obtain a signal-to-noise ratio of so many decibels.

The argument which is put forward in favour of weighted signal-to-noise ratios is that the human ear does not have a flat frequency response, and therefore does not hear different types of noise equally. A given level of hiss may sound much louder than the same level of rumble or hum. Noise weighting filters are an attempt to overcome this shortcoming.

We have designed a composite filter to provide weighting characteristics conforming to the DIN and CCIR standards, as these are the most commonly used in measuring high fidelity audio equipment. The filter characteristics are shown in the graphs below.

Both filter characteristics emphasise midrange noise and severely attenuate noise at the high and low ends of the audible spectrum. The DIN characteristic, for example, boosts noise in the region of 5kHz by more than 8dB. Above 10kHz, the response takes a nosedive at the extreme rate of 48dB/octave. At the low frequency end, the DIN response starts off at a rate of 6dB/octave below 1kHz but the rate approaches 12dB/octave around the 20Hz mark.

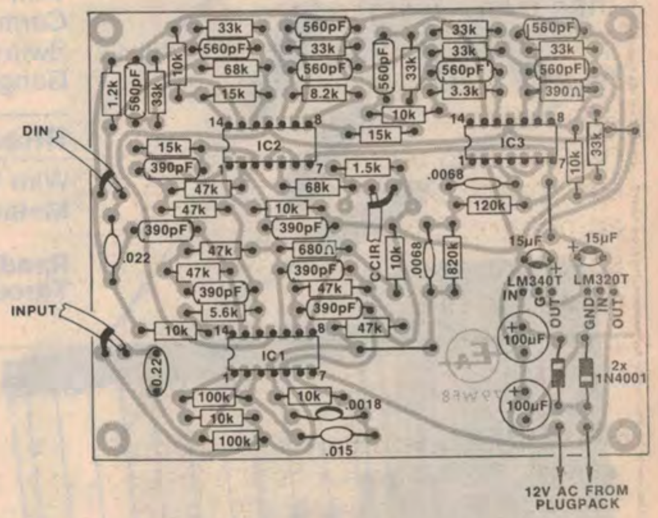
By comparison, the CCIR characteristic boosts the noise centred on about 7kHz by almost 13dB. Above 10kHz, the response slopes off rapidly at about 13dB/octave, which is not quite as drastic as the high frequency roll off for the DIN filter. The low frequencies are rolled off below 1kHz at the modest rate of 6dB/octave.

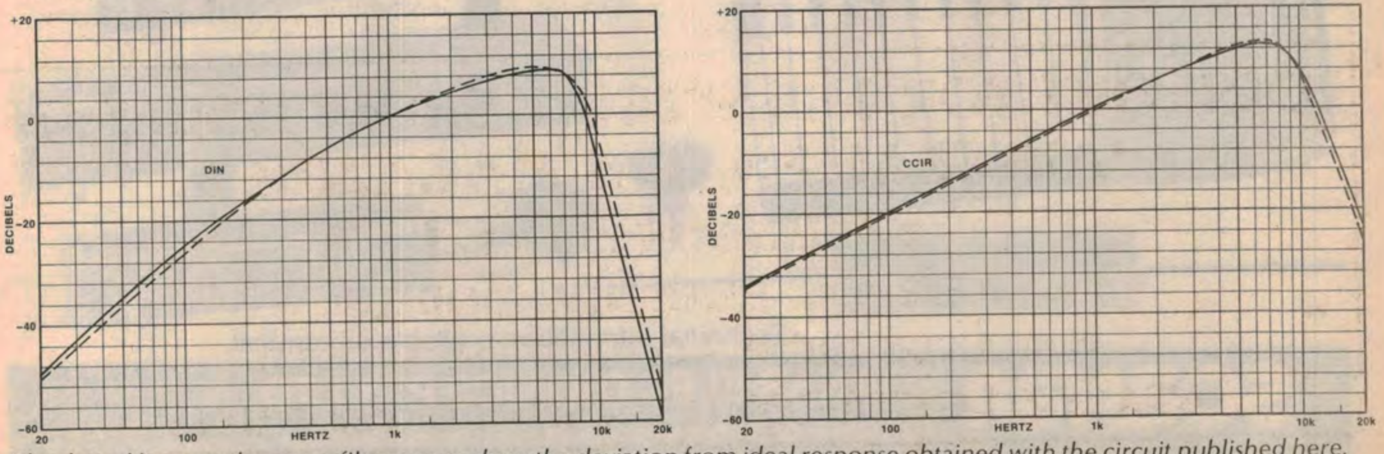
Our approach to the design of this composite filter was to use basic second and third order filters in cascade to make up the desired response — DIN or CCIR. The third page of this article shows the basic circuits of second and third order filters, each using an op amp in the non-inverting mode.

As shown in the diagram, the second order filter has two resistors and capacitors marked R and C, respectively. The R and C values determine the "corner" frequency of the filter, ie the frequency at which rolloff begins. Since this is a second-order filter, it has two RC time-constants and gives a maximum slope beyond the corner frequency of 12dB/octave.

As shown, the op amp circuit is a low pass filter, which means that it passes the frequencies below the corner frequency (also known as the

Three uA4136 quad op amps provide the active circuitry for these weighting filters.





The dotted lines on these two filter curves show the deviation from ideal response obtained with the circuit published here.

"breakpoint" or "-3dB point") and blocks those above. By interchanging the positions of the components marked R and C, the circuit can be changed to a high pass filter, which has the opposite function to a low pass filter.

In the same circuit, resistors R1 and R2 control the negative feedback around the op amp and thus the "Q" of the filter. "Q" is a term originally used to describe the sharpness of tuned circuits. Used in this context, "Q" describes the sharpness of the filter characteristic in the region of the corner frequency.

By the way, the filters used in these circuits have a "Butterworth" response. For the purpose of this article, we can define Butterworth filter response as providing a reasonably sharp corner characteristic together with minimum phase anomalies in this region and having a flat frequency response within the pass-band.

Together with the second-order filter is shown the circuit for a third-order filter. This is merely a second-order filter with a passive RC network tacked on the end and followed by a unity-gain op amp buffer (or "voltage-follower"). A third-order filter gives a maximum slope beyond the corner frequency of 18dB/octave.

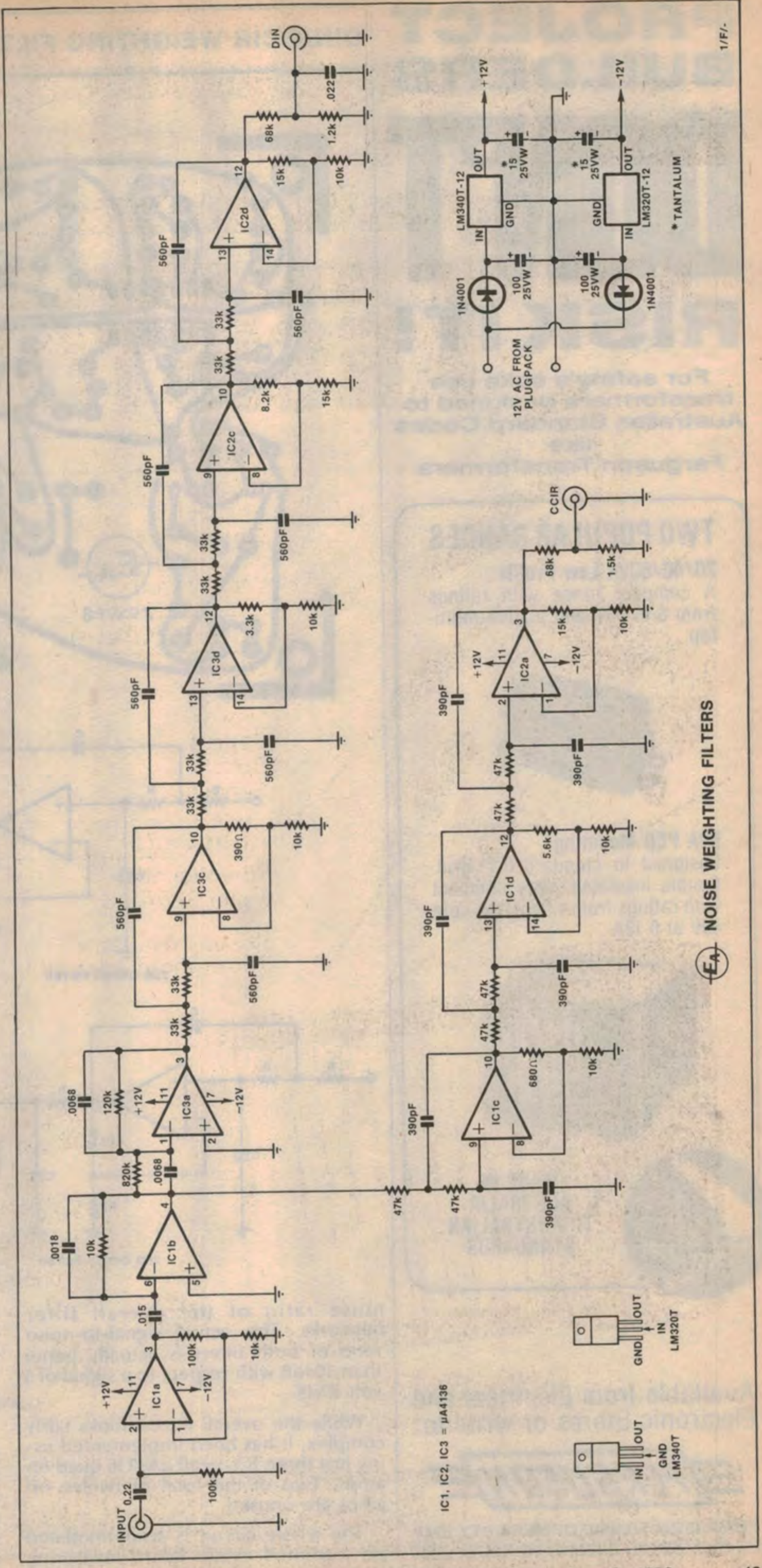
In the same way, a 4th-order (24dB/oct) filter can be made by cascading two second-order filters. A fifth order (30dB/oct) filter can be obtained by cascading second and third-order filters.

Reference to the complete circuit of the composite DIN/CCIR filter will show how we have used combinations of second-order filters to obtain the desired slopes. The CCIR filter for example uses three second-order filters in cascade, to obtain the 36dB/octave treble cut-off slope. Similarly, the DIN filter uses four second-order filters to obtain the 48dB/octave treble slope.

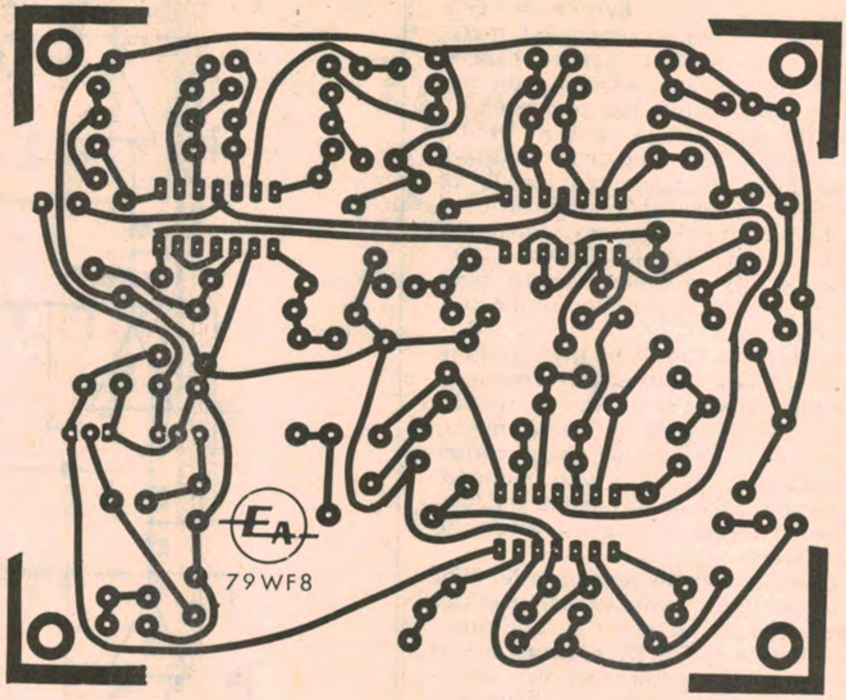
The bass rolloff slope of the CCIR filter (6dB/oct) is provided by op amp IC1b, acting as a high-pass filter. The more complex bass rolloff slope for the DIN filter is provided by the combined effects of IC1b and IC3a.

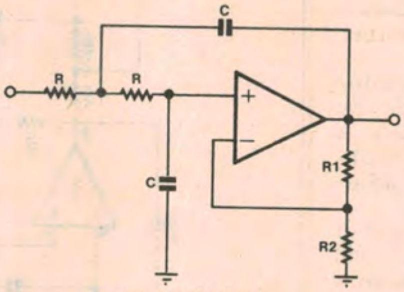
One of the design assumptions for these filter stages is that each will be driven by a low source impedance. This condition is met automatically when cascading filters, since each stage has a low output impedance. However, the first filter stage still needs a buffer and this function is provided by IC1a. This stage also provides a signal gain of 10. This is balanced by a passive attenuator at the output of the DIN and CCIR filters, to restore the overall gain to unity.

The reason for first amplifying by ten and then attenuating by the same amount is to improve the signal-to-

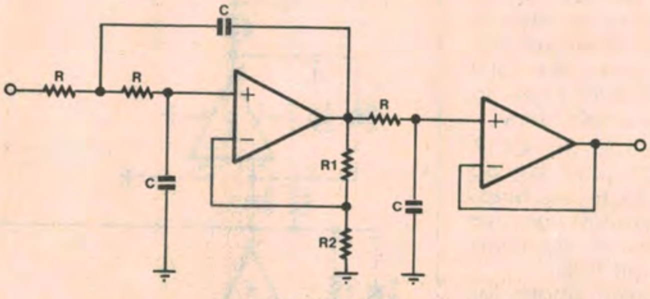


EA NOISE WEIGHTING FILTERS





2ND ORDER FILTER



3RD ORDER FILTER

At top is the full size artwork for the PCB. At left, diagrams of the basic second and third-order low-pass filters. Interchanging capacitors and resistors marked R & C produces high-pass filters.

noise ratio of the overall filter networks. The actual signal-to-noise ratio of both filters is actually better than 104dB with respect to a signal of 1 volt RMS.

While the overall circuit looks fairly complex, it has been implemented using just three ICs — all uA4136 quad op amps. Two of the total of twelve op amps are unused.

The whole circuit is accommodated on a printed circuit board measuring 105 x 86mm (coded 79WF8). This includes the power supply rectifier and filters and the positive and negative regulator ICs. We powered our prototype from a 12VAC plugpack transformer.

We mounted the PCB in a diecast box to obtain effective shielding. The combination of effective shielding and external power transformer keeps hum to an absolute minimum and helps achieve the excellent signal-to-noise ratio quoted above.