THD analyser for audio circuits

This article describes a spot frequency audio distortion analyser, designed and built by a reader, Laurie Tunnicliffe of Mulgrave, Victoria. Measurements can be made at 100 Hz, 1 kHz and 10 kHz, and the final resolution of the instrument is 0.01%.

IN RECENT YEARS there has been a trend towards considering the transient behaviour of audio circuits rather than their steady-state behaviour. Although the attention given to this side of circuit design is not unwelcome, total harmonic distortion (THD) analysis is far from redundant.

While the transient behaviour is a go/no go situation, the THD behaviour is a measure of how well a circuit will perform. For instance, it is not a question of to what degree an amplifier will slew limit, or to what extent the internal loop will overload; these transient characteristics are barriers, and until reached will have no effect on the amplifier's performance.

THD measurements are therefore still a valuable analytical tool when developing new circuits or measuring and giving figures of merit to existing circuits.

When a single frequency is passed through an amplifier with a non-linear transfer curve, other frequencies are produced. These other frequencies are integer multiples of the test signal and sum of these is the THD.

There are a number of ways of measuring the harmonic distortion of an amplifier. The most recent is the 'Fourier Analyser', which is a computer-based instrument that samples the output waveform and performs the necessary mathematics (Fourier transform) to break down the waveform into its component parts. These instruments however cost tens of thousands of dollars. A technique becoming popular is the use of a spectrum analyser, which is a swept bandpass filter. The results are displayed on a CRO. THD is then calculated from

$$THD = \sqrt{\left(\frac{F1}{F}\right)^2 + \left(\frac{F2}{F}\right)^2 + \left(\frac{F3}{F}\right)^2 \cdots}$$

N.B. Square root sign in formula applies to sum of all terms.

Another technique used by Quad and described elsewhere[1], compares the input and output of an amplifier (using a differential amp), the difference being distortion. In Quad's experiment, music is used as the source and a monitor amp/speakers are used to listen to the distortion played by itself. This then becomes a 'real-time' distortion analyser, and any non-linearities, whether transient or steady state, are revealed.

The last and most commonly used method is to eliminate the fundamental signal and read the resultant harmonics on a moving coil meter. This is sometimes called 'Noise and THD' measurement, as hum and 'electronic noise' are lumped together with the harmonies. This is the technique used for the instrument presented here.

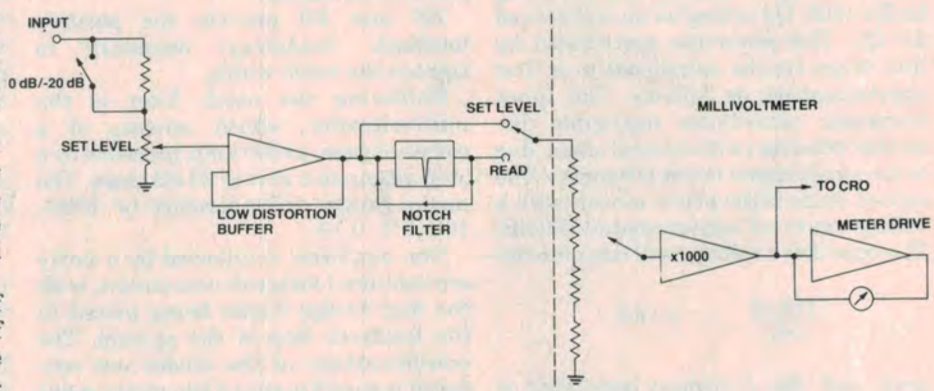


Figure 1. **Block diagram of the THD analyser described in this article.**

Block diagram

Refer to Figure 1 for the instrument's block diagram.

The input is applied to a buffer stage via a 0 dB/−20 dB attenuator. This allows easier control of the 'set level pot' when large signal levels are being measured. The buffer stage provides a low impedance source to drive the notch filter. The design of the buffer is of major consideration, as any non-linearities it contributes will be indistinguishable from the signal being tested.

There are a number of options available when designing a notch filter. A derivative of the Wien bridge was chosen due to its simplicity and the fact that it only requires two variable reactances to balance.

Bootstrapping around the filter is necessary to tighten up the notch width. Without this, the attenuation of the second harmonic would be excessive. With the amount of bootstrapping used the second harmonic is attenuated by less than 1 dB.

The notch filter is followed by a millivoltmeter and reads the average value of the harmonics relative to the fundamental. The meter is calibrated to read full scale for 0.775 V RMS input, and therefore the meter can be used separately to measure dBs into a 600 ohm load, relative dB (dBV) or millivolts, giving the instrument a dual function.

A CRO output is taken after the x1000 amplifier, so that the residual harmonics can be investigated. This will often give considerable insight into the cause of the distortion.

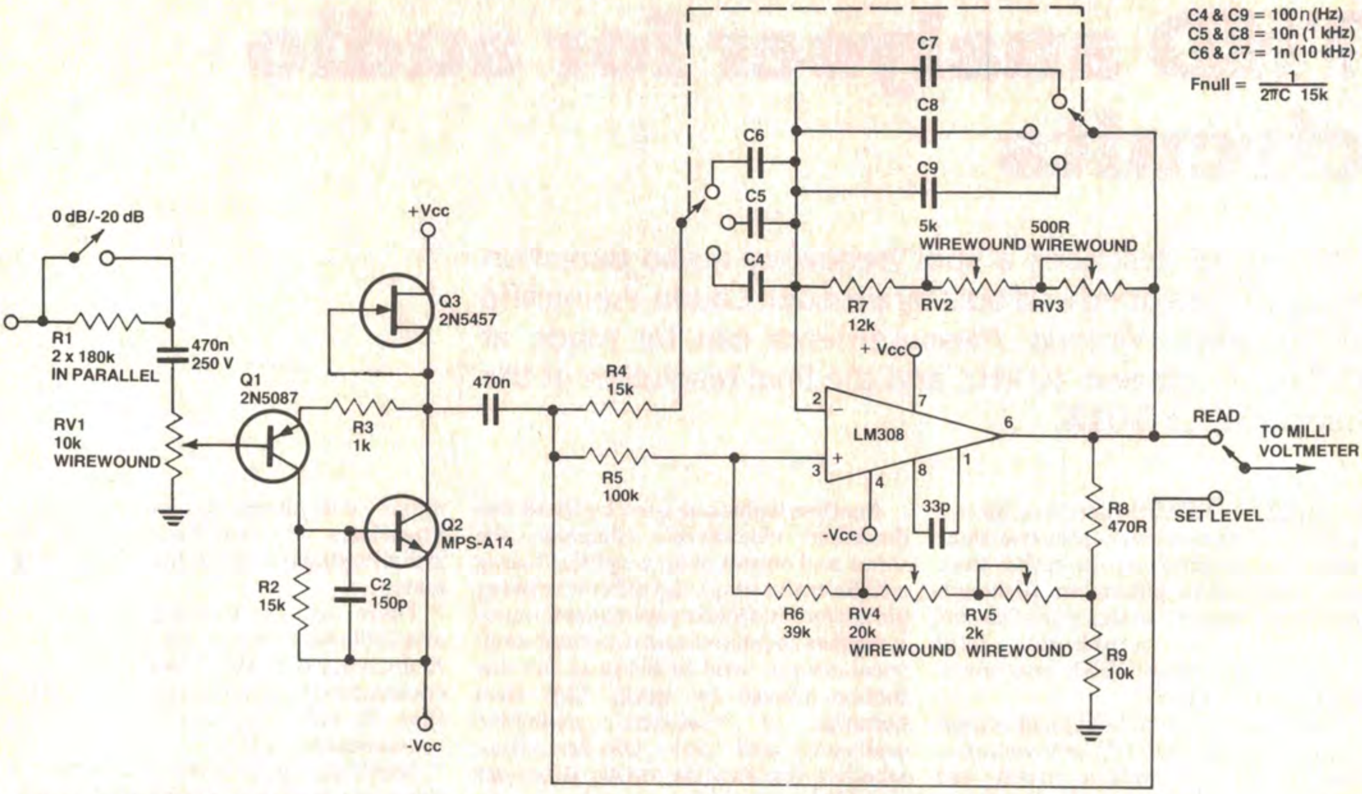


Figure 2. **Circuit diagram of the THD analyser (above and opposite).**

Circuit description

By examining Figure 2 the complete circuit diagram can be understood.

Q1 and Q2 form the non-inverting buffer with Q3 acting as an active load for Q2. The distortion contributed by this stage can be calculated to a first approximation as follows. The input transistor contributes negligible distortion relative to the second stage, due to the small signal levels it handles. The second stage is the prime mover, with a voltage gain of approximately 60 dB. The base drive voltage will therefore be

$$\frac{.775 \times \sqrt{2}}{1000} = 1.1\,\text{mV}$$

peak and the distortion generated is 1.1% second harmonic[2]. Since the buffer stage has 100% feedback (unity gain), the loop gain is also 60 dB and the distortion is reduced to

$$\text{loop gain} = \frac{1.1}{1000} = .0011\%$$

The buffer feeds the notch filter, which may be looked upon as a frequency dependent differential amp. At the notch frequency, the parallel arm and the series arm balance to give the same impedance ratio as the resistive arms. The input then appears as a common mode signal to a differential amp, and the output is zero. The common mode rejection ratio of the op-amp is of particular importance; however, most op-amps have a CMRR of at least 80 dB, which is sufficient to give a 0.01% resolution.

R8 and R9 provide the positive feedback (bootstrap) necessary to tighten the notch width.

Following the notch filter is the millivoltmeter, which consists of a constant gain, x1000 amp, preceded by a step attenuator giving 20 dB steps. The meter ranges will therefore be 100%, 10%, 1%, 0.1%.

The amplifier is followed by a fairly conventional meter-driven circuit, with the four bridge diodes being placed in the feedback loop of the op-amp. The non-linearities of the diodes are rendered insignificant and the meter reads accurately, even at the low end of the scale. Diodes 5 and 6 are used to protect the meter from an overload.

In order to achieve a wide enough bandwidth for the x1000 amp, an externally compensated op-amp was necessary. The op-amp used in the meter circuit is a dual low-noise device, and therefore helps to keep the instrument noise level low and reduces parts count.

C12 rolls off the frequency response above 70 kHz. This helps improve the square wave response by reducing any ringing, and also reduces high frequency noise. This will allow measurement up to the seventh harmonic of 10 kHz, and should prove adequate.

Construction

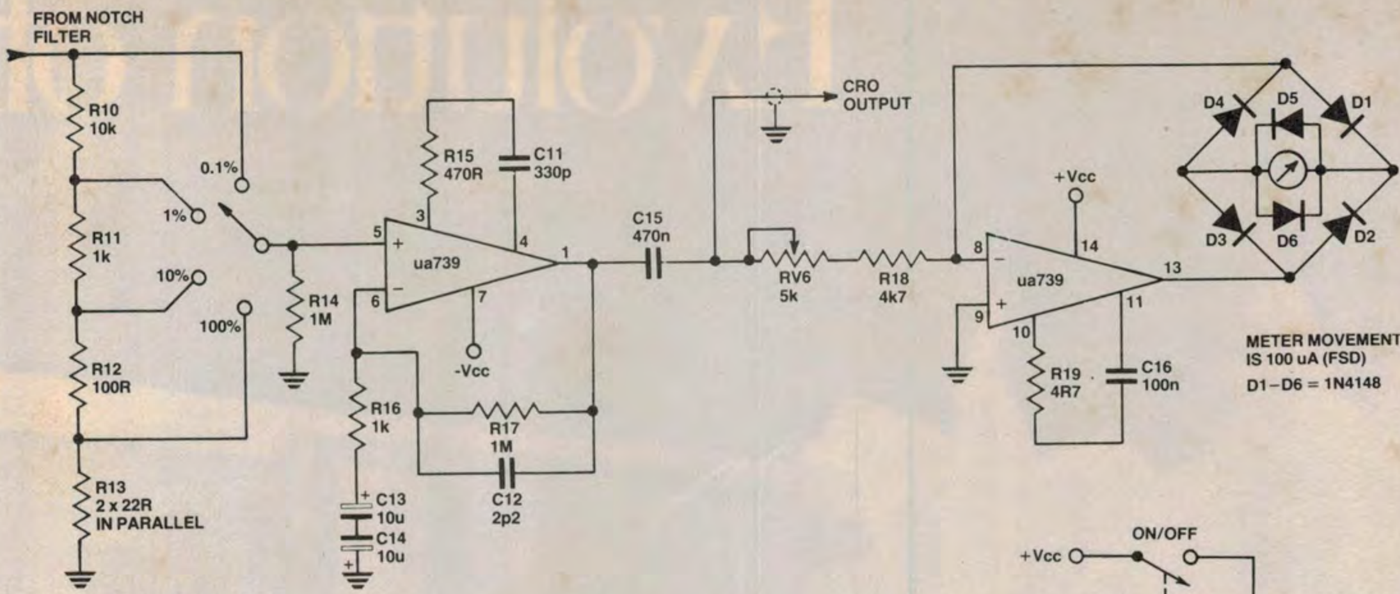
If accuracy and stability are to be reproduced, all components must be close tolerance, high stability. I used ¼-watt metal film resistors, which are available from Dick Smith. The filter capacitors are either styroseal or green caps; ceramic capacitors should be avoided as they are voltage dependent.

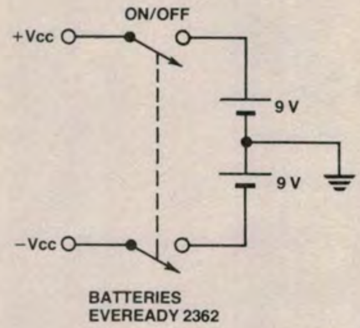
Veroboard should also be avoided, as stray capacitance across the strips causes problems when you are looking for one part in 10 000.

I used tagstrip and wire-wrap sockets for the ICs and found this easy to work with, giving satisfactory results.

My prototype was built in a diecast box; however, I had to rewire it three times, using shielded cable for every connecting wire, before a stable layout was found. For this reason I strongly suggest building the instrument as two separate units – a notch filter and a millivoltmeter. This will also give the versatility of being able to use one or the other independently.

The nulling pots are wirewound, having the advantage of a continuous track. The continuity of a wirewound pot is limited by the fact that a step in resistance equivalent to one wire winding is the smallest change possible. Carbon pots are certainly worse, as they sometimes jump resistance value mid-track. The fine control pots are single turn and provide reasonable ease of nulling at low levels. However, if you





feel it is worth the extra cost, ten-turn (spiral wound) pots will alleviate having to be careful when adjusting the controls.

The only adjustment to be done is the calibration of the meter. This is accomplished by applying a 0.775 V RMS, 1 kHz signal to the input with the unit switched to 'set level', and adjusting RV6 for full-scale deflection of the meter. Alternatively RV6 and R18 can be changed to 1k and 6k8 (fixed values) with less than 1% error in meter reading.

Operation

The instrument is operated as follows:

- Set Level/Read switch to Set Level
- Range switch to 100%
- Adjust Set Level pot for FSD of meter
- Set Level/Read switch to Read
- Adjust the nulling pots for minimum meter reading
- THD is now read from the meter and range switch.

The nulling procedure is accomplished by starting with the coarse pots, and alternately adjusting them for minimum meter deflection until it becomes difficult to proceed. The process is then repeated with the fine pots. If the user has a CRO available, this will assist the nulling procedure.

Figures 3, 4 and 5 show some oscilloscope photographs of the input and output of the meter. Figure 3 is an under-biased class B amplifier and shows spikes in the residual. This is a common waveform from class B amplifiers and the meter can be used to set the bias level for optimum.

Figure 4 shows second harmonic distortion from a voltage-driven common emitter amplifier. The input signal was 1.1 mV peak and the meter reading was 1.1%. This confirms the analysis used for the buffer stage.

Figure 5 is third harmonic caused by a thermistor-stabilised Wien bridge oscillator. THD was 0.02% at 100 Hz. (This is one of the problems encountered when using a thermistor at low oscillator frequencies.)

It should be noted that a low distortion oscillator will need to be used when making measurements. The residual distortion of the oscillator may set the lower limit to the measurements.

Some performance measurements

Final resolution of the meter was 0.005% at 100 Hz and 1 kHz and 0.01% at 10 kHz. Below this, drift in components' values with temperature, circuit noise, distortion introduced by the buffer and the filter stage and CMRR limitations of the filter all take their toll. However, distortion values of less that 0.01% are purely academic in my view, regardless of what hi-fi manufacturers' sales departments would have us believe.

References
1. P.J. Baxandall, Wireless World, November 1977, pp. 63-66.
2. E.F. Taylor, Wireless World, August 1977, pp. 28-32.

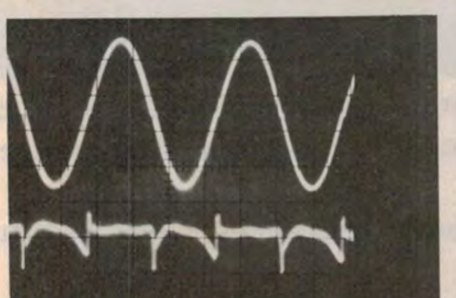


Figure 3. **Class B amp, meter reading 1.5%.**

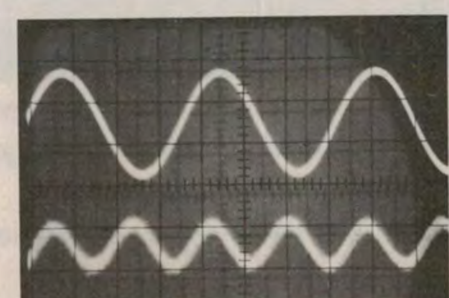


Figure 4. **Second harmonic distortion, meter reading 1.1%.**

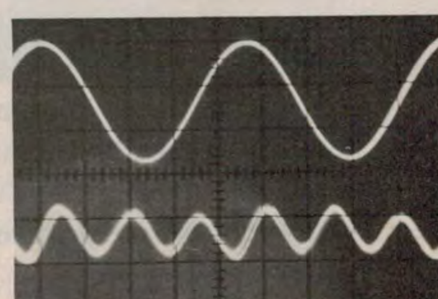


Figure 5. **Third harmonic distortion, THD 0.02%**