AVC for PCs

limits differences in sound level

An annoying phenomenon (not restricted to PCs) is that each and every programme that produces sound does so at a different level. This means almost constant adjustment of the volume control to ensure audibility of one programme and protection of the ear drums with another one. The control circuit described in this article is designed to obviate this nuisance: it constantly monitors the signal even at the output of the soundcard and adjusts it when required. Use of the circuit is not restricted to PCs; it may also be used as a dynamic limiter in existing audio equipment.

Design by T. Giesberts

Brief specification

Power output	*1.2 W*
Maximum input	*1 V*
Compression	*10:1*
Supply line	*12 V, 6 VA*
Output load	*8 Ω (LSP); 10 kΩ (line)*
Input sensitivity	*280 mV (gain line in to out = 0 dB; distortion at output = 1%)*
	120 mV (gain line in to out = max; distortion at output = 1%)
Line in to LSP out (input voltage = 200 mV)	
THD+N	*0.25% (2×0.5 W)*
Signal-to-noise	*70 dB for 0.5 W output at maximum gain*
Channel separation	*>45 dB*
Line in to line out (input voltage = 200 mV; no loudspeaker connected)	
THD+N	*0.047%*
Signal-to-noise	*80 dB*
Channel separation	*>73 dB*

In general, the signal levels in current audio equipment are equalized and standardized (although there are still some exceptions). Software manufacturers do not seem to know or care about this. Anyone who has ever opened two different sound programs will know of the quite different levels various effects often have. This is obviously an annoying situation and one which makes the constant adjusting of the volume control a necessity.

The present circuit offers a solution to this problem. It consists of a dynamic compressor with a control range of 10:1 which ensures that very loud and very soft sound passages are attenuated or amplified respectively. This results in a much narrower dynamic range of audio signals which makes adjusting the volume control a much less frequent necessity. It proves that something that appears difficult in software can be easily achieved by a small electronic circuit.

DESIGN

The block diagram of the automatic volume control (AVC) circuit is shown in **Figure 1.** The stereo audio signal at the output of a sound card used in a multimedia PC is applied to the line input. The active part of the circuit consists of two integrated amplifiers that contain a variable preamplifier and a compact output amplifier.

The signal from the output amplifier is freed from any direct voltage and then applied to a discrete rectifier. After the rectified signal has been processed, it is used to control the amplification factor of the preamplifiers. The control circuit is based on a number of discrete operational amplifiers. The design is an OR-type, so that the sound channel (lefthand or righthand) with the highest peak signal level determines the amplification factor of

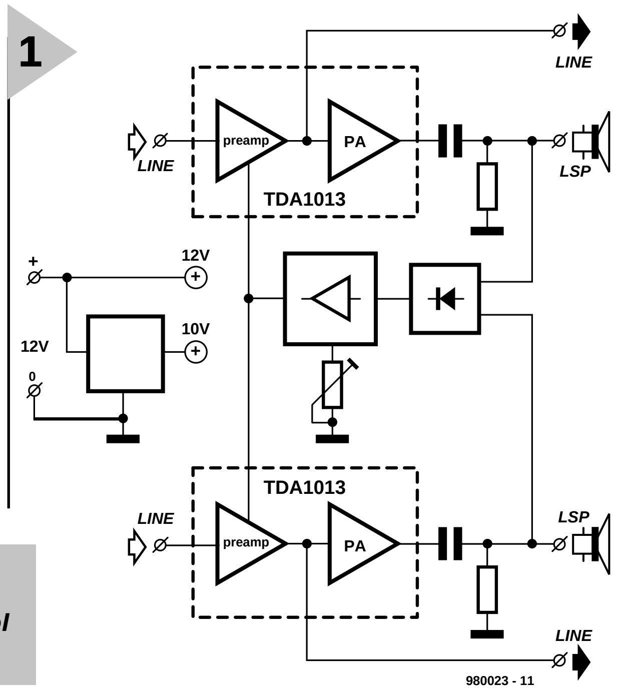


Figure 1. Block diagram of the automatic volume control circuit for PCs.

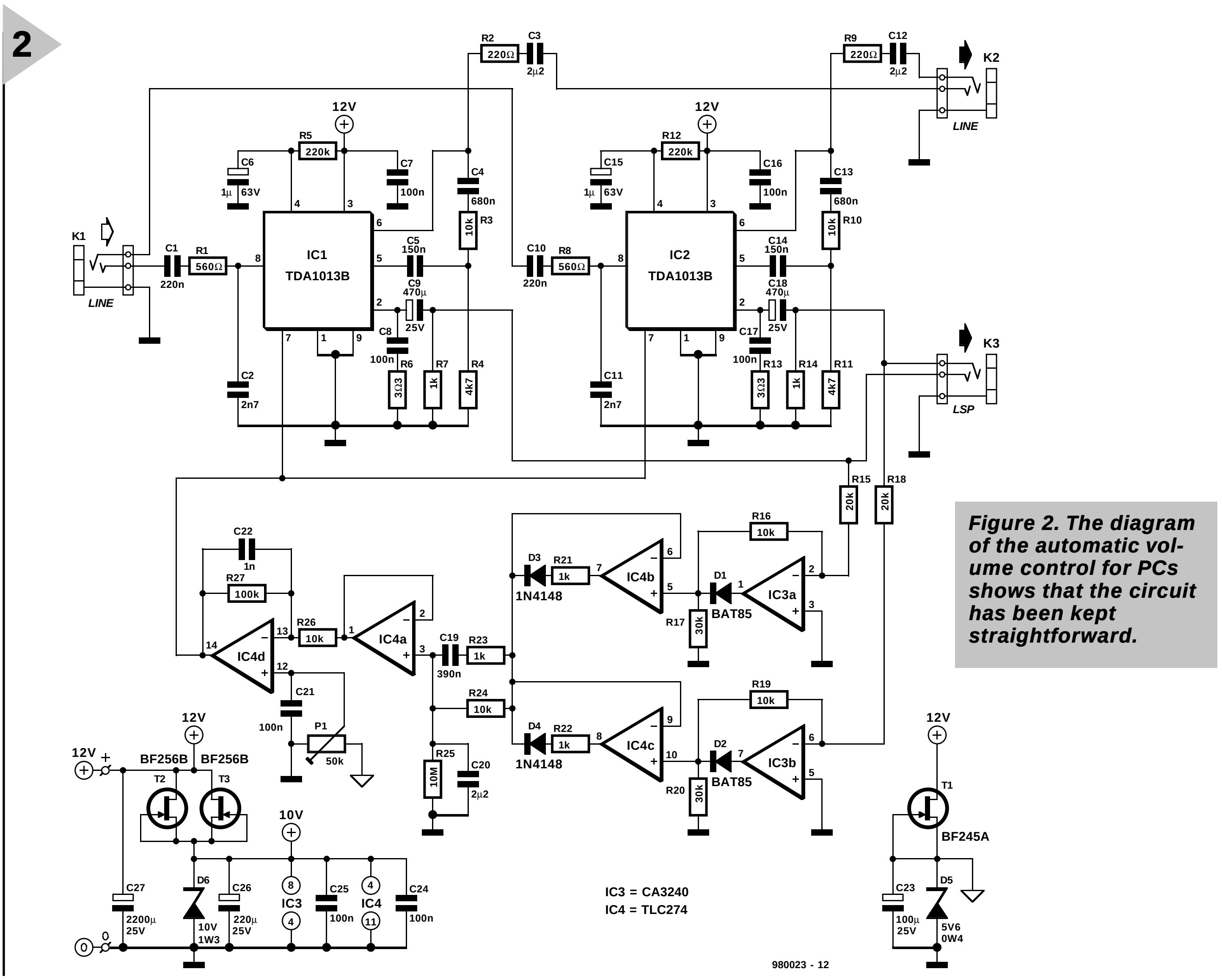


Figure 2. The diagram of the automatic volume control for PCs shows that the circuit has been kept straightforward.

Parts list

Resistors:
R_1, R_8 = 560 Ω
R_2, R_9 = 220 Ω
$R_3, R_{10}, R_{16}, R_{19}, R_{24}, R_{26}$ = 10 kΩ
R_4, R_{11} = 4.7 kΩ
R_5, R_{12} = 220 kΩ
R_6, R_{13} = 3.3 Ω
$R_7, R_{14}, R_{21}, R_{22}, R_{23}$ = 1 kΩ
R_{15}, R_{18} = 20 kΩ
R_{17}, R_{20} = 30 kΩ
R_{25} = 10 MΩ
R_{27} = 100 kΩ
P_1 = 50 kΩ (47 kΩ) preset

Capacitors:
C_1, C_{10} = 0.22 µF
C_2, C_{11} = 0.0027 µF
C_3, C_{12}, C_{20} = 2.2 µF metallized polyester (MKT), pitch 5 or 7.5 mm
C_4, C_{13} = 0.68 µF
C_5, C_{14} = 0.15 µF
C_6, C_{15} = 1 µF, 63 V, radial
$C_7, C_8, C_{16}, C_{17}, C_{21}, C_{24}, C_{25}$ = 0.1 µF
C_9, C_{18} = 470 µF, 25 V, radial
C_{19} = 0.39 µF
C_{22} = 0.001 µF
C_{23} = 100 µF, 25 V, radial
C_{26} = 220 µF, 25 V, radial
C_{27} = 2200 µF, 25 V, radial

Semiconductors:
D_1, D_2 = BAT85
D_3, D_4 = 1N4148
D_5 = zener diode 5.6 V, 400 mW
D_6 = zener diode 10 V, 1.3 W
T_1 = BF245A
T_2, T_3 = BF256B

Integrated circuits:
IC_1, IC_2 = TDA1013B
IC_3 = CA3240E
IC_4 = TLC274CN

Miscellaneous:
K_1–K_3 = 3.5 mm stereo audio socket for board mounting
PCB Order no. 980023-1 (see Readers Services towards the end of this issue)

3

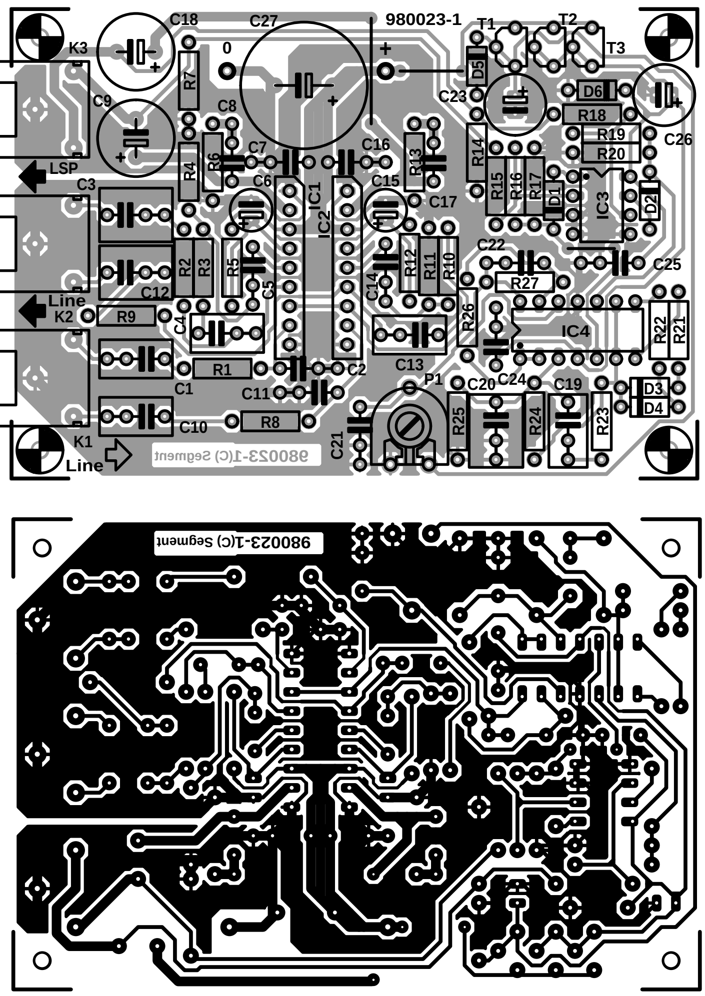


Figure 3. The printed-circuit board for the automatic volume control.

the stereo preamplifier.

The values of various components in the control circuit are chosen to ensure a fast attack time and a long release time. This ensures that short-duration signal peaks are effectively suppressed, whereupon the circuit recovers (relatively) slowly from the damping action.

Power for the circuit is derived from a standard 12 V mains adaptor.

CIRCUIT DESCRIPTION

In the circuit diagram in **Figure 2**, the preamplifier-output amplifier combination is contained in IC_1 and IC_2. This type of IC is a compact 4 W audio amplifier with integral voltage-controlled volume control. The range of the logarithmic volume control is 80–90 dB with control voltages between 6.5 V (+5 dB) and 2.0 V (–80 dB).

The control voltage is applied to pin 7 of the IC. The line output is at pin 6, which is linked via a capacitor to pin 5, the input of the output amplifier..

The circuit has three stereo terminals: line in, line out, and power out.

The power output is 2.5 W for a loudspeaker impedance of 8 Ω and a supply line of 18 V, which is sufficient for most applications.

The analogue input signal at the line input, K_1, is applied to pin 8 of IC_1,raised in the preamplifier and output via pin 6. The transfer between pins 8 and 6 depends on the control voltage at pin 7.

The line signal is attenuated and its level made suitable for inputting to the output amplifier by networks R_3-R_4 and R_{10}-R_{11}. Assuming a supply line of 12 V, the output amplifier is driven fully ($P_{o(max)}$ = about 1.2 W into 8 Ω) by an input signal of 90 mV.

RC networks are provided at the inputs (R_1-C_1-C_2 and R_8-C_{10}-C_{11}) and the line outputs (R_2-C_3 and R_9-C_{12}).

The output amplifier outputs are provided with large electrolytic capacitors, C_9 and C_{18}.

The supply lines are decoupled by C_7 and C_{16}.

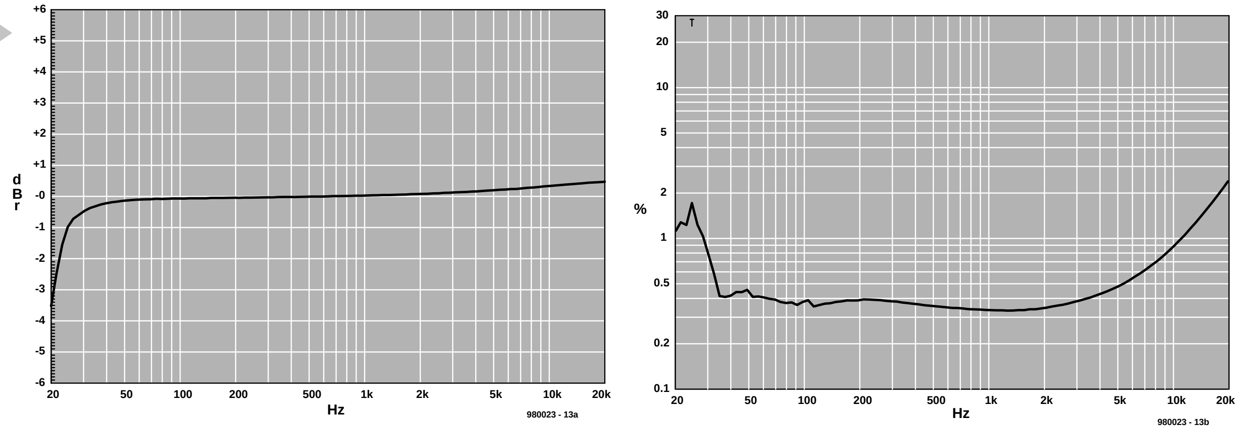
Filters R_6-C_8 and R_{13}-C_{17} ensure that the amplifiers remain stable at high frequencies.

RECTIFICATION AND REGULATION

The audio signal to be rectified is taken from the loudspeaker terminals and applied to IC_{3a} and IC_{3b}. The following description is based on IC_{3a}.

Negative signals are inverted by the op amp and amplified by a factor that depends on the ratio R_{15}:R_{16}. In the present circuit, this is –2, that is, attenuation. With positive signals, the op amp is overdriven and its output negative. Diode D_1 is then cut off and half the input voltage is available at its cathode [$R_{17}/(R_{15}+R_{16}+R_{17})$]. This means that the op amp behaves as a full-wave rectifier/amplifier, whose amplification is the same (0.5) for both

4



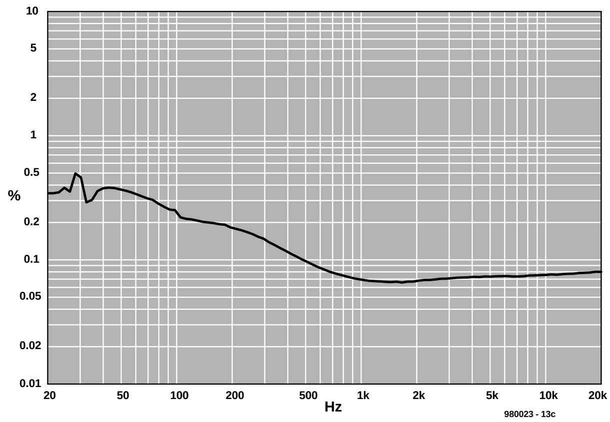


Figure 4. Some graphs to show a few of the electrical properties of the AVC. (a) is the frequency response curve measured between the line input and loudspeaker input terminals. (b) is the distortion characteristic measured at the loudspeaker output terminals (5 W into 8 Ω). (c) is the distortion characteristic measured at the line output terminals terminated into 10 kΩ.

halves of the input signal.

Operational amplifiers IC_{4a} and IC_{4b} are half-wave rectifiers whose outputs are interlinked by diodes D_3 and D_4. Because of these diodes, the output with the highest potential determines the extent to which capacitor C_{20} is charged via resistor R_4. Network R_{23}-C_{19} has been added to ensure that fast signal fluctuations are passed on very rapidly.

Capacitor C_{20} is discharged slowly via resistor R_{25}, so that the control circuit returns to its default setting when no or a smaller input has been applied for some time. The potential across C_{20} is buffered by IC_{4a}, while IC_{4d} ensures that the (fixed) default level is added to the signal. The resulting control signal is applied to the control input (pin 7) of IC_1 and IC_2.

With component values as specified, the compression is 10:1; in other words, a 20 dB change at the input results in a 2 dB change at the output.

The setting of P_1 depends on the signal level at the input of the circuit. Since this level varies largely from one sound card to another, the design provides a wide control range.

SUPPLY LINES

As mentioned earlier, the circuit is powered by a standard 12 V mains adaptor, which is applied directly to the output amplifier. All other circuit elements are supplied with a regulated 10 V potential. This voltage is produced with the aid of current source T_2-T_3 and zener diode D_6.

The reference voltage of 5.6 V is produced with the aid of current source T_1 and zener diode D_5.

CONSTRUCTION

The circuit is best built on the printed-circuit board shown in **Figure 3** (see Readers Services towards the end of this issue). Start the construction with placing audio sockets K_1–K_3, the three wire bridges, and all solder pins, and follow these with first the passive components, and then the active ones. Mind the polarity of the electrolytic capacitors, diodes, transistors, and ICs. After it has been fitted, set the preset to minimum volume (anticlockwise).

Solder the output leads from the standard mains adaptor from which power is derived to the relevant pins on the board. If the board is to be housed in an enclosure, a plug-and-socket arrangement should be used for linking the output from the adaptor to the board.

Check that the output voltage of the adaptor does not rise above 18 V with small loads.

When all is connected, the circuit can be tested. Passive loudspeakers may be linked directly to the LSP output terminals, but active ones should be connected to the line output terminals.

Finally, connect a sound source, for instance, the line output of a sound card or the output of a Walkman™ to the input of the circuit and adjust P_1 for the desired volume. From then on, any fluctuations in the signal input level will be minimized automatically.

[980023]