

Personal Sound to Light Unit

small but sophisticated

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Disco nights are great fun from time to time but you don't want to overdo it in regard of the sound volumes you're exposed to for a couple of hours. Arguably there's no less pleasure in enjoying dance music in the privacy of your home, study or student digs. However, the true disco feeling is not obtained without a matching sound to light unit, so here's a really small version.

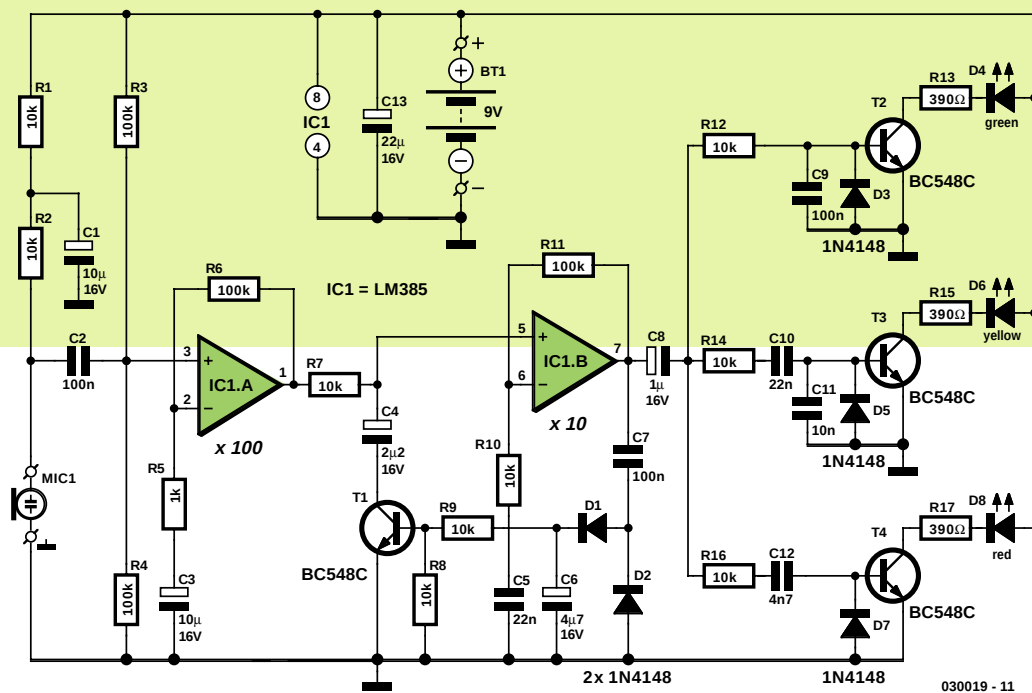


Figure 1. The circuit of the sound to light unit consists of a controlled-gain amplifier followed by a 3-way frequency splitter.

A sound to light unit converts music signals into light pulses. In most cases, three channels are used to cover different frequency ranges. The bass channel will then indicate the 'beat' of the music by a more or less rhythmical flash, while the two other channels represent the higher frequency ranges of the music channel.

The 'private' version of a sound to light unit discussed in this article employs three coloured LEDs instead of powerful flashing lights or floodlights as used in most discos. For the rest, it has practically the same functionality as the 'real thing' experienced on Saturday nights. However, a direct connection to the music amplifier is not necessary as the unit we've in mind has been designed to pick up the sound information through a microphone. Another peculiarity of the circuit is the automatic sensitivity adaptation to the music volume detected. In this way the circuit can work without any controls.

The circuit draws an average current of just 20 mA and works happily off a 9-volt PP3 block battery. However, the low current requires super bright LEDs to be used to ensure sufficient brightness.

Amplify it

High amplification is required to enable the tiny signals produced by an

electret microphone to be turned into bright flashes from LEDs. The circuit diagram in **Figure 1** shows that we employ two operational amplifiers with a total gain of about 1000 times. The input of the first opamp (IC1.A) is connected to the electret microphone capsule via coupling capacitor C2. The microphone is given a certain DC bias level obtained from the 9-V supply rail by resistors R1 and R2. R1 together with electrolytic capacitor C1 decouple the supply voltage for the sensitive microphone while R2 equals the microphone impedance. Capacitor C2, then, ensures that the microphone bias level does not appear at the opamp input. In other words, it will only pass the alternating component, which is caused by sound picked up by the microphone. The + input of the opamp has its own bias voltage supplied by potential divider R3-R4. Because the two resistors have the same value, the supply voltage is effectively halved, i.e., 4.5 V exists at the junction of R3 and R4 (assuming a nominal 9 V supply). This bias voltage will also exist at the output of the first opamp and, because of R7, at the input of the second one (IC1.B) whose output will also copy this dc level. In this way, R3 and R4 keep both opamps biased at half the supply voltage. Both opamps are used in the non-inverting configuration

hence provide unity (≈ 1) dc gain.

For alternating signals, however, the gain is much greater. In the case of the first opamp, the gain is determined by the ratio between resistors R6 and R5, or R11/R10 for the second opamp. Just look at the relevant resistor values and you'll discover that IC1.A is configured for a gain of 100 and IC1.B for a gain of 10.

The signal level at the output of IC1.B is rectified by D1 and D2, smoothed by C6 and then used to drive n-p-n transistor T1. To the signal voltage at the output of IC1.A, the combination of R7 and T1 looks like a voltage divider. With rising signal levels, the rectified voltage on C6 also rises and the transistor is driven harder because of the larger base current supplied by R9. The result is a lower resistance in the transistor and consequently a lower AF signal behind R7.

You may wonder why we did not use an n-p-n transistor without direct current in the collector circuit. Alternatively you might have expected to see a FET at this position, its drain-source junction acting as a controlled resistance. It is less known that virtually the same function may be obtained from a regular switching/AF transistor like the ubiquitous BC548C. An n-p-n transistor, too, represents a variable resistance that can be controlled with

Components list

Resistors:

R1,R2,R7-R10,R12,R14,R16 = 10kΩ
R3,R4,R6,R11 = 100kΩ
R5 = 1kΩ
R13,R15,R17 = 390Ω

Capacitors:

C1,C3 = 10μF 16V radial
C2,C7,C9 = 100nF

C4 = 2μF 16V radial
C5,C10 = 22nF
C6 = 4μF 7 16V radial
C8 = 1μF 16V radial
C11 = 10nF
C12 = 4nF7
C13 = 22μF 16V radial

Semiconductors:

D1,D2,D3,D5,D7 = 1N4148
D4 = LED, green (see text)

D6 = LED, yellow (see text)
D8 = LED, red (see text)
IC1 = LM385N (with socket)
T1-T4 = BC548C

Miscellaneous:

BT1 = 9V PP3 (6F22) battery with clip-on lead
MIC1 = electret microphone
PCB, available from The PCBShop

hin a certain range. However, for a low-distortion volume control, only a tiny signal level (of the order of millivolts) may be applied to the collector. This condition is not satisfied here as the output voltage is regulated to about 1 V_{pp}. If the second stage has a gain of 10, about 100 mV_{pp} can be found at the collector. At such a level, distortion occurs that will not be acceptable in other applications. No problem for the sound to light unit, however, because the output signal is used to control LEDs than drive an audio amplifier. If you do want to use such a 'volume control' for audio applications, you should make sure a much smaller signal level is handled, which is probably easiest realised by moving the automatic volume control towards the circuit input.

Filter

The output signal supplied by the second opamp drives the LED controls by way of simple filters. Each LED driver stage consists of a transistor (T2, T3 and T4). To prevent the transistor bases from being charged with negative levels, anti-parallel diodes (D3, D5, D7) are used on each base terminal. Each transistor is only actuated on the positive half cycle of the drive voltage. Overall, however, the higher frequencies do cause an impression of average brightness.

The filters consist of simple RC networks broadly dimensioned for a cut-off frequency using the formula

$$f_c = 1 / (2 \pi R C) \quad [\text{Hz}]$$

For example, the low-pass section R12/C9 is dimensioned for about 160 Hz using 10 kΩ (R12) and 100 nF (C9). All lower frequencies in the music signal will therefore pass through this 'channel'. The mid-tone channels contains a combined high-pass / low-pass

filter R14/C10+C11. The treble channel is driven by a simple high-pass R16/C12. If necessary the cut-off frequencies may be changed to suit individual requirements and that's easiest done by making small changes to the capacitor values.

Printed circuit board

If the circuit of a relatively simple design, the actual construction of the sound to light unit is made even simpler by a printed circuit board (Figure 2). The two opamps we've discussed are contained in a single IC type LM385, which is best fitted in an IC socket (look at the notch in the IC body). The single-sided PCB has no wire links. No problems are expected to arise if you watch the polarity of the diodes, electrolytic capacitors and

LEDs and work carefully all the way. The electret (or 'condenser') microphone insert can be almost any available type as long as it has two terminals.

The maximum LED current amounts to about 18 mA so the circuit is suitable for LEDs described as 'standard', 'bright' and 'superbright'. Using the latter the effect of the 3-way filtering is more pronounced, while for a relatively small room with not too much light ordinary LEDs will be found to be perfectly suitable. If you decide to use low-current LEDs, the value of series resistors R13, R15 and R17 should be increased to 3.3 k or even 3.9 k. The resulting reduction in current consumption will enable your 9-V battery to last much longer.

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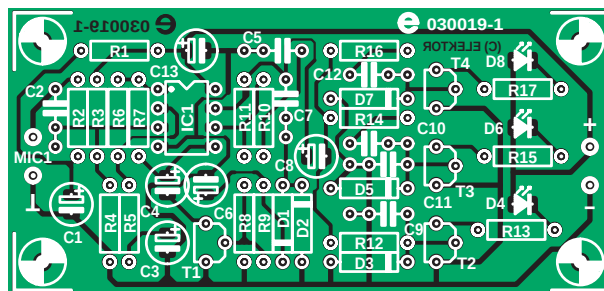


Figure 2. The single-sided PCB has no wire links.

