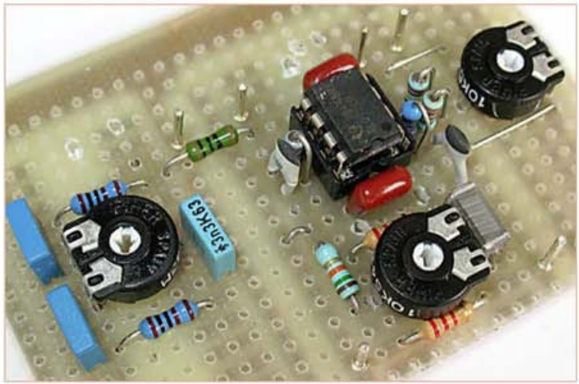
# A higher note...

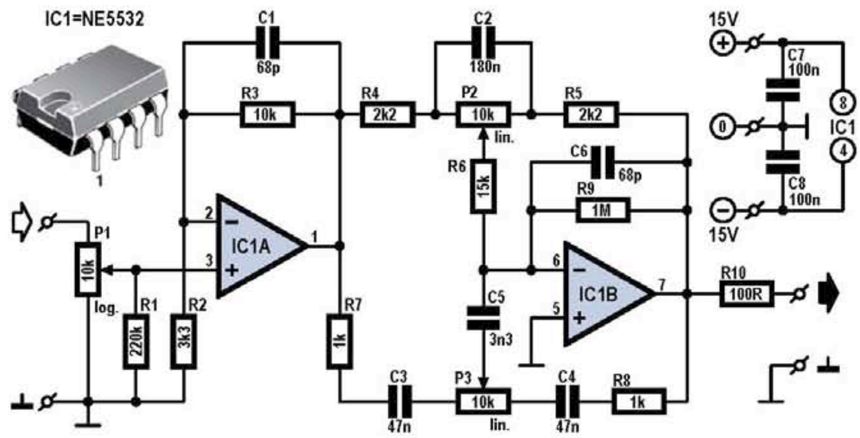
By Ton Giesberts (Elektor Labs)

If as a child you are sent to the butchers to fetch an ounce of ham, you'll no doubt have heard the good man saying "Does it matter if it's a little more?" well after cutting the stuff. Likewise, the trigger to design a tone adjustment circuit was a question I recently overheard within an all-technical environment: "Does it matter if the tone's a little higher?"

In the article on the tube sound mimicking amplifier elsewhere in this Xmas supplement, we mentioned on the side that the audio world can be divided in to two factions: the side containing vacuum tube enthusiasts and the side propagating transistors. This isn't the full story though, since we can draw a line between two other groups: the purists and the relativationists. To the purists, 'equalising' (or tone adjustment) is a very dirty word, and the ideal preamplifier consists of nothing but a length of copper wire (preferably gold-plated), although they have been known to add a potentiometer here and there for volume control to keep the peace around the neighbourhood. The second group is less strict and doesn't have a problem with amplifying or muffling pieces of a music signal to fit their liking. If you belong to the latter, this tone adjustment design is something you might be interested in

The design of this equaliser could be classified as 'classic'. Furthermore, the circuit also amplifies the signal at least by 4 times, which



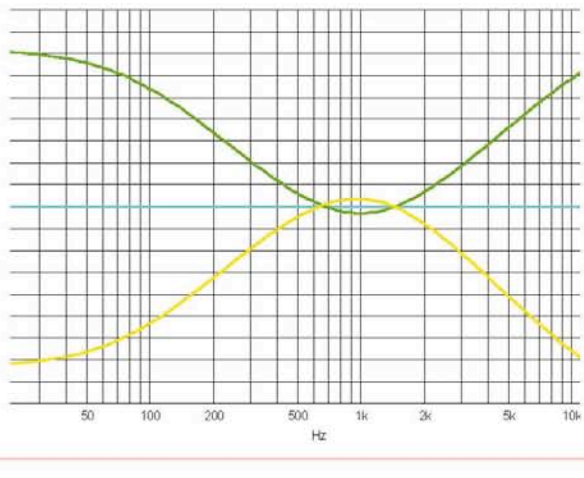



allows this circuit to bear the designation of 'full-fledged preamplifier. It is the ideal supplement to the above mentioned PWM amplifier, especially when you're not using high-end speakers. The range of adjustment is approximately 14 dB (at 20 Hz and 20 kHz), so if this isn't enough, there's definitely something wrong with the speaker connected up!

Looking at the circuit diagram from the left to the right, the first thing you see is potentiometer P1, which controls the volume. This is followed by the buffer/amplifier chip, IC1a. This is a non-inverting amplifier which is configured to produce a gain of 4 times (R3/R2 + 1). At a ±15 V symmetrical power supply, this amplifier can easily process a 2 V signal from a MP3, CD or DVD player.

Next comes IC1b which takes care of the actual tone adjustment. This opamp is wired as an inverting amplifier with two negative feedback loops in parallel, one for the high frequencies, and one for the low ones. The workings of this part of the circuit aren't hard to understand if we look at the two extreme ends of the frequency range: very high and very low.

First, the very low frequencies. The capacitors C3, C4 and C5 create a high-pass filter which effectively hinders all low frequencies from being transferred, removing P3's influence on the low frequencies. Capacitor C2 also blocks low frequencies, causing the amplification of the low frequencies to be determined by the setting of P2. The maximum amplification is (R5 + P2) / R4 = 5.5× and the minimum amplification is R5 / (R4 + P2) = 0.18×.




Now for the very high frequencies. As opposed to the low frequencies, C2 short-circuits all high frequencies, removing P2's influence on them. C3, C4 and C5 also have a very low resistance, which causes the high frequencies to be passed through P3, allowing them to be controlled. Superficially, one might say the range of amplification is between (R8+P3) / R7 = 11× and R8 / (R7+P3) = 0.09×. In practice, the amplification is between 6.5× and 0.15× due to the fact that R4 through R6 are actually in parallel to the high-frequency circuit. In any case, P3 controls the high tones.

Of course we've tested this equaliser in the lab, the results of which are shown in the graph. The green curve is the frequency distribution when both potentiometers (P2 and P3) are set at their maximum, and the yellow curve was measured with P2 and P3 at their minimum. The blue curve, which is actually a very neat straight line, was measured with P2 and P3 at their middle position. From this graph you can see the range is very wide!

Volume control P1 is at the input of the circuit instead of at the output to prevent the equalisation from being overdriven.

The actual construction is fairly straightforward. Just like in the picture, building the circuit is easy on a piece of prototyping board. Remember to solder neatly and to keep the wiring to the potentiometers as short as possible.

We picked an NE5532 dual opamp to take care of the amplification, since it's specially designed for audio signal processing. This opamp causes a distortion of just 0.002% at an output voltage of 1 V for frequencies up to 20 kHz. The drawback of this opamp is its relatively high power consumption (7.5 mA). If this is unacceptable, a 'regular' dual-opamp like the TL072 can also be used. This reduces the power consumption to approximately 3.8 mA, which allows the circuit to be powered by two 9 V batteries, instead of an AC power adapter. The distortion in the signal does increase to 0.007% at 20 kHz when using the TL072, which is still acceptable.

R2 can be left out of the circuit if you feel that the first amplification of the signal in IC1a is a bit high. The buffer will then amplify just 1×.

The circuit was designed for use with small loudspeakers. When larger speakers are used, the value of C2 needs to be increased, which decreases the bandwidth. The range of adjustment can be decreased by decreasing the capacitance of C5 a little.

Enjoy the build, and have fun using the circuit!

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