A N A U D I O C O M P R E S S O R

By Luiz Amaral
PY1LL/AC2BR

A problem observable when we use the computer in voice communication, normally via VOIP programs, is that the audio level which for certain microphones and sound boards can be insufficient even with the microphone gain in its maximum. This could be solved easily with a low-noise pre-amplifier.

Another problem we note on the computer and on the hamradio transmissions are the audio level variations with different distances between the microphone and the operator or with different voice levels of the latter.

In this case it is necessary the use of an automatic gain control circuit for compensating the audio level variations, keeping this the most constant as possible. There are many commercial (or from amateurs) designs that execute well that task.

My idea of designing a compressor was based in the fact that great part of the existing designs uses many discrete active components, like bi-polar or FET transistors, etc that, in general, have their parameters values with much great spread among samples. The integrated circuits, for example the operational amplifiers, have less spread of the characteristics among units due their construction and their operation with negative feedback.

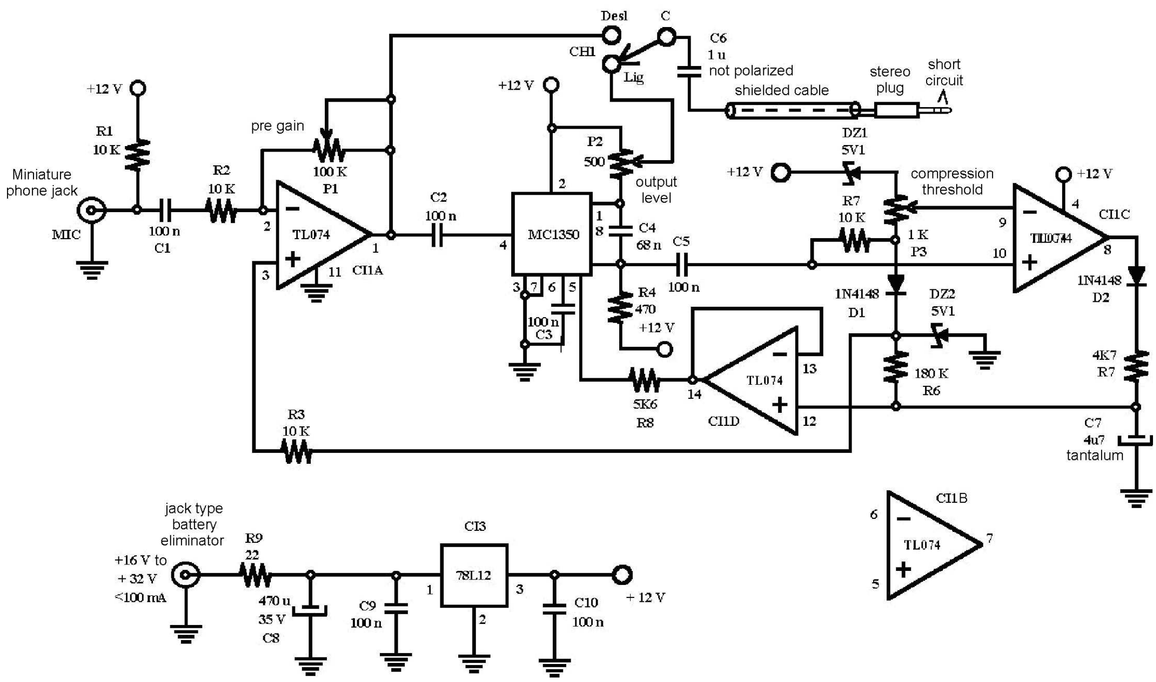
Willing to use an integrated circuit as automatic gain control amplifier, I searched the Internet for many, always verifying their existence in the Brazilian market. Some were technically fine for the project, but, or very expensive or difficult to find, especially in Brazil.

Thus, I decided to use the integrated circuit MC1350 from Motorola that, although easily findable, was not developed for audio, but for video IF amplification.

Its gain, more than the necessary and the possible operation from DC to video IF frequencies, suggested that it would be perfect to be uses in audio if some care were taken.

In the figure we see the schematic diagram of the compressor. A quad operational amplifier TL074 with FET inputs (only three used indeed) besides the MC1350 and a low power 12V voltage regulator 78L12. The input is designed for electrets microphone, commonly used in computers, but the compressor must work fine with other types too.

AUDIO COMPRESSOR



The signal is pre-amplified by the TL074 first section (CI1A) and excites the MC1350 (CI2) that has two outputs: one, with DC isolation, that is the signal final output and another one that is amplifies by a TL074 section (CI1C) with open mesh, that is, with maximum gain. The output of this section is half-wave rectified by D2 (there was not the need of full-wave rectification) and filtered by resistor R7 and the tantalum (for less loss) capacitor C7.

The DC voltage so got is lead to the gain control input of CI2 under correct impedance (≈5KΩ) through the third section of the TL074 (CI1D). The diodes DZ1, DZ2 and D1 are used for correct DC polarization needed by the IC's.

As the signal reaches determined threshold (compression threshold), the DC voltage got from its peak is delivered to the gain control of CI2, decreasing it: greater the signal, smaller the gain of the circuit. This maintains the output rather constant (from the threshold). This is controlled by P3 and obviously depends on the adjustment of the pre-gain (P1). The output level is adjusted by P2 and its level will depend on the net gain of the audio system we want to use (computer/sound board, radio transmitter, etc). Not having a professional finality, the attack and recovery times are fixed and determined by resistors R7 and R6 and capacitor C7.

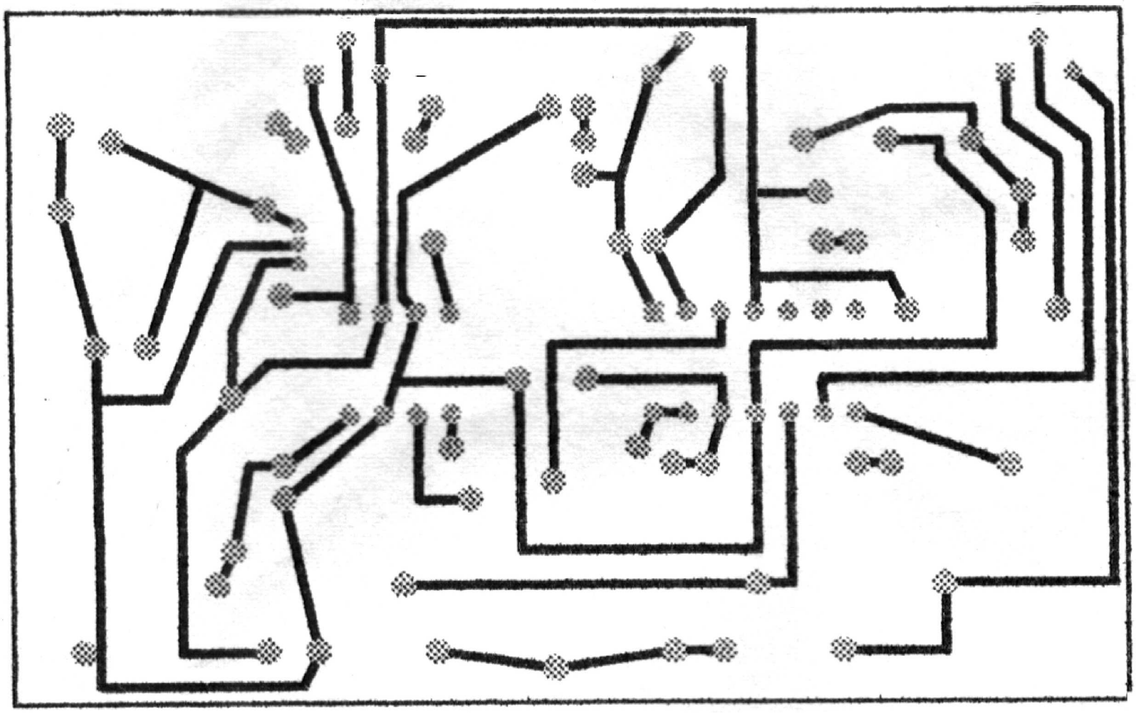
Here an advice: as CI2 was designed for frequencies of some tens of megahertz, its operation in audio below the compression threshold, that means, with maximum gain, is instable and presents unwanted oscillations. The capacitor C4, connecting both phase opposite outputs, eliminate such oscillations while limits the audio high frequencies response.

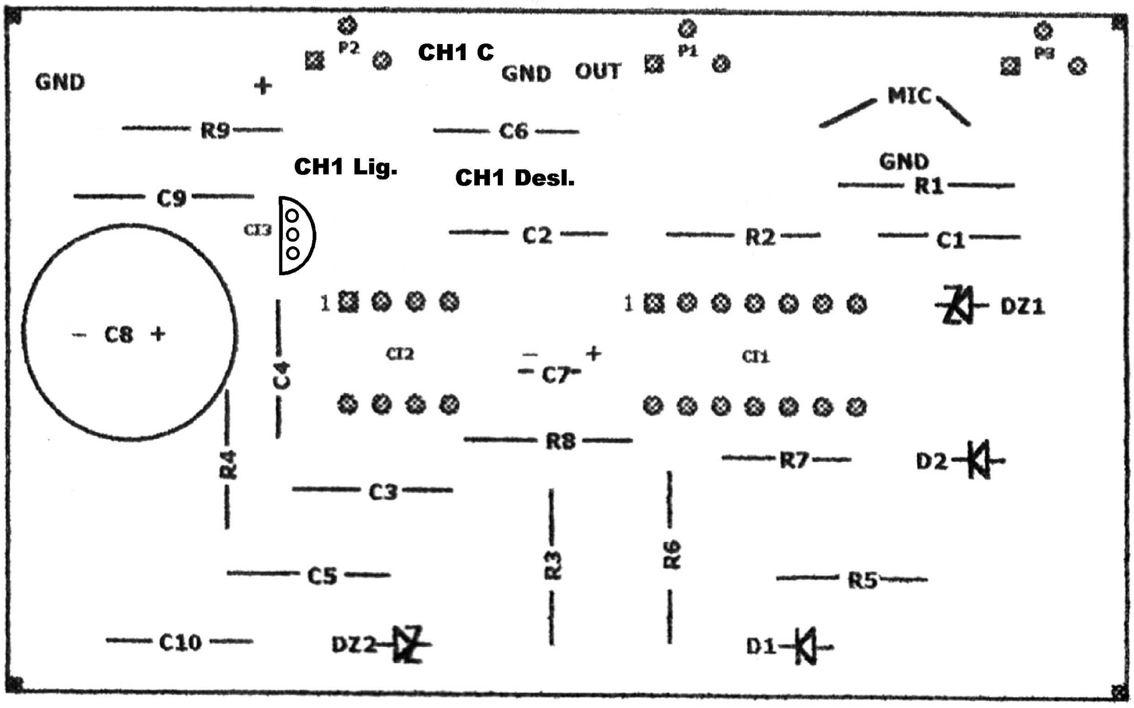
The MC1350 was designed for a 12V DC supply. I’ve tried (for operation with computers) so, to use the +12V that exists in PC’s. The result was simply disastrous due the great digital noise present on those DC lines. The necessary RC filtering introduced an unavoidable voltage drop that impaired the IC operation. I had, with no other simple choice, to use a +12V battery eliminator (that, under low consumption, has an output much higher than that) and a +12V regulator (CI3), getting a noise free regulated +12V supply.

The assembly, in my computer use case, was made with a universal printed board attached to a metallic face retired from an old analog modem; so, my compressor is mounted in the back of the computer box and inside it as it were a normal internal accessory.

A female microphone connector (identical to the sound board one) is used as input and a shielded cable coming from the circuit output and with a male connector (similar to the microphone one) is connected to the soundboard microphone input.

So, the signal from the microphone passes through the compressor before reaching the soundboard. Clearly one can install a convenient switch to eliminate the compressor in the signal path, if we desire to use the microphone in its not-compressed conventional mode. This switch was installed in my prototype, as a printed circuit was developed for the project as shown in the figures below. I have not tested it because my prototype is in a ‘spider assembly’. Moreover, the three potentiometers in my case are only trimpots, as I don’t feel the need to alter their adjustments once established.





GND
+
P2
CH1 C
GND
OUT
P1
P3
MIC
R9
C6
CH1 Lig.
CH1 Desl.
GND
R1
C9
CI3
C2
R2
C1
1
1
DZ1
- C8 +
C4
CI2
- C7 +
CI1
R8
R7
D2
R4
C3
R3
R6
C5
R5
C10
DZ2
D1