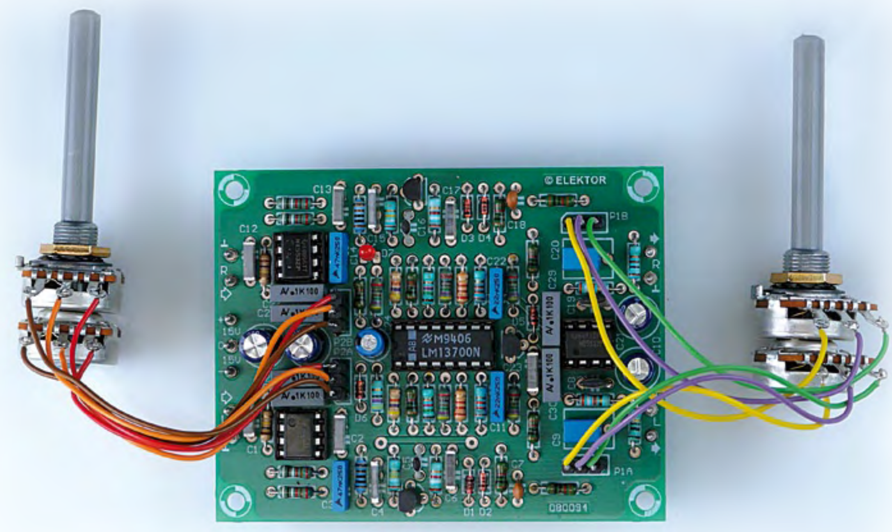
Booster for Audio Signals

Cranks up bass & treble



By Markus Aebi (Switzerland)

Dull and washed-out sound? With modern recording technologies this is unlikely to happen.
There are nevertheless situations where a fuller sound would do no harm (live bands, for example, but cheap headphones could also benefit). A small amount of 'effect' could make the sound experience just 'perfect'.

Weak bass and dull treble frequencies are things of the past with this circuit. The ingenious principle of the project described here makes the reproduction more delicate, fuller and subjectively louder. The circuit was originally intended to be built into a mixing desk (immediately before the Master Fader), but works just as well as a standalone device when the input and output voltages are adjusted appropriately (about $1.5\ V_p$/0 dB at the first op amp).

The principle

The high and low frequencies are processed independently of each other. The left and right channels are identical, with the exception of the part of the circuit controlling the amplification of the low frequencies, which is used for both channels and therefore does not need to be implemented twice (see **Figure 1**).
After the high-pass filter with C4 (C15 for the right channel) the high frequencies are amplified 'quick and dirty' with the circuitry around T1 (T2). The diode pair D1, D2 (D3, D4) generates higher harmonics which are added to the original signal with potentiometer P1A (P1B). This makes it sound 'fresher'.
The low frequencies are separated from the original signal with the network around IC1B (IC2B) and subsequently passed on to the current-controlled operational transconductance amplifier (OTA), which is configured as a VCA here (Voltage Controlled Amplifier, IC4A and IC4B respectively). The 'Dynabass' potentiometer P2A (P2B) determines how much of the original signal is processed. RC network R16/C11 acts as a variable low-pass filter (and also has a variable phase shift with respect to the original signal, which results in a subjective amplification of the low frequencies). The values of the RC pair affect the tuning of the filter. The selected resistor of 33 kΩ in combination with a capacitor of 22 nF gives optimum tuning.
The control current of the VCA, which determines the gain of this IC, is generated with the circuit around T3. This circuit works as a kind of limiter circuit (quasi limiter function) and shows via LED D7 how much 'control' is applied. The processed signal is added to the signal immediately after the 'Harmonics'-potentiometer (P1A and P1B respectively).
When both potentiometers are set to their minimum positions the signal passes through unchanged.
An external symmetrical power supply of ±15 V completes the story. The current consumption at this voltage is about 40 mA for the positive supply rail and 35 mA for the negative supply rail. Each IC is provided with decoupling capacitors and the PCB is fitted with additional 100 µF buffer electrolytics.

Construction

When assembling the PCB you follow the traditional procedure: first the 'small' parts such as resistors and diodes, then the 'bigger' parts such as capacitors and transistors. We used sockets for the ICs so that swapping the op amps for a different sound is very easy. The construction is not all that difficult because no SMD components are used.
In our prototype we chose to use headers and sockets for the connections to the two double potentiometers. These are mentioned in the parts list. It is, of course, also possible to omit those headers and connect the potentiometers directly to the board using short wires.
The double-sided PCB has been made as compact as possible (88 × 69 mm) and has not been designed with a specific enclosure in mind (see **Figure 2**). The PCB layout can be downloaded from the project page [1].

Measurements

The two graphs summarise how the circuit influences the signal. The graph in **Figure 3** shows the amplitude characteristic of the low-pass filter. This shows that the low-fre-

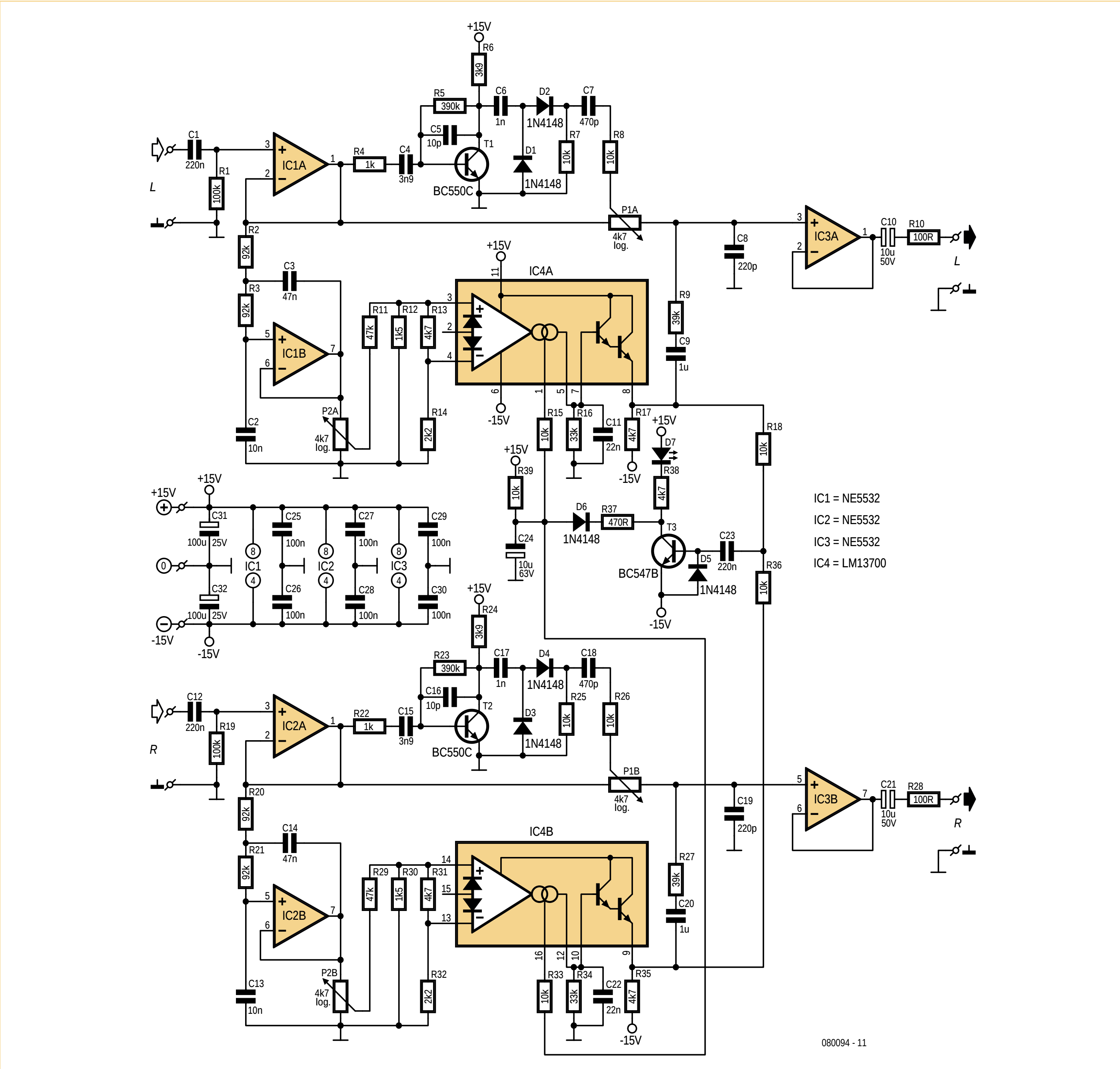


Figure 1. The schematic may strike as quite sizeable, but the dimensions of the PCB are very reasonable.

quency part of the circuit operates below about 100 Hz. At the high-frequency end the circuit operates from around 5 kHz, based on the clipping of T1 (T2).

The second curve (**Figure 4**) shows the ratio between the amplitudes of the input and output signals of the VCA (green curve), measured at pin 9 of IC4. It shows that the curve is linear up to about –10 dB, after which compression occurs. The effect of this is that the low frequencies are accentuated a bit more and the entire sound sounds fuller.

The same graph also shows the distortion of the output signal at 60 Hz and 1 V input

COMPONENT LIST

Resistors
R1,R19 = 100kΩ
R2,R3,R20,R21 = 92kΩ
R4,R22 = 1kΩ
R5,R23 = 390kΩ
R6,R24 = 3.9kΩ
R7,R8,R15,R18,R25,R26,R33,R36,R39 = 10kΩ
R9,R27 = 39kΩ
R10,R28 = 100Ω
R11,R29 = 47kΩ
R12,R30 = 1.5kΩ
R13,R17,R31,R35,R38 = 4.7kΩ
R14,R32 = 2.2kΩ
R16,R34 = 33kΩ
R37 = 470Ω
P1,P2 = 4.7kΩ logarithmic, stereo

Capacitors
C1,C12,C23 = 220nF, lead pitch 5 or 7.5mm
C2,C13 = 10nF, lead pitch 5 or 7.5mm
C3,C14 = 47nF, lead pitch 5 or 7.5mm
C4,C15 = 3.9nF, lead pitch 5 or 7.5mm
C5,C16 = 10pF, ceramic, lead pitch 5mm
C6,C17 = 1nF, lead pitch 5 or 7.5 mm
C7,C18 = 470pF, ceramic, lead pitch 5mm
C8,C19 = 220pF ceramic, lead pitch 5mm
C9,C20 = 1µF, lead pitch 5 or 7.5mm
C10,C21 = 10µF 50V, radial, bipolar, lead pitch 2.5mm, diam. 8.5mm max.
C11,C22 = 22nF, lead pitch 5 or 7.5mm
C24 = 10µF 63V, radial, lead pitch 2.5mm, diam. 6.3mm max.
C25–C30 = 100nF, lead pitch 5 or 7.5mm
C31,C32 = 100µF 25V, radial, lead pitch 2.5mm, diam. 8.5 mm max.

Semiconductors
D1–D6 = 1N4148
D7 = low current LED, red, 3mm
T1,T2 = BC550C
T3 = BC547B
IC1,IC2,IC3 = NE5532, 8-DIP case
IC4 = LM13700 16-dip (e.g. Farnell # 1651866)

Miscellaneous
11 pcs PCB solder pin
4 pcs 3-way pinheader (P1,P2)
4 pcs 3-pway socket
PCB, # 080094-1 see [1]

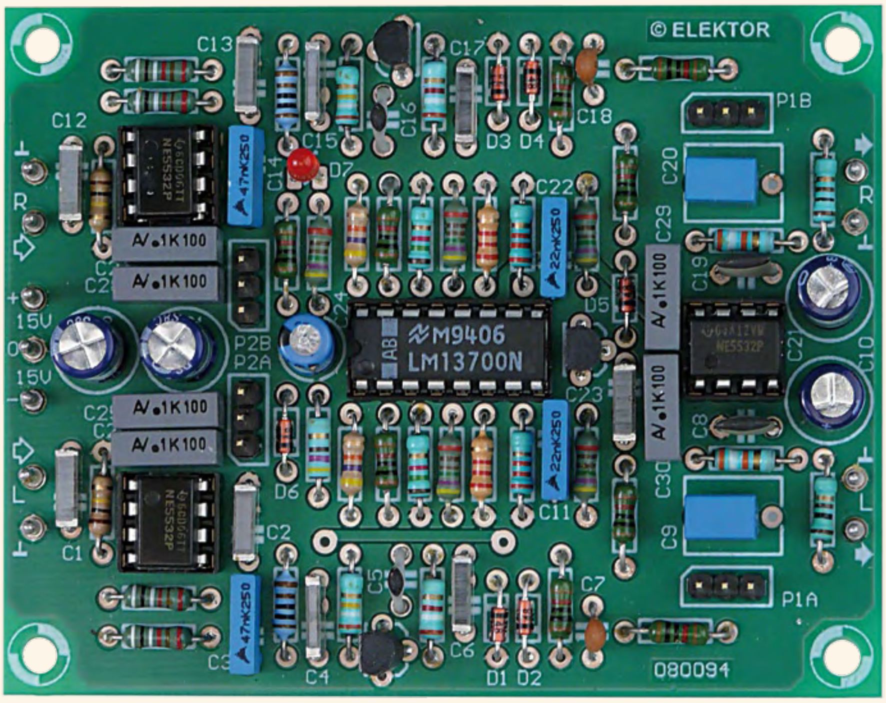


Figure 2. The components are quite close so the circuit will fit on a compact PCB.

voltage, measured at pin 9 of IC4 (blue). The –10 dB value is also represented in this curve, with the difference that the distortion increases exponentially from that point onwards (this makes sense because the signal is compressed above that value).

This is naturally not a circuit of particular interest for audiophiles. But they probably only use flawless signal sources. For everyone else who would like their sound to be fresher and livelier this booster circuit is a good alternative to an equalizer.

(080094-I)

Internet Link

[1] www.elektor.com/080094

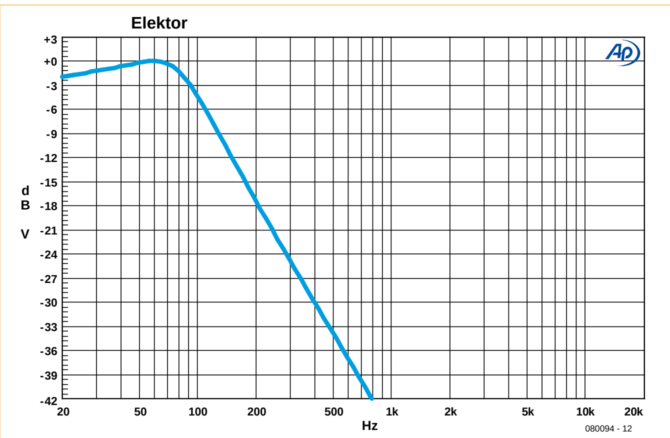


Figure 3. This clearly shows the frequency range of the low frequency part of the circuit.

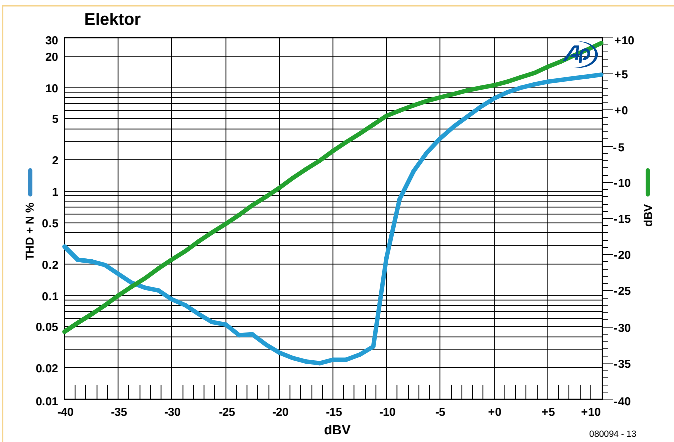


Figure 4. The gain and distortion of the low frequencies depends on the input voltage.