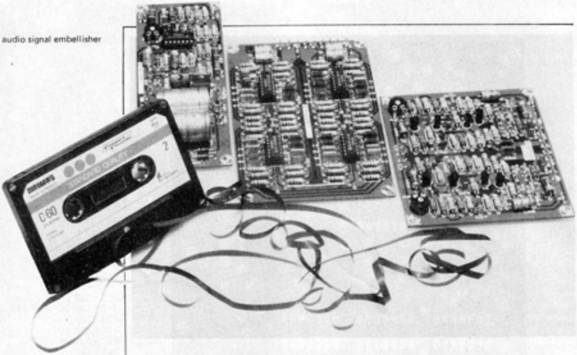

audio signal embellisher

from an idea by
J.F. Brangé

signal restoration with stereo simulation

It is often unavoidable to have to connect an item of mono equipment that is rather less than hi-fi to a modern stereo installation. Although this may give some improvement in the resulting sound quality, the reproduction remains monaural (mono) invariably with a level of hum and noise which by present-day standards is unacceptable. We have designed a circuit which by hum suppression, stereo simulation, and dynamic noise limiting (DNL) gives a greatly enhanced performance. The stereo effect is created by splitting the audio spectrum into sixteen frequency bands which are fed alternately to the left and right-hand channels.

Ever since the arrival of hi-fi audio equipment and the introduction of stereo, our aural senses have been spoilt to the point of addiction. Nowadays when we listen to ordinary monaural music, we soon feel there's something missing. If in addition the sound is accompanied by hum and noise, this feeling soon becomes one of disappointment or even annoyance. However, sometimes there is no alternative to the poor sound source, if only for the simple reason that we don't want to throw away perfectly good equipment. This could, for instance, take the form of simple cassette recorders, AM receivers, sound projectors, and TV sets or video recorders. The last three are particularly prone to being neglected by audio designers. While the picture quality is praised (often deservedly so) as hi-bri (high brilliance), more often than not the sound is a disgrace by modern standards.

Spatial sound

We are aware of depth in sound because we have two ears. As the sound waves reach each ear at a slightly different time and with a slightly different amplitude, the brain receives two separate signals. It is able to deduce the relative position of the sound source from the differences: our ears form a true stereo receiver! The shape of the ear also plays a role: if you want to know more about this, we refer you to 'our remarkable sense of pitch' in the May 1979 (UK) issue of Elektor.

What can we do with a mono sound? It is impossible to convert it into true stereo, because the subtle differences between the left and right-hand channels just cannot be added afterwards. What we can do is to create artificial differences by splitting the sound into a number of frequency bands and then feed these selectively to the left or right-hand channel of the stereo installation. This is, by the way, the method

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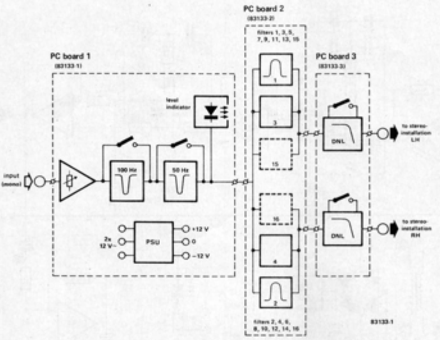

Figure 1. Block schematic of the entire circuit. The three separate modules are shown in dashed lines.

used in the TDA 3810 stereo-IC featured in 'pseudo stereo' in our November 1983 issue. The present design is rather more radical and effective: the audio spectrum is split into sixteen bands by means of active filters. If the filter outputs are numbered 1 . . . 16 in order of ascending centre frequency, all odd-numbered frequency bands are fed to the left-hand channel, and all the even ones to the right-hand channel. The result is truly remarkable: the sound, which at first seemed to come from between the speakers, now seems to 'hang in space' around the speakers.

The block schematic

The block schematic in figure 1 clearly shows that the design consists of three distinct main parts: each of these is housed on a separate printed-circuit board.
The input of the circuit is a pre-amplifier (with variable sensitivity), followed by a 100 Hz and a 50 Hz band-stop filter (sometimes called a 'notch' filter). These filters respectively reject the 100 Hz fundamental frequency of a double-phase rectified voltage and the 50 Hz fundamental of a single-phase rectified voltage. Both filters can be switched out.
The next element is a level indicator which is useful when the input sensitivity is set. Nothing sophisticated, just a simple amplifier and LED which blinks away quietly when the sensitivity is set correctly.
Next, we come to the heart of the design: the sixteen active band-pass filters. The outputs of the odd-numbered filters, and those of the even-numbered ones, are separately combined and are then, in principle, suitable for processing in a stereo installation.
We have, however, added dynamic noise limiting (DNL) stages which, if required, can be switched off or be omitted altogether. Some of you may even use this part of the design only.

The circuit diagrams

There is a circuit diagram for each of the three mains parts of the design: the pre-amplifier, band-stop filters, and power supply (figure 2), the sixteen-element active band-pass filter (figure 3), and the DNL stages (figure 7).

The pre-amplifier, band-stop filters, and power supply

The input sensitivity is preset by means of P1. Pre-amplifier A1 has a gain of about 10 dB and is followed by active band-stop filters A2 (100 Hz) and A3 (50 Hz). The output of A3 is fed to the band-pass filters on the second printed-circuit board (see figure 3), and also to the level indicator stage. After amplification in A4, the signal is applied to the base of T1 via C13. When it exceeds a certain level, T1 conducts to light LED D1.
The power supply for the entire design consists of the customary mains transformer, bridge rectifier, voltage regulators, and smoothing capacitors. The output is symmetrical: ± 12 V at 85 mA.

The band-pass filters

The sixteen band-pass filters (see figure 3) are identical in construction. The basic diagram of one of them is shown in figure 4: a common filter circuit with an opamp as the active element and RC combinations to give the required frequency response and Q factor. As you can see from the formulas

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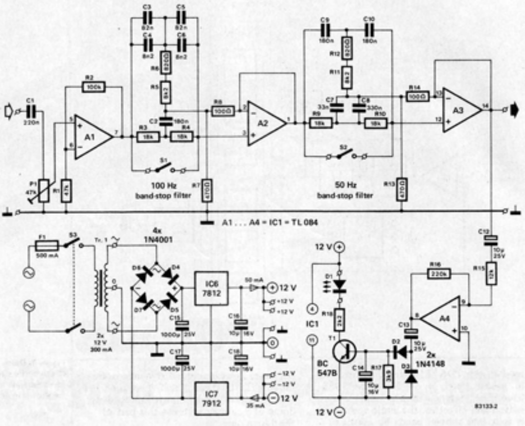

Figure 2. The circuit of the pre-amplifier, band-stop filters, and power supply.

in figure 4, if a fixed value is chosen for R1 and R2, the centre frequency becomes inversely proportional with the value of capacitance C. By appropriate values of C in the sixteen filters, the centre frequencies are varied, but the Q factor and gain A_o, remain the same.

The DNL stages

For those of you who are not completely familiar with the operation of a dynamic noise limiter, here is a short description. The simplest noise limiter is a low-pass filter. Unfortunately, its action is somewhat radical and affects the audio signal. A dynamic noise limiter is a low-pass filter with variable cut-off profile which only functions during soft passages (when the noise is most audible) by suppressing those frequencies to which the ear has the highest sensitivity, that is, about 1 . . . 10 kHz. The amount of suppression is therefore dependent upon the level of the input signal. During loud passages, the cut-off frequency is shifted upwards so that the entire audio range is passed, including the noise, but this is then, of course, masked by the audio signal. At lower levels of signal input, the cut-off frequency is lowered, so that a relatively larger amount of noise is suppressed. The action of a DNL is illustrated by the graphs in figure 5: for an input signal, U_i, of 2.0 mV, the attenuation with respect to the output level at 1 kHz is 10 dB at 7.5 kHz and 20 dB at 10 kHz. The slope is then approximately −18 dB/octave. With input signals above about 8 mV, the response is virtually flat to 20 kHz!

The input stage, A, (see figure 6) ensures correct impedance matching between the band-pass filter and the DNL. From here, the signal is fed to two channels: the upper one consists of a high-pass filter (B), amplifier (D), variable attenuator (E), and fixed attenuator (G), while the lower one comprises a phase shifter (C) and a fixed attenuator (F). The output of the DNL is the sum of the outputs of the two channels which are, of course, in anti-phase.

For low levels of input, U_i, the output, U_1, of the phase shifter is, apart from the phase shift, identical with U_i. The output, U_2, of the high-pass filter contains only the high-frequency content of U_i. Signals U_1 and U_2 are, as already stated, in anti-phase so that if they are summed the high-frequency content of U_i is cancelled out. The net result is therefore that of a low-pass filter. When the level of input signal rises, the variable attenuator in the upper channel comes into operation and reduces the contribution of U_2 to the output signal, U_o. The high-frequency portion of U_i is then no longer (or to a lesser degree) suppressed and U_o will tend to resemble U_i more and more. Turning to the circuit diagram (see figure 7), the input amplifier, transistor T2, in con-

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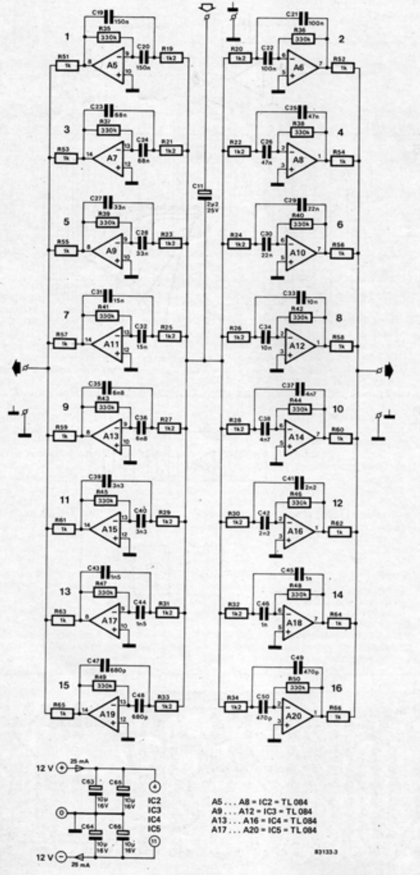

Figure 3. Circuit diagram of the sixteen-element band-pass filter unit. The stereo effect is obtained by feeding the frequency bands alternately to the left and right-hand channels.

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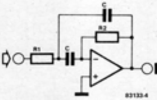

centre frequency: $f_0 = \frac{1}{2\pi C \cdot \sqrt{R_1 \cdot R_2}}$

amplification at f_0: $-A_0 = \frac{R2}{2R1}$

Q factor: $Q = \pi R2 C f_0$

bandwidth: $B = \frac{1}{\pi R2 \cdot C}$

Figure 4. Basic circuit of a band-pass filter showing the formulas for calculating the various filter characteristics.

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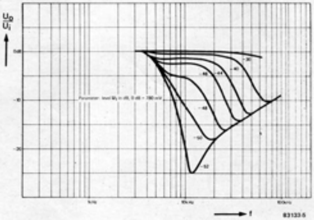

Figure 5. Transfer characteristic of the DNL: the filter action is dependent upon the level of the input signal.

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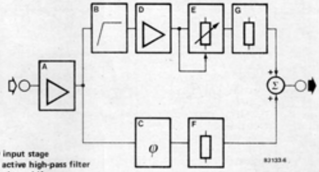

A = input stage
B = active high-pass filter
C = phase shifter
D = amplifier
E = variable attenuator
F, G = fixed attenuator

Figure 6. Simplified block schematic of the DNL.

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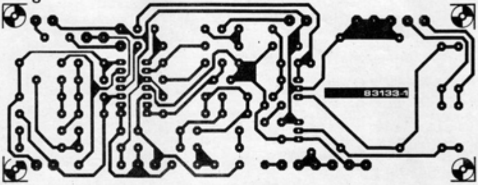

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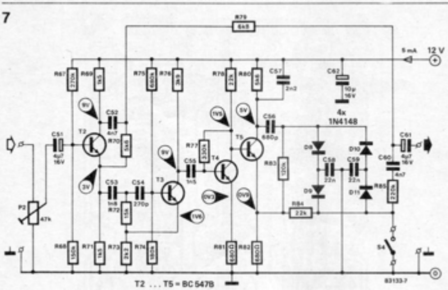

Figure 7. The circuit diagram of the DNL: two such circuits are required, one for each channel.

junction with C52 and R70, forms the phase shifter. The output of the phase shifter is taken to the DNL output via fixed attenuator R70/R79.
The active high-pass filter, formed by C53, C54, T3, and R72 . . . 76, is followed by amplifier T4 and a variable attenuator consisting of T5 and associated components. The collector as well as the emitter of T5 feed a signal to the diode bridge D8 . . . D11. Capacitors C58 and C59 are charged to the emitter voltage via R83/D8 and R84/D11 respectively. If the audio signal level lies below the forward voltage of the diodes, these will not conduct. The signal from T5 is then taken directly to the DNL output where it is summed with the signal from the phase shifter. As the two signals are in anti-phase, the cut-off frequency is about 6 . . . 7 kHz and filter action is at a maximum.
When the audio signal is greater than the diode forward voltage, the diodes conduct and present a low impedance to audio frequencies. A low-pass filter is then formed by R84, C58, C59, which causes the higher frequencies to be attenuated. The end result will be that fewer (or hardly any) high frequencies are removed from the final output signal, which shows up as a flattening of the overall frequency response.

Construction

As stated before, the design is built up from three modules: pre-amplifier plus power supply plus band-stop filters, the sixteen-element band-pass filter, and the DNL stages. This type of construction makes it possible for everyone to choose which part(s) of the design he needs: some of you may not want the stereo effect, in which case all you have to do is omit the sixteen-element band-pass filter. If the DNL unit only is built, it is, of course, necessary to add a suitable power supply.
When the printed-circuit boards shown in figures 8 . . . 10 are used, no particular problems should be encountered in the construction. During the building of the power supply, make sure that one voltage regulator IC is turned 180° with respect to the other. In view of the small current consumption, these ICs do not need heat sinks.
The band-pass filter board is best commenced by wiring in the four wire bridges which are to be located under IC2 . . . IC5: this will make things a lot easier later on.
The DNL board consists of two absolutely symmetrical halves: it is possible to cut it into two and have two independent mono DNLs! In contrast to the remainder of the

Parts list (DNL)
Circuit: figure 7
PC board: figure 10

Resistors:
R67,R67' = 270 k
R68,R68' = 150 k
R69,R69',R71,R71' = 1k5
R70,R70',R80,R80' = 5k6
R72,R72' = 15 k
R73,R73' = 2k2
R74,R74' = 180 k
R75,R75' = 680 k
R76,R76' = 3k9
R77,R77' = 330 k
R78,R78',R84,R84' = 22 k
R79,R79' = 6k8
R81,R81',R82,R82' = 680 Ω
R83,R83' = 120 k
R85,R85' = 220 k
P2,P2' = 47 k (50 k) preset

Capacitors:
C51,C51',C61,C61' = 4µ7/16 V
C52,C52',C60,C60' = 4n7
C53,C53' = 1n8
C54,C54' = 270 p
C55,C55' = 1n5
C56,C56' = 680 p
C57,C57' = 2n2
C58,C58',C59,C59' = 22 n
C62,C62' = 10 µ/16 V

Semiconductors:
D8 . . . D11,D8' . . . D11' = 1N4148
T2 . . . T5,T2' . . . T5' = BC 547B

Miscellaneous:
S4 = DPST switch

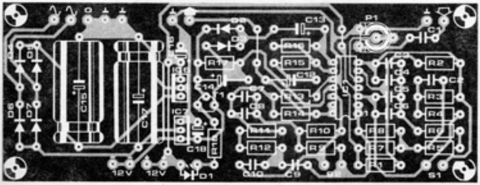

Figure 8. Layout and component side of the printed-circuit board for the pre-amplifier, band-stop filters and power supply.

Parts list (filters and power supply)
Circuits: figures 2 and 3;
PC boards: figures 8 and 9

Resistors:

R1 = 47 k
R2 = 100 k
R3,R4 = 18 k
R5,R11 = 8k2
R6,R12 = 820 Ω
R7,R13 = 470 Ω
R8,R14 = 100 Ω
R9,R10 = 18 k
R15 = 12 k
R16 = 220 k
R17 = 3k9
R18 = 2k2
R19 . . . R34 = 1k2
R35 . . . R50 = 330 k
R51 . . . R66 = 1 k
P1 = 47 k (50 k) preset

Capacitors:

C1 = 220 n
C2,C9,C10 = 180 n
C3,C5 = 82 n
C4,C6 = 8n2
C7,C27,C28 = 33 n
C8 = 330 n
C11 = 2μ2/25 V tantalum
C12,C13 = 10 μ/25 V
C14 = 10 μ/16 V
C15,C17 = 1000 μ/25 V
C16,C18 = 10 μ/16 V
tantalum
C19,C20 = 150 n
C21,C22 = 100 n
C23,C24 = 68 n
C25,C26 = 47 n
C29,C30 = 22 n
C31,C32 = 15 n
C33,C34 = 10 n
C35,C36 = 6n8
C37,C38 = 4n7
C39,C40 = 3n3
C41,C42 = 2n2
C43,C44 = 1n5
C45,C46 = 1 n
C47,C48 = 680 p
C49,C50 = 470 p
C63 . . . C66 = 10 μ/16 V

Semiconductors:

D1 = LED
D2,D3 = 1N418
D4 . . . D7 = 1N4001
T1 = BC 547B
IC1 . . . IC5 = TL 084
IC6 = 7812
IC7 = 7912

Miscellaneous:

S1,S2 = SPST switch
S3 = DPST switch (mains)
Tr1 = supply transformer
2 x 12 V/300 mA
F1 = fuse, delayed action,
500 mA
fuse carrier
printed-circuit boards
83133-1 and 83133-2

9

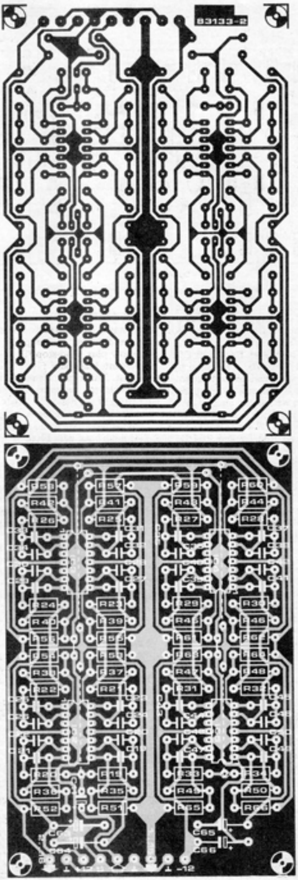

Figure 9. Layout and component side of the printed-circuit for the sixteen-stage band-pass filter.

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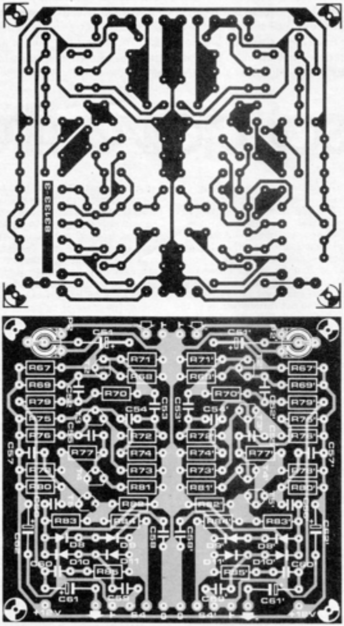

Figure 10. Layout and component side of the DNL board: as the DNL should be suitable for stereo, the board consists of two symmetrical halves.

design, the DNL needs only a single supply line: +12 V and earth.

Calibration

With the output of a tuner or record player connected to the input of the pre-amplifier board, adjust the overall sensitivity by means of P1 until LED D1 quietly blinks in rhythm with the incoming audio signal.

Because the DNL is a variable filter, the action of which is dependent upon the signal level at the base of T2, preset P2 should be adjusted carefully. Connect an a.c. voltmeter (input impedance at least 100 kΩ) between the wiper of P2 and earth, and inject a signal of about 1 V into the input terminals of the DNL. Adjust P2 for a reading 775 mV on the voltmeter. If the input signal was derived from a tuner, or record player, it may be necessary to re-adjust P1 slightly.

If you have no access to a suitable a.c. voltmeter, adjust the preset(s) by ear. Make sure that with a reasonably large input signal the high frequencies are not cut. If that happens, the input signal is too small and must be adjusted with P2. If this has already been set for maximum sensitivity, adjust P1 also. If this still does not give a satisfactory result, the output from the signal source (tuner, record player, tape recorder) is too low, in which case an extra amplifier has to be added.

Final note:

The DNL can be inserted almost anywhere into the audio chain, but as its 0 dB input level must correspond to 775 mV it must be located before the volume control.

In audio technique, all voltages are referred to the 'normal level'. This is 1 mW into 600 Ω (= 775 mV across 600 Ω) and is conventionally designated 0 dBm.