Ultrasonic Sound Detector

track down high frequencies

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Using the circuit described here you can quickly and simply check whether harmful sounds with a high frequency are being produced somewhere nearby, especially important at locations where the so-called ultrasonic teen-deterrents against loitering youths are installed. Ten LEDs indicate the sound pressure of signals with frequencies between 16 and 40 kHz.

These days, to harass loitering youths into leaving, an 'ultrasonic' sound system is often used (such as the Mosquito). Such a system generates, in the vicinity of a popular gathering place, a sound with frequencies between 17 and 20 kHz. This sound appears to be very irritating to youths, with the result that they soon move on. Older people appear to be unable to hear these frequencies and are therefore not bothered by them.

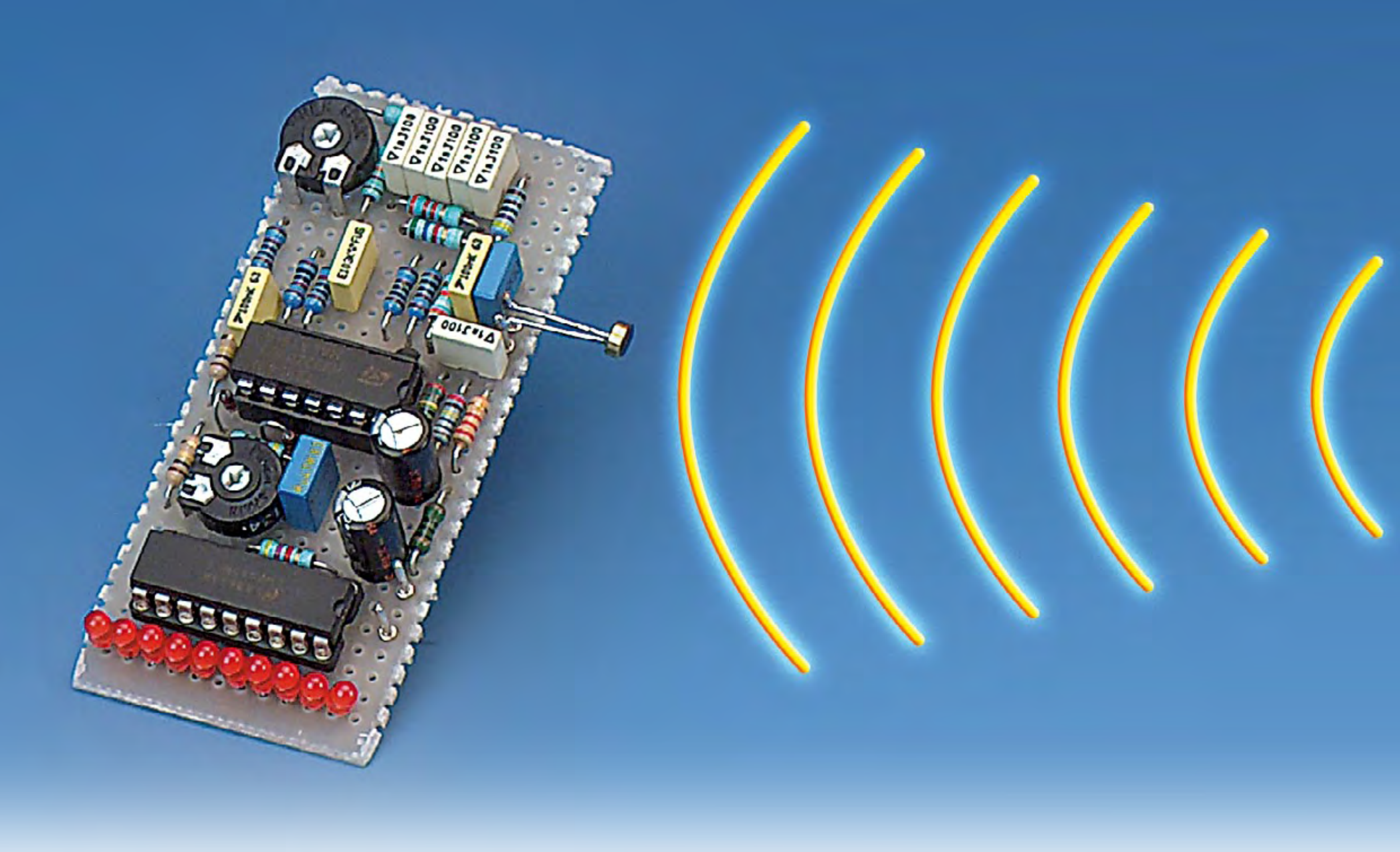
The term 'ultrasonic' is actually not quite correct. Ultrasonic sound is normally used to describe the range of frequencies that are above the range of human hearing, that is, above 20 kHz. Because the frequencies used are below 20 kHz, they are not only audible to loitering youths (insofar as they don't have any hearing damage), but they are even better heard by babies, small children and pets. The sound pressure is not all that high, according to data provided by the manufacturer. But if the ultrasonic deterrent is nevertheless considered to be annoying, it is very likely to be even more irritating to babies and small children (and that is not taking into account the potential hearing damage by prolonged exposure).

Babies and pets cannot say what is wrong (with the exception of parrots perhaps). So if you are walking through the city with a baby there is the chance that the child is exposed to these types of sound without the parents realising.
So that you can quickly get an indication whether harmful sounds with a high frequency are being produced somewhere, we designed an indicator that can detect these signals and measures their intensity. Ten LEDs show the sound pressure of sound in the frequency range from 16 to 40 kHz. Lower frequencies are suppressed by a fifth-order, high-pass, Butterworth filter so that the indicator will not react to speech and other everyday sounds. When designing the measuring range for the indicator the decision was made for maximum deflection at a sound pressure of 90 dB. If you are exposed to 90 dB for more than an hour then there is the risk of permanent hearing damage. Whether you can also suffer hearing damage from frequencies you cannot hear any more, wear of the auditory ossicles for example, has not been researched yet.

Schematic

The circuit shown in **Figure 1** may look quite involved, but it is really not that bad. The circuit comprises a microphone amplifier, a fifth-order high-pass filter, an active rectifier, a buffer stage and finally the display driver with ten LEDs. For driving these LEDs we used the old, trusty LM3915 in dot-mode (we deliberately did not choose bar-mode because of the current consumption). This IC is configured for its simplest application with REF ADJ connected to ground. In this configuration the input signal has to be 1.28 V_{DC} (typically) for 'full deflection'.
Working backwards from this value and the sensitivity of the electret-microphone that is used, we can calculate the required gain. For the microphone we chose a type that can be obtained from Farnell (the Kingstate type KEEG1542TBL-A). This choice was mainly determined by the frequency response curve shown in the datasheet (which can be downloaded courtesy Farnell). This shows a slight increase towards 20 kHz, which leads us to conclude that it will still be usable at frequencies a little over 20 kHz. While there may be a lot of ripple in the curve above 20 kHz this is not a problem since we're only looking for an indicative measure of sound pressure. At 90 dB the microphone module will give an output of about 8 mV. When we take into account that the rectifier contains a peak detector (makes a difference of $\sqrt{2}$ in the gain) and that the sensitivity of the microphone can vary by up to ±3 dB, we therefore need a circuit with a minimum gain in the region of 80 to 160.

The input contains a gain stage (IC1A) which amplifies 40 times. This also cuts off the lowest frequencies at the same time (C1 and C4). C3 and C5 compensate a little for the slight drop of frequency response above 20 kHz. For the opamps the ST type TS924IN quad rail-



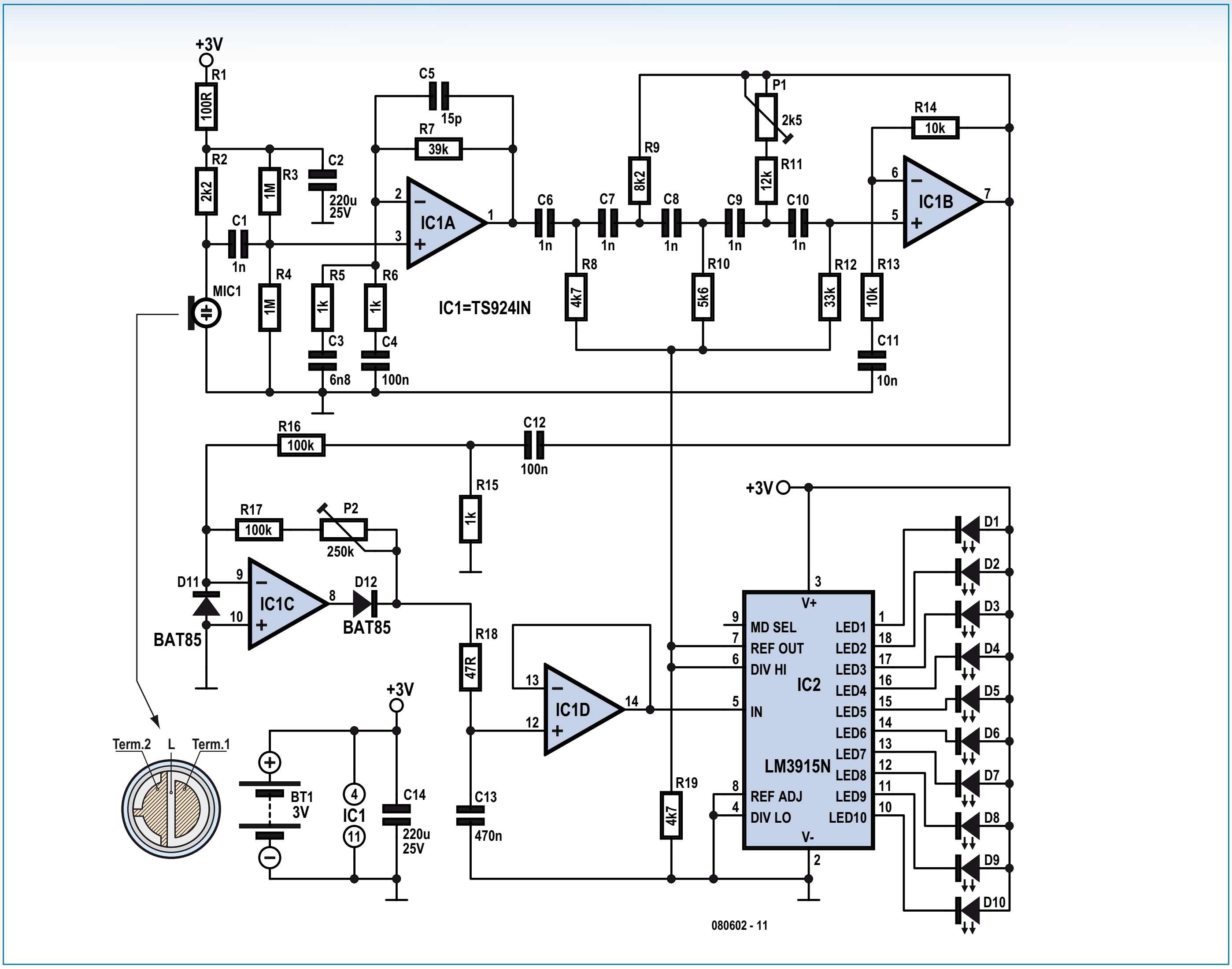


Figure 1. With this circuit you can very easily track down high-frequency audio signals.

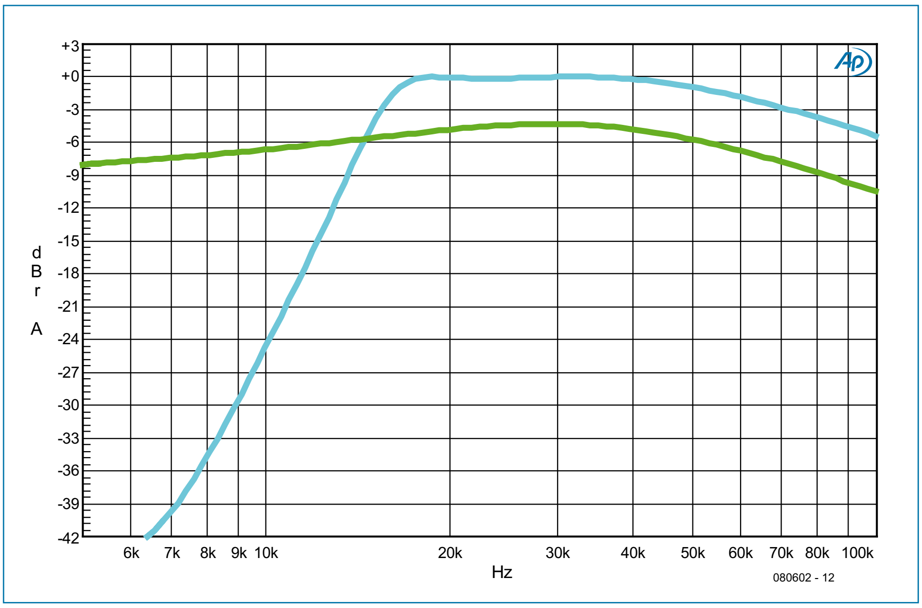


Figure 2. The frequency characteristic of the input amplifier on its own (green) and the input amplifier followed by the high-pass filter (blue).

to-rail device was selected, which can deliver no less than 80 mA. This opamp has a relatively large GBW (gain-bandwidth product) of 4 MHz, so that the bandwidth of the input amplifier at a gain of 40 is still 100 kHz (C5 is only for RF decoupling), more than enough for this application. This is followed by a steep filter built around IC1B that passes only frequencies above 16 kHz (–3 dB). The measurement curves (**Figure 2**) show the amplitude response of the input amplifier on its own and that of the input amplifier followed by the filter. For the proper operation of the fifth-order filter a buffer is not enough, but the opamp needs to have a gain of 2 (this can be seen clearly in the measurement curves). This is a fortunate circumstance since the other stages now don't need to have as much gain.

The filter contains an adjustable component to allow for the compensation of component tolerances. P1 can be used to make the curve as straight as possible. If you do not have the means to measure this then you can use the nominal value of 870 Ω for P1 instead. It is still recommended to measure and select C6 through C10 for equal values.
The rectifier around IC1C is a standard implementation, where the usual diode between the output of the opamp (pin 8) and the inverting input (pin 9) is not necessary because the output cannot go negative. To ensure correct operation, the rectifier is AC-coupled (R15/C12) to the output of the filter. D11 protects the input of the opamp from excessive negative input voltages.
The output voltage of the rectifier (cathode of D12) charges capacitor C13 to the peak value via R18. The purpose of R18 is to limit the maximum charging current. With the values as shown, C13 is charged relatively quickly, the RC time-constant is only 22 µs. The total amplification of the circuit can be adjusted over a wide range with P2 so that other (less sensitive microphones) can be used as well. The discharge time of C13 varies somewhat with the setting of P2, but that is not important here.
The fourth opamp (IC1D) is used as a buffer to drive the LM3915.
Although the resolution of the LM3915 is 30 dB (10 × 3 dB) but because of the simple design of the rectifier the voltage across C13 is not exactly proportional to the input signal. The bottom LED D1 lights up at –21 dB compared to the voltage required to turn on the topmost LED (D10).

The current consumption of the whole circuit is between 11 and 15 mA, good for six days of continuous use. The circuit appears to work down to nearly 2 V, so that instead of alkaline batteries rechargeable batteries could be used as well. This is much more environmentally friendly.

If you like to experiment, you could try to make a small parabolic reflector so that you can pinpoint the source of the ultrasonic sound. The microphone needs to be mounted at the focal point of the reflector. Perhaps even a small horn is already enough...

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