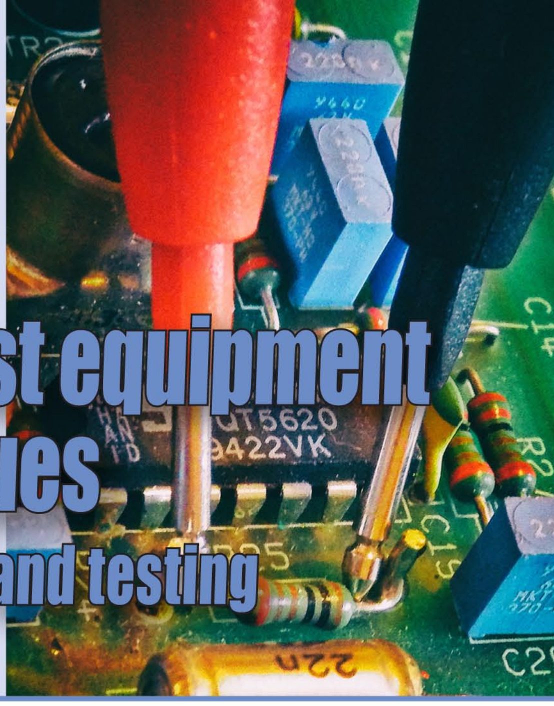
Teach-In 2018

Get testing! – electronic test equipment and measurement techniques

Part 6: Audio frequency measurement and testing

by Mike Tooley



Welcome to *Teach-In 2018: Get testing! – electronic test equipment and measurement techniques*. This *Teach-In* series will provide you with a broad-based introduction to choosing and using a wide range of test gear, how to get the best out of each item and the pitfalls to avoid. We'll provide hints and tips on using, and – just as importantly – interpreting the results that you get. We will be dealing with familiar test gear as well as equipment designed for more specialised applications.

Our previous *Teach-In* series have dealt with specific aspects of electronics, such as PICs (*Teach-In 5*), Analogue Circuit Design (*Teach-In 6*) or popular low-cost microcontrollers (*Teach-In 7* and *8*). The current series is rather different because it has been designed to have the broadest possible appeal and is applicable to all branches of electronics. It crosses the boundaries of analogue and digital electronics with applications that span the full range of electronics – from a single-stage transistor amplifier to the most sophisticated microcontroller system. There really is something for everyone in this series!

Each part includes a simple but useful practical *Test gear project* that will build into a handy gadget that will either extend the features, ranges and usability of an existing item of test equipment or that will serve as a stand-alone instrument. We've kept the cost of these projects as low as possible and most of them can be built for less than £10 (including components, enclosure and circuit board).

This month

In this, the sixth part of *Teach-In 2018*, *In theory* looks at the specifications used for audio frequency (AF) equipment, including power amplifiers, preamplifiers, filters and tone controls. *Gearing up* introduces AF test equipment and techniques, while *Get it right!* helps you avoid test and measurement pitfalls and provides useful hints and tips that will help you to improve the accuracy and relevance of your measurements. Finally, our sixth *Test gear project* is a low-distortion 1kHz signal source that can be used to carry out a variety of useful tests and measurements.

In theory: AC measurement

Several different parameters are used in performance specifications for audio equipment such as power amplifiers, preamplifiers, filters and tone controls. Many are obvious, but a few need a little explanation. Those that readers will doubtless already be familiar with include:

- Voltage gain (expressed as a voltage ratio or in dB)
- Input and output impedance
- Frequency response (or bandwidth limits)
- Output power.

Several other parameters may be less obvious – they include:

- Phase/frequency response
- Damping factor (DF)
- Transient response (rise/fall time and slew rate)
- Cross-talk
- Distortion (HD, THD and IMD)
- Hum and noise
- Signal-to-noise ratio (SNR).

Not every parameter is important in a particular application, so before we look at how these specifications are measured it is worth explaining what each means and how it impacts on the performance of a particular item of audio equipment. To put these parameters into context, Table 6.1 lists the specifications

Table 6.1 Specification for a high-performance audio power amplifier

Parameter	Quoted specification
Number of channels	Two (note 1)
Rated power output	75W into 8Ω at less than 0.03% THD (notes 2 and 3)
Voltage gain	28dB
Input sensitivity for rated output	1V
Frequency response	20Hz to 20kHz at -1dB and 5Hz to 100kHz at -3dB
Phase response	+5°/–15° over the frequency range 20Hz to 20kHz
Signal-to-noise ratio	Greater than 120dB over the range 20Hz to 20kHz
Total harmonic distortion (THD)	Less than 0.03% (note 4)
Intermodulation distortion (IMD)	Less than 0.03% (note 4)
Rated load impedance	4Ω to 16Ω
Damping factor	Greater than 400
Slew rate	Better than 50V/μs
Input impedance	28kΩ
DC output offset	Less than ±5mV

Notes
1. Left (L) and right (R) channels of a stereophonic system
2. Per channel with both channels driven
3. Full-bandwidth power (FTC) is 60W
4. At all power levels up to, and including the rated output power.

Fig.6.1. Some prototype amplifiers, filters and tone controls ready for testing

of a high-performance audio power amplifier.

Voltage gain

The voltage gain provided by an amplifier is simply the ratio of output voltage to input voltage (where input and output voltages are both specified in the same units – eg, both RMS or both peak-peak values).

Input sensitivity

The input sensitivity of an amplifier is the input voltage needed to produce the amplifier's rated output.

Input and output impedance

The input impedance of an amplifier is the impedance 'seen' looking into the amplifier's input. Similarly, the output impedance of an amplifier is the impedance 'seen' looking back into the amplifier's output. This means that the output impedance of an amplifier is the internal impedance of the amplifier at its output. On page 38 we explain this concept in more detail by using an equivalent circuit.

Frequency response

The frequency response of an amplifier is usually expressed in terms of its lower and upper cut-off frequencies. 'Cut-off' is a confusing term since it might imply that there is no amplification below and above the lower and upper cut-off frequencies. This is not the case, and the term simply refers to a given reduction in output voltage or power. In most cases, the cut-off frequency is taken to mean the frequency at which the output voltage has fallen to 70.7% of its mid-band value (this is the point at which the output *power* falls to exactly half of the mid-band value).

Output power

The output power produced by an amplifier is the maximum power that it can deliver under linear conditions (ie, with a sinusoidal input and output). When this is the case we can express the power delivered to a load (of given impedance or resistance) in RMS watts. Since distortion increases with output power this parameter is often specified at a specific level of distortion (eg, 10W for 0.1% distortion or 15W for 1% distortion).

To overcome the ambiguities associated with the specification of audio output power, several standards have been introduced. The notion of 'continuous average power' was first introduced in the US and several other countries as a means of dispelling some of the myths associated with audio power measurement and its specification. The term defines the output of an amplifier when delivered on a sustained basis and the Federal Trade Commission (FTC) standard, originally published in 1974, requires that, before measuring the output power, an amplifier should be 'preconditioned'. The FTC states that this process should involve operating the amplifier with all channels simultaneously driven at one third of rated output power for a period of one hour. The FTC standard also specifies the temperature (25°C) at which the measurement should be carried out and further states that it should be carried out at 'all frequencies within the power band' and 'without exceeding the rated maximum percentage of total harmonic distortion' (THD). Importantly, the FTC requires that the rated power output should be sustainable for a period of not less than five minutes.

The most recent international standard for specifying amplifier performance is defined in IEC (BS EN) 60268 Part 3, *Sound system equipment: Amplifiers*. Published in 2013, this standard applies not only to conventional analogue amplifiers but also to the analogue parts of analogue/digital amplifiers that form part of a sound system for professional or household applications. The standard specifies the characteristics that should be included in specifications of amplifiers and the corresponding methods of measurement. In particular, it defines methods for measuring the short-term, long-term and temperature-limited output power of an amplifier.

Phase response

The output signal of an amplifier may not rise and fall in sympathy with its input, and there will often be an amount of angular shift (expressed in degrees) between the two signals. For example, if the output signal from an amplifier is inverted when compared with its input there will be 180° of phase shift between the input and output signals. The input signal is usually taken to be the reference signal and the output may shift forwards (leading) or backwards (lagging) over a range of frequencies.

Damping factor

The output impedance of a power amplifier is usually very small (a fraction of an ohm) but is rarely specified and is difficult to measure. Instead, damping factor (DF) is used as a measure of the output impedance relative to that of the load (usually a much higher value). DF is simply the ratio of the load impedance (often 4Ω, 8Ω or 15Ω) to the amplifier's internal impedance (usually less than a tenth of an ohm). In matched systems (such as 600Ω line amplifiers) the input and output resistances will both be the same, yielding a unity DF.

Transient response

Transient response is important in high-quality and wideband amplifiers, and it can be expressed in several different ways including rise/fall time and slew rate. Rise time is the time taken for the output voltage to swing from its most negative value to its most positive value, whereas fall time is the time taken for the output voltage to swing from its most positive value to its least positive value. Rise time is conventionally measured between the 10% and 90% points of the leading edge of a positive-going transient, while fall time is measured between the 10% and 90% points of a falling-edge transient. Both times should be very small (ideally no more than a few tens or hundreds of nanoseconds). An alternative measure is based on the rate at which a perfect (ie, zero time) transient voltage rises or falls at the output of an amplifier. In effect, the specified parameter is the rate of change of voltage with time.

Cross-talk

When more than one channel is present within an amplifier there may be some leakage of signal from one channel into the other. This can arise from various causes, including inadequate screening and poor power supply design. Cross talk is usually measured in decibels (dB) and is the ratio of signal from one channel appearing in the other. Cross-talk figures are usually high, with values of 90 to 120dB being typical.

Distortion

The two most common forms of non-linear distortion are harmonic distortion (HD) and intermodulation distortion (IMD). HD is often specified in terms of the *total* amount of harmonic distortion present (THD). In both of these forms of distortion extra frequency components are added to a signal due to non-linearity of an amplifier's transfer characteristic. If an amplifier had a perfectly linear transfer characteristic there would be no HD and no IMD.

Distortion is usually expressed in decibels (dB) or as a percentage of the rated output. Even-order harmonic distortion is generally caused by an asymmetrical transfer characteristic whereas odd-order harmonic distortion is caused by a symmetrical non-linearity. IMD arises from different signal frequency components mixing together to produce new frequency components.

Hum and noise

Hum and noise within an amplifier can also be added to a signal. These are also unwanted components present at the output that are not present at the input. Hum is simply the appearance of a signal at mains supply frequency (or twice the mains supply frequency). Hum can be carried on supply voltage rails, where it appears as a small AC signal

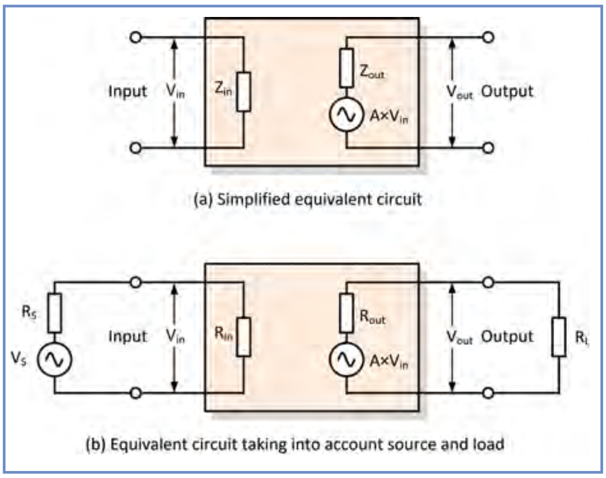


Fig.6.2. Equivalent circuits of an amplifier

superimposed on the DC supply. It can also find its way into an amplifier when stray magnetic fields (such as those that surround power transformers) induce current into nearby wiring. Hum can be reduced, if not completely eliminated, by good power supply design, using screened signal cables, and by adequate screening and grounding of chassis and chassis-mounted components.

Noise is a random fluctuation superimposed on a wanted signal. Unfortunately, all electronic components produce noise – but some produce more noise than others. The amount of noise that they produced depends not only on the type, construction and material used in the component, but also on the electrical conditions under which it is operated (ie, current and voltage) as well as, very significantly, the temperature. Noise is a particular problem within high-gain amplifiers where noise generated in the first stage receives the full benefit of the gain provided by the subsequent stages.

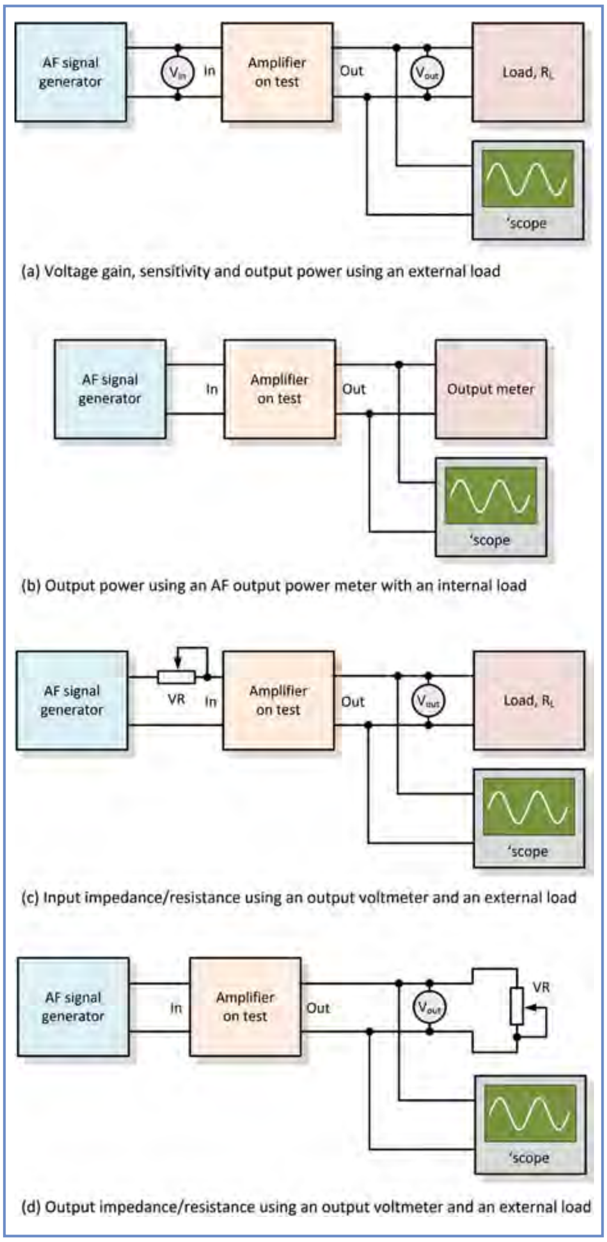


Fig.6.3. Arrangements for measuring basic amplifier specifications

Signal-to-noise ratio

Signal-to-noise ratio (SNR) is the ratio of wanted signal to the amount of noise present and is usually expressed in decibels (dB). Since distortion will invariably also be present and can be difficult to separate during measurement, a more practical measure is often used. SINAD, or 'signal in noise advantage', is a measure of the signal quality that takes into account the presence of distortion. SINAD is calculated from the ratio of total output signal power (ie, signal power plus noise power plus the power arising from distortion components) to the noise-plus-distortion power.

As might be expected, SINAD performance is only slightly worse than the SNR performance but the SINAD figure is usually a more reliable measure of performance, particularly when a significant amount of noise and distortion is present. Note that, since the response of the human ear favours the middle frequency range (from about 500Hz to around 5kHz), noise is often specified A-weighted in relation to its effect within this band.

Equivalent circuit of an amplifier

Before making meaningful measurements on an amplifier it is useful to have an idea of what goes on 'inside the box' and what happens when the 'box' is connected to a source and load. For this, we can make use of an equivalent circuit, like that shown in Fig.6.2(a). This simplifies the internal circuit to just three components and ignores the effect of changes at the output affecting the input. The three components present are:

- Input impedance, Z_{in}
- Output impedance, Z_{out}
- Voltage source, $A_V \times V_{in}$

When the amplifier is driven from a source and connected to a load, the equivalent circuit becomes a little more complicated and is shown in Fig.6.2(b). It assumes that in the mid-band frequency range the reactive components are negligible, so we've further simplified the circuit by replacing the impedances with their corresponding resistances. Thus, the voltage at the input of the amplifier will be:

$$V_{in} = V_{in} \times \frac{R_{in}}{R_{in} + R_S}$$

While at the output of the amplifier it will be:

$$V_{out} = (A_V \times V_{in}) \times \frac{R_L}{R_L + R_{out}}$$

If $R_{in} >> R_S$ and $R_{out} << R_L$ then the overall voltage gain (G) will be given by:

$$G = \frac{V_{out}}{V_{in}} = \frac{A_V \times V_{in}}{V_{in}} = A_V$$

Measuring amplifier specifications

Measuring voltage gain, sensitivity and frequency response

To measure the voltage gain, sensitivity and frequency response of an amplifier, a sinewave signal at a frequency in the centre of the amplifier's mid-band range (usually 1kHz) is applied to the input, and the amplifier is adjusted for an undistorted output by observing the output waveform using an oscilloscope. The amplifier should be connected to a suitably resistive load (not a loudspeaker) as shown in Fig.6.3(a). RMS-reading AC voltmeters should be used to measure the input and output voltages from which the voltage gain can be calculated (see earlier). Similarly, the sensitivity can be measured by adjusting the amplifier for its rated output power and measuring the corresponding input voltage (any gain or volume control fitted to the amplifier must usually be set to maximum before making this measurement).

The arrangement shown in Fig.6.3(a) can also be used to measure an amplifier's frequency response. The output frequency of the AF signal generator is adjusted over the amplifier's full working range and, at each end of this range the frequency at which the output voltage falls to 70.7% of its mid-band value is located. These are the two 'cut-off' frequencies. If the amplifier's response is

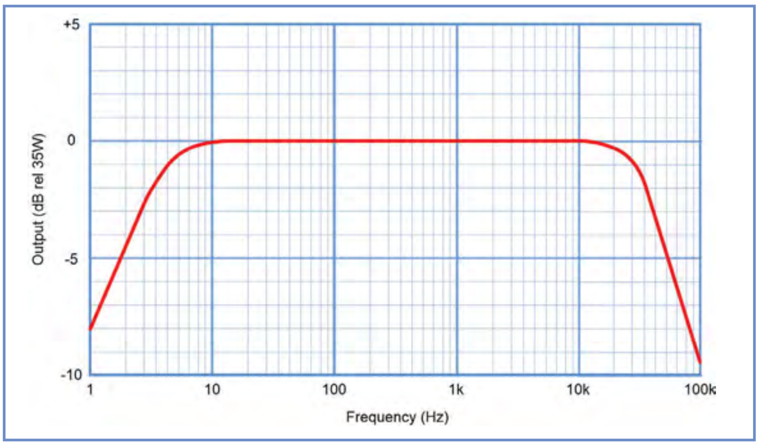


Fig.6.4. Measured frequency response of a Technics SU-Z22 power amplifier

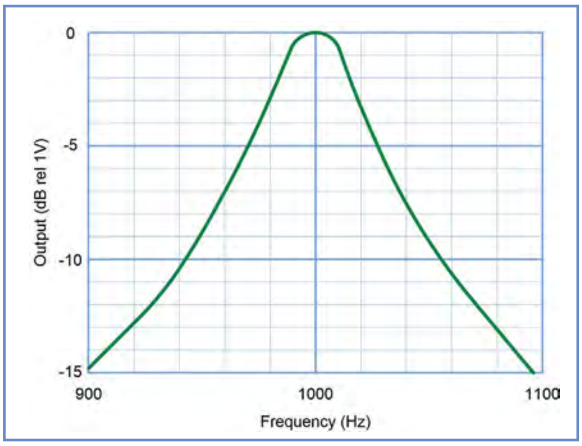


Fig.6.5. Measured frequency response of a 1kHz filter (ref: 0dB at 1kHz)

not substantially 'flat' then it is advisable to take a series of readings and plot the frequency response as a graph. Some typical frequency response graphs are shown in Fig.6.4 and Fig.6.5.

Output power

Output power should be measured under sinusoidal conditions and should be the power that the amplifier can deliver on a continuous basis. Since values of voltage and current measured are both expressed in terms of 'root mean square' quantities, this power is sometimes referred to as 'RMS power'. These values can be read from a voltmeter or ammeter calibrated for sinewave operation (as is invariably the case with instruments that are designed to make conventional AC power line measurements). The term 'RMS power' is, however, somewhat misleading since the power that is indicated by this measurement is actually the *average* power over a cycle of the wave. When this power is dissipated in a resistor it appears as heat.

To put this into context, let's assume that you have a load of known resistance (not a loudspeaker) connected to an amplifier, and that the amplifier is supplied with a sinewave signal source within the middle of the audio band (usually 1kHz). We also need to ensure that the output of the amplifier (the voltage or current that we are going to measure) is not distorted (ie, that it is truly sinusoidal). In order to do this we would need to observe the output waveform, checking with an oscilloscope that it has not become clipped or otherwise distorted. If we now measure the RMS AC voltage we can determine the output power (P_{out}) from:

$$P_{out} = I_{out} \times V_{out} = \left(\frac{V_{out}}{R_L}\right) \times V_{out} = \frac{V^2_{out}}{R_L}$$

If, for example, we had used an 8Ω load and measured an undistorted 10V RMS developed across it we would be able to determine the power output from:

$$P_{out} = \frac{V^2_{out}}{R_L} = \frac{10^2}{8} = \frac{100}{8} = 12.5\text{W}$$

If we had used an oscilloscope (instead of an AC meter) to measure the output voltage we would probably find it much easier to measure the peak-to-peak value of the waveform. We can then convert this value, $V_{out(pk\text{-}pk)}$, to an RMS value and use that value in our calculation, as follows:

$$P_{out} = \frac{V^2_{out}}{R_L} = \frac{\left(\frac{V_{out(pk-pk)}}{2\sqrt{2}}\right)^2}{R_L} = \frac{\left(V_{out(pk-pk)}\right)^2}{8R_L}$$

Putting this into context with some typical figures, let's assume that the oscilloscope indicates a peak-peak voltage of 20V with a load having a resistance of 4Ω. The output power would be calculated from:

$$P_{out} = \frac{\left(V_{out(pk-pk)}\right)^2}{8R_L} = \frac{20^2}{8 \times 4} = \frac{400}{32} = 12.5\text{W}$$

Measuring output power

The arrangement for measuring output power is similar to that used for voltage gain and sensitivity measurement, as shown in Fig.6.3(b). If an AF output power meter is available then this can usually provide the load required, but if such an instrument is unavailable a separate external load will be required. The test load should not only be purely resistive, but it should also be adequately rated in terms of power dissipation. For most practical purposes this means that one or more high-power resistors will be required, and using a suitable combination of series and/or parallel components it is possible to provide several different load resistances.

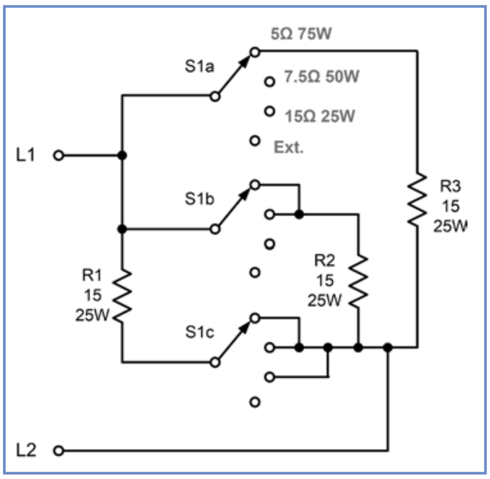


Fig.6.6. Circuit of the output load

The circuit of a simple but effective load is shown in Fig.6.6. The load can be switched to provide resistances of 5Ω, 7.5Ω and 15Ω at maximum power ratings of 75W, 50W and 25W respectively. To realise these values we used three high-power aluminium-clad wire-wound 15Ω resistors. Each resistor should be rated at 25W when mounted and used according to the manufacturers recommendations (note that the manufacturer suggests de-rating by 50% when the resistors are not used with a heat-dissipating surface (ie, a heat sink) or when they are used at a high ambient temperatures. The resistors are Arcol part number 'HS25 15RJ' (or similar) and they are available from several electronic component suppliers including Rapid Electronics (stock code 62-8430). Note that the switched 5Ω, 7.5Ω or 15Ω resistances offered by our output load should prove satisfactory with amplifiers rated for 4Ω, 8Ω or 15Ω loads.

Measuring input impedance/resistance

An arrangement for measuring input resistance is shown in Fig.6.3(c). A variable resistor or decade resistance box – 'VR' – is inserted at the input. This resistor is initially set to zero and the amplifier is adjusted for normal operation (without clipping or distortion evident from the waveform displayed on the oscilloscope). The output voltage is then noted, after which VR is adjusted until the output falls to exactly half this value. At this point, the value of resistance is measured (or read from the decade resistance box). The input

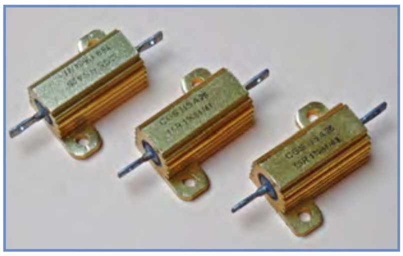


Fig.6.7. The three 15Ω 25W wire-wound resistors used in the switched output load

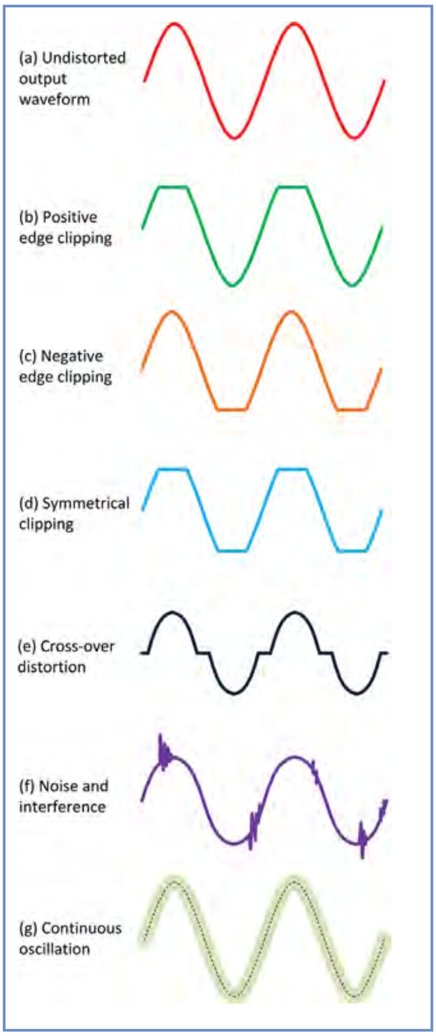


Fig.6.8. Using a sinewave signal to test for various distortion conditions

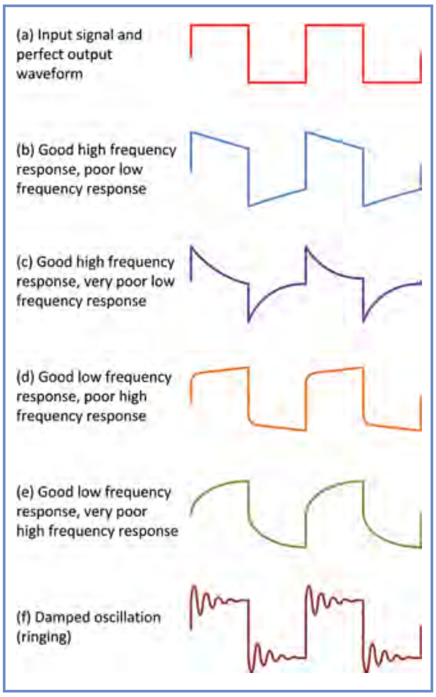


Fig.6.9. Square-wave testing can provide a rapid means of checking the frequency response and transient performance of an amplifier

resistance of the amplifier can then be calculated from: R_{in} = VR + R_s

Here, R_s is the source impedance of the signal generator (often 600Ω or 50Ω).

Measuring output impedance/resistance

An arrangement for measuring output resistance is shown in Fig.6.3(d). A variable resistor or decade resistance box (VR) is used as a load. This resistor is initially left disconnected (ie, open-circuit) and the amplifier is once again adjusted for normal operation (but without a load present). The output voltage is then noted, after which VR is reconnected and adjusted until the output falls to exactly half its previous value. At this point, the value of resistance is measured (or read from the decade resistance box). The output resistance (R_{out}) can then be calculated from: R_{out} = VR. Note that this method is *unsuitable* for use with power amplifiers because they have an extremely low output impedance and the excessive current demand may cause damage to the output stage.

Waveform testing

If you have a signal generator and an oscilloscope handy you can carry out a very quick check on the performance of an amplifier by simply observing the output waveform produced when a signal is applied to the input of the amplifier. The amplifier will need to have a load connected (see above) and the test signal frequency should normally be 1kHz at a level that preserves linearity and avoids any risk of over-driving the amplifier. Two different tests can be applied; one using a sinewave input and the other using a square wave. The former provides a quick check on linearity and distortion, while the latter can be used for a quick assessment of frequency response.

Sinewave testing

A quick inspection of an output waveform will usually provide you with a clue as to what type of distortion has been introduced by an amplifier, but first it is necessary to ensure that the input signal is free from distortion and this involves checking that it is a reasonably pure sinewave. Most signal generators (see page 43) are able to produce sinewave outputs with sufficiently low levels of distortion that cannot be discerned by the human eye when a waveform is viewed on an oscilloscope. As the level of distortion increases – typically to about 5% or more – the distortion starts to become visible in the form of a departure from a pure sinewave shape. This makes it possible to carry out a quick check on distortion by simply viewing the shape of the output waveform.

Fig.6.8 shows some representative test waveforms. The undistorted sinusoidal input signal is shown in Fig.6.8(a). This is also the ideal shape for the output waveform, which should be perfectly sinusoidal if the amplifier is not introducing any distortion. The effect of clipping is shown in Figs.6.8(b) to 6.8(d). In the case of Fig.6.8(b) the positive edge of the waveform has been clipped. Notice the flattening effect this has on the positive excursions of the signal. Fig.6.8(c) shows a similar effect applied to the negative edge of the waveform. These two conditions usually point to an incorrect bias adjustment where an applied signal becomes increasingly distorted (and clipped) such as whenever the amplitude of the input signal exceeds a certain value.

Symmetrical clipping (ie, clipping of both positive and negative peaks) is illustrated in Fig.6.8(d). This condition usually results from applying an input signal of excessive amplitude. Reducing the amplitude below a critical value (ie, below the point at which clipping starts to occur) will often correct the problem and reduce the distortion to an acceptable amount. Fig.6.8(e) shows cross-over distortion which can be a problem when insufficient standing current is available in a complementary push-pull output stage. Note how the output signal remains at zero until the input signal reaches a certain value and how this form of distortion affects both positive and negative-going half cycles of the input waveform.

Apart from the repetitive waveform defects that we've seen thus far, a signal might also become contaminated by random fluctuations in signal level (noise) and sudden spikes of interference caused by switching and other transients induced into wiring (both internal and external). This problem is illustrated in Fig.6.8(f).

Last, you might occasionally come across a circuit that, while intended to act as an amplifier, also acts as an oscillator! The reason for this is that, at some frequency (usually very much higher than the highest designed signal frequency) the internal phase shift becomes such that the feedback becomes positive (instead of the intended negative feedback), consequently de-stabilising the amplifier and resulting in continuous oscillation. Designers of high-gain amplifiers usually try to avoid this problem by using only local feedback (ie, feedback over a single stage). This effect is illustrated in Fig.6.8(g). Note that the amplitude of the parasitic oscillation is often significantly reduced due to the limitations of the frequency response of the oscilloscope used to observe the distorted waveform.

Square wave testing

An alternative to using a sinewave for testing an amplifier is using a square wave. This can provide you with a rapid assessment of the frequency response of an amplifier. Square waves comprise an infinite number of sinusoidal harmonic components added to the fundamental sinewave, and so any defects in the frequency response of an amplifier will show up very quickly from an examination of the shape of the waveform of the output signal when a square wave input is applied. As a result, it is possible to assess whether the frequency response is good or poor (a perfect square wave output would correspond to a perfect frequency response).

Fig.6.9 shows waveforms that correspond to several different frequency response characteristics. A perfect square wave output, Fig.6.9(a), indicates that the amplifier under test has a flat frequency response. Figs.6.9(b) and 6.9(c) are typical of an amplifier having a good high-frequency response coupled with a poor low frequency response, while Figs.6.4(d) and 6.4(e) are indicative of poor high-frequency response and good low-frequency response.

The damped oscillation, or 'ringing', shown in Fig.6.9(f) occurs when the amplifier's step response (ie, its response to a sudden and very rapid change in signal level) produces momentary oscillation. This can occur in amplifiers where appreciable inductive reactance is present at the same time as capacitive reactance. The combined effect of these two opposite reactances results in a resonant effect, where the sudden changes imposed by the rising or falling edges of the square wave signal causes energy to oscillate back and forth between the two opposite reactive components. The transfer of energy between the two components decays due to losses (resistance) present in the circuit and so the oscillation eventually settles to a steady value at one or other extreme of amplitude.

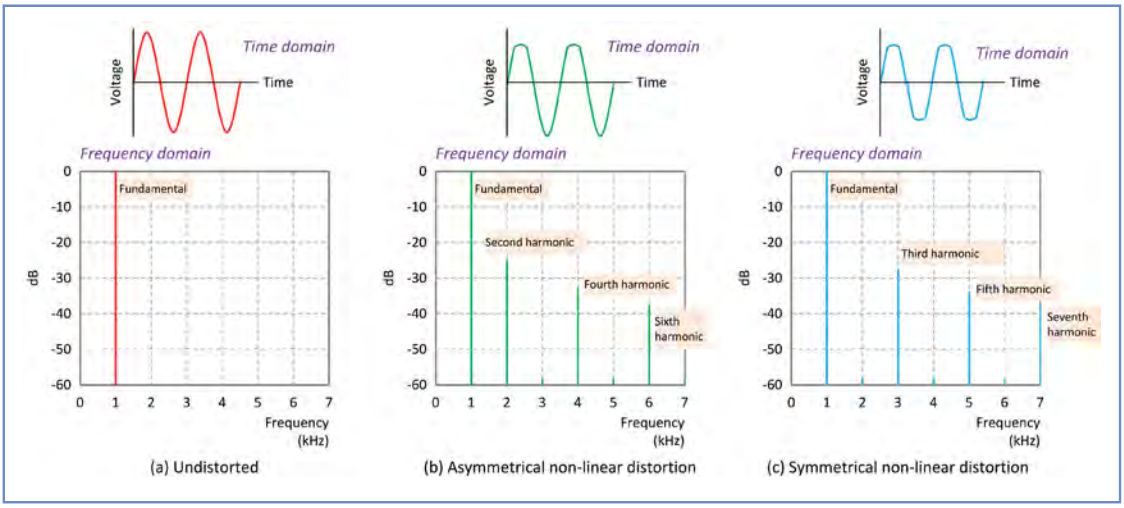


Fig.6.10. Effect of a non-linear transfer characteristic on the quality of an output waveform

Measuring noise and distortion

There are several different forms of distortion and all of them may, to a greater or lesser extent, be present at the same time. If an amplifier has a perfectly linear transfer characteristic (ie, voltage out plotted against voltage in) and a perfectly flat frequency response (ie, voltage out plotted against frequency) it will not produce any distortion (it might, however, be susceptible to hum and noise, as we explain later). Conversely, if the frequency response and/or transfer characteristic is imperfect then this will result in the production of distortion but this may, or may not, be a problem depending upon the severity of the non-linearity and the degree of aberration in the frequency response. Let's first consider the effect of non-linearity in the transfer characteristic.

Distortion due to transfer characteristic non-linearity

Fig.6.10(b) shows what will happen when the transfer characteristic is asymmetrical. In this case, the characteristic flattens-off beyond a particular positive-going input level. The result is that the output signal becomes prematurely truncated or 'clipped'. A similar effect would be produced if the negative-going part of the transfer characteristic had become flattened while the positive-going part remained linear. However, in this case the negative edge of the output signal would be clipped. The resulting harmonics produced will include the second, fourth and sixth, and so on, as shown in Fig.6.10(b). None of these components are present in the undistorted pure sinewave signal shown in Fig.6.10(a).

Fig.6.10(c) shows the effect of symmetrical non-linearity in the transfer characteristic. In this case, both positive and negative edges of the output waveform have become clipped. The resulting harmonics produced will include the third, fifth, seventh, and so on, as shown in Fig.6.10(c). Notice also that the output amplitude has become reduced when compared with the undistorted output waveform shown in Fig.6.10(a).

Note that an increase in the amplitude of a test signal will usually result in a significantly greater increase in the level of harmonic distortion and the amplifier becomes 'overdriven'. The problem will increase in severity with a further increase in input signal to the point that the distortion produced will very quickly reach an unacceptable level. Fig.6.11 shows the effect of overdriving a power amplifier on the amount of total harmonic distortion generated. Note the rapid increase in THD when the rated output power (8W) is exceeded.

Distortion due to aberrations in frequency response

As well as distortion resulting from non-linearity of its transfer characteristic, an amplifier will also introduce distortion due to imperfections in its frequency response. It can be a sobering experience to apply a square wave to an amplifier and find that the output waveform is not very square. The reason for this is that for a square wave to be perfectly reproduced, an amplifier would need to have a perfect frequency response. This, of course, is never actually the case.

Measuring distortion

The accurate measurement of distortion involves the use of some relatively sophisticated test equipment in the form of a spectrum analyser or wave analyser. When carrying out laboratory checks on prototypes we use a wave analyser as well as a distortion factor meter. The former instrument is capable of measuring the level of the individual harmonics present in an output signal, while the latter provides us with a figure for the circuit's THD performance in a particular bandwidth (we usually restrict our own audio measurements to an upper limit of 100kHz).

The distortion factor meter comprises a wide band (100kHz) voltmeter combined with a variable frequency notch filter that can be tuned so that it eliminates the fundamental frequency component present in the amplifier's output. In use, a reference level is set at 100 % with the filter switched out and then the level is measured again with the filter selected. In this condition, the signal measured is residual distortion present and its level (in relation to the fundamental) can be read from a panel meter in decibels (dB) or as a percentage.

It is important to be aware that, when using a distortion factor meter rather than a more complex wave analyser, the instrument will respond to all in-band signals, including noise, hum and other non-harmonically related components.

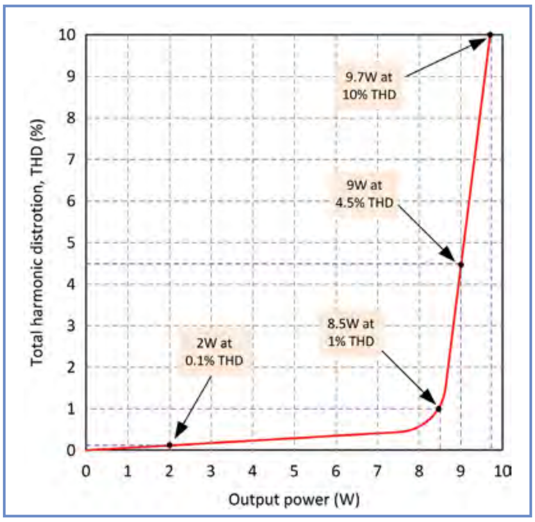


Fig.6.11. Overdriving a power amplifier results in a rapid rise in total harmonic distortion

Nevertheless, this type of instrument can be very effective when carrying out a quick assessment of the performance of an amplifier.

When working at very low levels of THD (typically less than 0.1%) the ultimate accuracy of the measurement becomes dependent on the quality of the input signal (which must be as near perfect as possible). We use a Radford low-distortion oscillator for our distortion measurements. This instrument is capable of producing a sinewave test signal with less than 0.003% and typically 0.001% THD. Note that popular low-cost function generators can often produce as much as 0.5% THD, and so this type of signal source is unsuitable for carrying out meaningful THD measurements.

How the THD is measured is important too. Measuring THD typically requires sinewaves, but measuring THD with a signal having a level just below that which would produce clipping can be instrumental in hiding other forms of distortion, notably cross-over distortion which disproportionately affects lower signal voltages. In most situations a THD measurement done at 10dB below rated output power (eg, at 1W RMS for an amplifier rated at 10W RMS) would be a much better indicator of sound quality.

Using PC-based software

An alternative to using stand-alone test equipment for audio measurements is using one or more virtual instruments. These use PC-based software and hardware and can be used for a variety of measurements including the analysis of distortion and noise. Solutions that use external hardware connected via a USB port are usually more accurate than internal sound cards to achieve the required analogue-to-digital conversion. Nevertheless, provided that your PC has a fast sound card with a low noise floor measurements can be made that are comparable with stand-alone test equipment (and often at much lower cost).

Fig.6.12 shows a virtual instrument display being used to check a sound card output and to ensure that the THD was sufficiently low to enable accurate low-level measurements to be carried out. As you can see, a great deal of information is available from the software. The virtual oscilloscope and signal analyser windows are simultaneously displayed. Notice that the frequency scale is logarithmic and that it extends from 20Hz to 20kHz, covering the full audio frequency range. Careful examination of the frequency spectrum shows the fundamental with an amplitude of 1V, together with odd-order harmonic components with descending amplitude. This suggests that the distortion arises from symmetrical rather than asymmetric non-linearity, as discussed earlier.

Various performance figures are reported in windows on the right. The first of these is the THD figure (0.0009%).

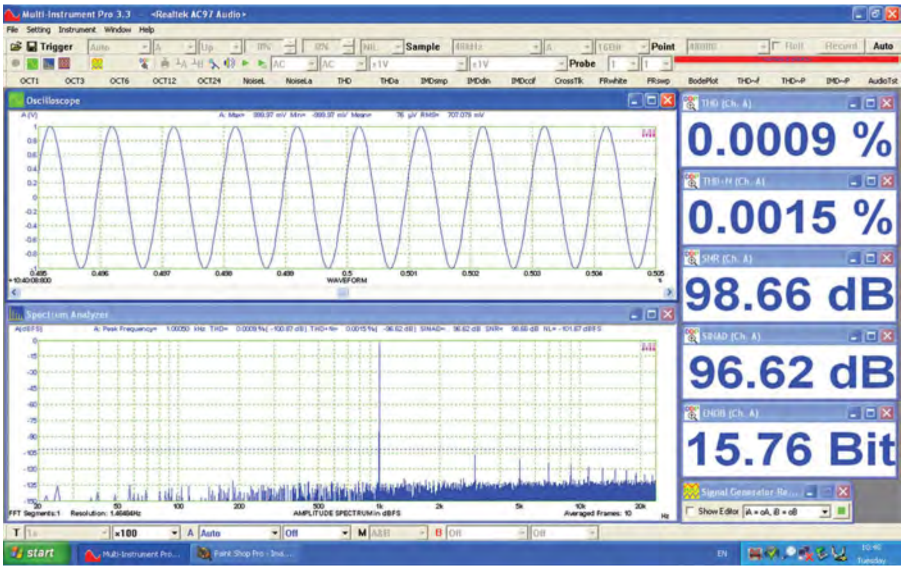


Fig.6.12. Virtual instrument display during tests on a sound card

This is extremely low. The next figure shows THD plus noise and hum (THD+N) and this would be the same figure that a distortion factor meter would report. The THD+N figure is, of course, greater than the THD figure but it is still very acceptable at a mere 0.0015%. The third window shows the signal-to-noise ratio (SNR) and the reported value is a very acceptable 98.66dB.

The next window shows SINAD or 'signal in noise advantage' SINAD is a measure of the signal quality that takes into account the presence of distortion. SINAD is calculated from the ratio of total output signal power (ie, signal power plus noise power plus the power arising from distortion components) to the noise-plus-distortion power. As might be expected, the SINAD performance is only slightly worse than the SNR performance, but the SINAD figure is usually a more reliable measure of performance, particularly when a significant amount of noise and distortion is present in a system.

The last window shows the effective number of bits (ENOB), which provides an indication of the dynamic performance of the system in relation to that of an analogue-to-digital converter (ADC). Since the number of bits used to represent an analogue quantity is specified by the resolution of an ADC, the number displayed indicates the resolution of an ideal ADC that would operate with the same resolution as the circuit under consideration. The figure quoted here is just less than 16-bits and is indicative of a high-quality audio system.

Signal-to-noise ratio

The signal-to-noise ratio in a system is normally expressed in decibels (dB) and is defined as:

$$S/N = 10\log_{10}\left(\frac{P_{signal}}{P_{noise}}\right)\text{dB}$$

In practice, it is difficult to separate the signal present in a system from the noise. If, for example, you measure the output power produced by an amplifier you will actually be measuring the signal power together with any noise that may be present. Hence, a more practical measure is the ratio of (signal-plus-noise)-to-noise. Furthermore, provided that the noise power is very much smaller than the signal power, there will not be very much difference between the signal-to-

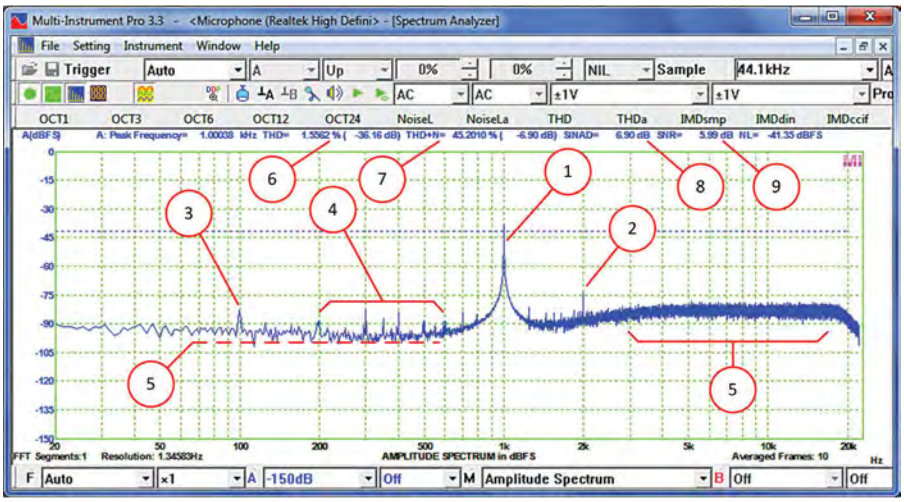


Fig.6.13. Spectral analysis of a signal with noise and distortion present

noise-ratio and the ratio of (signal-plus-noise)-to-noise; thus:

$$(S+N)/N = 10\log_{10}\left(\frac{P_{\text{signal+noise}}}{P_{\text{noise}}}\right) \text{ dB}$$

It might help to put this into context with some representative figures. Let's assume that, in the absence of a signal the noise power present at the output of a (rather noisy) amplifier is 100µW and when the signal is applied the output power increases to 400mW. The (signal-plus-noise)-to-noise ratio can be calculated from:

$$(S+N)/N = 10\log_{10}\left(\frac{400\text{mW}}{100\mu\text{W}}\right)$$

$$= 10\log_{10}(4000) = 10\times 3.6 = 36\text{dB}$$

The (signal-plus-noise)-to-noise ratio may also be determined from the voltages produced by an amplifier, in which case:

$$(S+N)/N = 20\log_{10}\left(\frac{V_{\text{signal+noise}}}{V_{\text{noise}}}\right) \text{ dB}$$

Spectral analysis – measuring signals in the presence of noise and distortion

Spectral analysis of a signal can be extremely useful in a wide range of practical situations and it can become invaluable when dealing with noise and distortion. As an example, the frequency spectrum of a 1kHz sinewave signal is shown in Fig.6.13 where there is appreciable levels of noise, hum and distortion present. The display was obtained using the Virtins Multi-Instrument Fast Fourier Transfer (FFT) software (see *Gearing up*). If you take a careful look at Fig.6.13 you should be able to recognise the following:

1. The fundamental of the wanted signal at 1kHz (with a level of about –40dB)
2. The second harmonic of the wanted signal at 2kHz with a level of about –75dB (35dB lower than the fundamental)
3. A component at 100Hz (twice the mains supply frequency) with an amplitude of about –82dB. This was caused by a small amount of ripple present on the amplifier's DC supply
4. Harmonics of the supply ripple at 200Hz, 300Hz, 400Hz...
5. A noise floor of about –97dB with a slight increase in noise between about 2kHz and 20kHz
6. A reported THD of 1.5562% (–36.16dB relative to the 1kHz fundamental)
7. A reported THD plus noise (THD+N) of approximately 45% (–6.9dB relative to the 1kHz fundamental) – contrast this with the THD figure without noise!
8. A reported SINAD figure of 6.9dB
9. A reported signal-to-noise ratio of 5.99dB (unacceptably low for most applications).

This example gives you an appreciation of just how useful spectral analysis can be when signals are contaminated with both noise and distortion.

Gearing up: wideband AC range extender

For general audio measurements you will need, as a minimum, a good quality sinewave signal source together with an oscilloscope and an AC millivoltmeter (both described earlier *Teach-In 2018*). In addition, a dedicated AF power meter and a THD analyser would also be useful. With the exception of a power meter (which will normally incorporate a suitably rated load) these instruments can also be realised using computer-based software and hardware.

Audio frequency signal generators

An audio frequency (AF) signal generator, 'audio oscillator', or 'RC oscillator', can be a very useful investment if you are planning to carry out measurements on a regular basis. A typical AF signal generator will be capable of providing a good quality sinewave output over a frequency range of at least 10Hz to 20kHz, and ideally higher. The equipment should be fitted with a calibrated adjustable attenuator so that output levels ranging from at least 1mV to 1V can be applied to the equipment on test.

For a new instrument you can expect to pay around £100 but second-hand AF signal generators are frequently available at bargain prices from as little as £20 to £50. Instruments by Farnell, Levell, Gould, Advance and Marconi are regularly available. You might be tempted to invest in a modern direct digital synthesis (DDS) waveform generator. Such instruments sell new for around £80 but, although they might appear to have an excellent specification in terms of frequency range and waveform capability, they invariably produce an output signal that just isn't good enough for THD measurement. A purpose-designed AF signal generator should be capable of producing an output with less than 0.01% THD. This is vastly better than the '≤0.8%' offered by low-cost entry-level DDS instruments. This same reservation applies to cheaper 'function generators' based on waveform

Fig.6.14. An audio signal generator (left) and millivoltmeter (right)

Fig.6.15. An RC oscillator with sine and square wave outputs in the range 1Hz to 3MHz, and a THD of less than 0.05% at 1kHz. Similar instruments are regularly available on-line

generator chips where the THD can be greater than 1%. By comparison, the author's Radford LDO3 low-distortion oscillator achieves a typical THD of 1,000-times better at a mere 0.001%. However, if you are not concerned with measuring distortion a general-purpose signal or waveform generator will be adequate for you needs.

It's perhaps also worth mentioning that a simple AF signal generator can make an excellent project for home construction and the complete circuit of a simple Wien bridge audio oscillator is shown in Fig.6.16. This particular design uses two low-cost operational amplifiers and produces variable outputs adjustable from less than 1mV to 1V RMS with four switched decade frequency ranges extending from 2Hz to 20kHz.

AF power meters

As mentioned earlier, a simple method for measuring audio power can be based

Get it right when using an AC voltmeter

- Always ensure that test signals are free from hum, noise and distortion. This is particularly important when carrying out THD measurements.
- Avoid over-driving amplifiers and filters and always keep input signals within the normal range for the equipment on-test
- When carrying out input/output impedance and gain measurements always ensure that the test frequency is set to the centre of the mid-band of an amplifier (or to the middle of the pass-band for a filter)
- Always check that the correct load impedance is used when carrying out audio power measurements and that the load is rated for continuous operation (if this isn't the case you may need to de-rate the load or conduct tests for a short time only)
- If you are using a virtual instrument based on a PC-based sound card it is well worth checking that you are using the full capability of your sound card (software settings often default to lower-than-optimum bit rates)
- Don't rely on measurements where THD and SINAD indications may be towards the end of the instrument's measuring range (accuracy will invariably be impaired as an instrument's limits are approached).

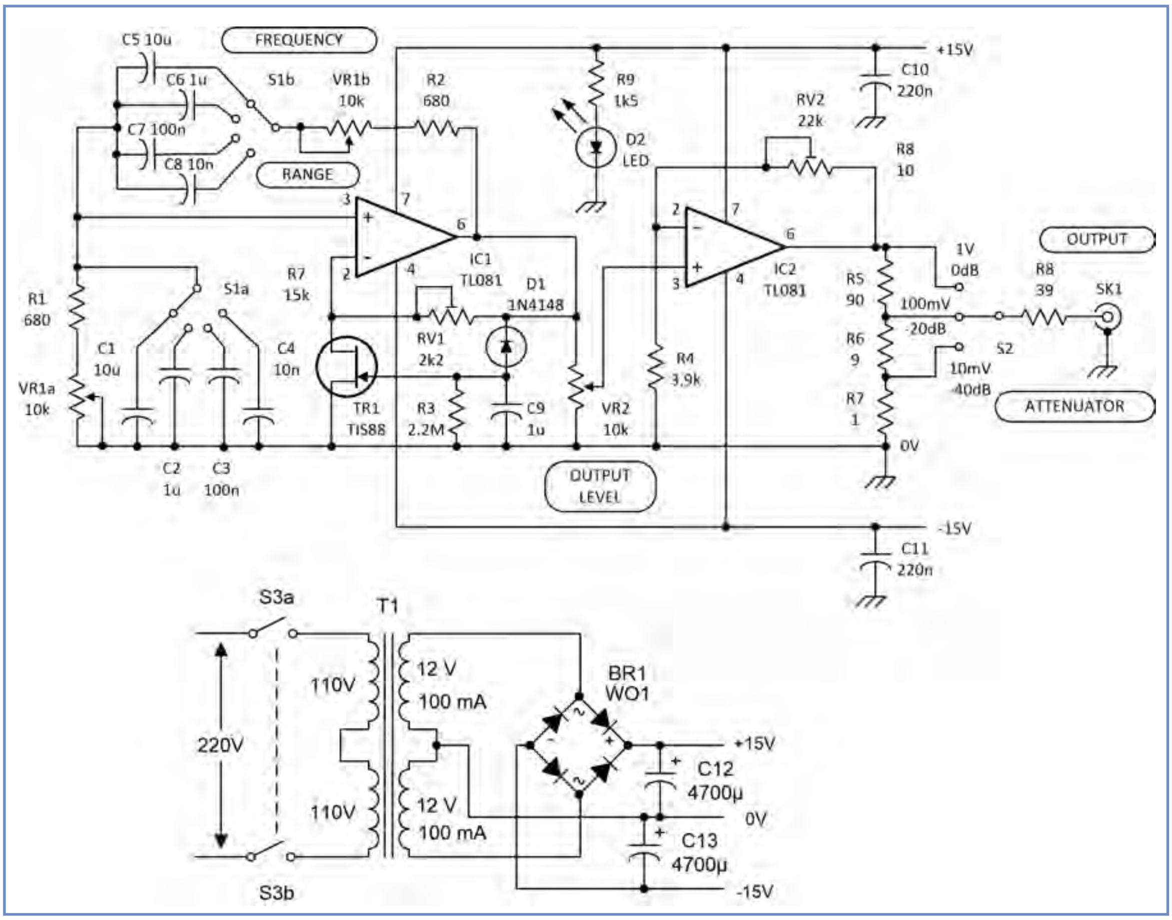


Fig.6.16. *Complete circuit of a simple AF signal generator suitable for home construction*

on nothing more than a suitably rated load resistor and an accurate AC voltmeter. However, this method requires calculation and can be a little tedious when a large number of measurements are required. Instead, a dedicated audio wattmeter can provide a much more convenient solution. Unfortunately, such instruments seem to be in limited supply, but the vintage Heathkit AW-1U is occasionally available from on-line auction sites and can often be obtained for around £20 to £50. This instrument (see Fig.6.17) can read power to ±1dB over the frequency range 10Hz to 250kHz and power levels of up to 50W (25W continuous) with internal loads of 3Ω, 8Ω, 15Ω and 600Ω. Marconi TF893 and TF2500 audio output meters are also available from time to time and they can be an excellent investment.

Fig.6.17. *A Heathkit vintage AW-1U audio output power meter*

Distortion analysers

Distortion analysers can be expensive and, as mentioned earlier, have been replaced by powerful PC-based FFT software and fast ADC hardware. However, instruments from HP/Agilent, Leader, Marconi and Keithley do become available from time to time at prices ranging from around £50 to over £500. The author's own Marconi TF2331 distortion analyser is shown in Fig.6.18. This instrument functions as both an analyser and wideband AC voltmeter and it can measure THD down to 0.01%. The TF2331's range switch and meter calibration is shown in Fig.6.19.

Phase meters

Phase angle can be measured with a limited accuracy using a dual-channel oscilloscope, but for accuracy and convenience a dedicated phase meter is invariably a better choice. It will provide a means of measuring the phase angle

Fig.6.19. *Range switch and meter calibration of the TF2331 distortion analyser (Fig.6.18)*

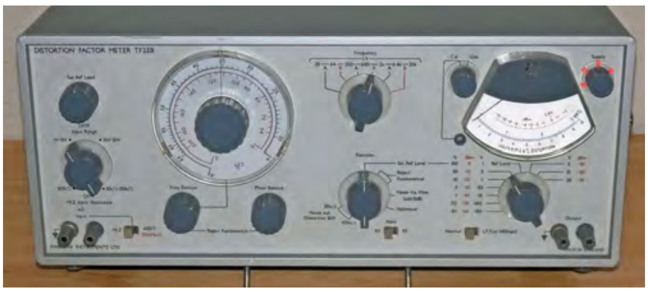


Fig.6.18. *The Marconi TF2331 distortion analyser*

Fig.6.20. *Phase meter with analogue display. The scale is calibrated from 0° to 180°, and the sign of the phase relationship (either leading or lagging) is indicated by a panel LED*

between two waveforms (eg, the input and output of an amplifier). It is usually displayed on a scale (either analogue or digital) with a range extending from 0° to ±180° or 0° to 360°. Note that conventional phase meters can often be erratic when small levels of noise and distortion are present, and the result is often incorrect and unstable indications. Modern phase meters overcome this by using Discrete Fourier Analysis (DFT) to reject any noise and distortion without the need for tracking filters.

Computer-based virtual instruments

In *Part 2*, we introduced software and hardware packages designed primarily for use as virtual oscilloscopes. The software supplied with these instruments usually has FFT capability and so it can also be a valuable tool for distortion analysis. So, if you have a reasonably fast PC with a good quality sound card you will be able to use these for a variety of audio measurements, not just observing waveforms. If you need a wider range of features and greater accuracy then it is worth considering PC instruments from Pico Technology, Virtins, Hantek and many others.

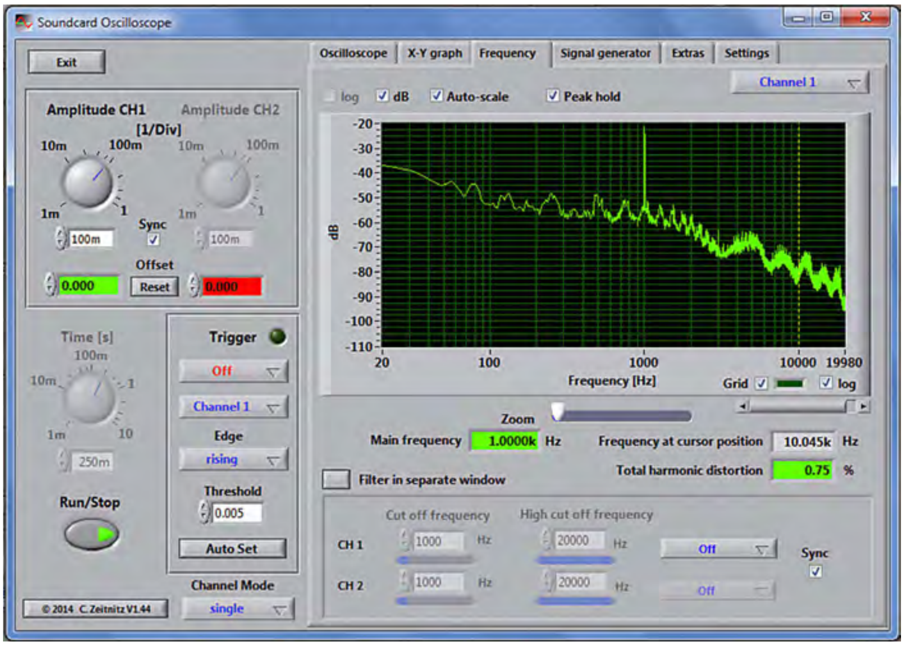


Fig.6.21. A typical frequency domain display produced by Christian Zeitnitz's software. it works with an internal PC sound card and it also provides you with a handy audio signal generator

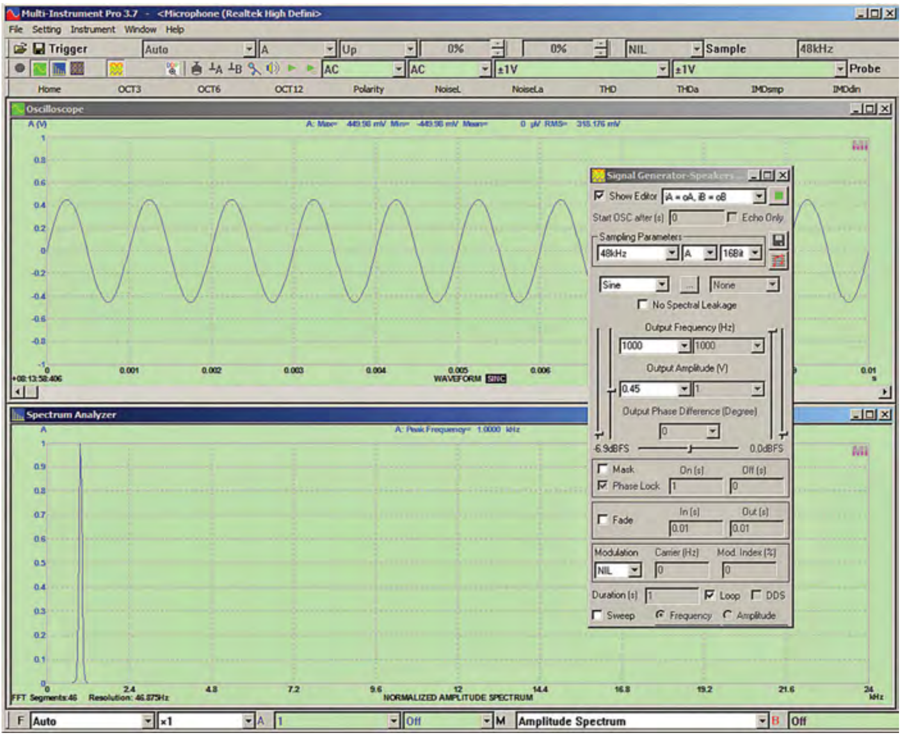


Fig.6.22. The Virtins Multi-Instrument signal generator producing a 1kHz sinewave with an amplitude of 450mV. Digital loopback has been enabled so that the waveform and frequency spectrum can be concurrently displayed in the oscilloscope and spectrum analyser windows

Test Gear Project: A handy test signal source

Our handy test signal source will provide you with a low-distortion 1kHz sinewave signal that will allow you to test a wide variety of audio circuits. It can be used for sensitivity, voltage gain, and input/output impedance measurements, as well as carrying out quick waveform checks for distortion.

The complete circuit of our *Test gear project* is shown in Fig.6.23. The circuit comprises an oscillator based on a twin-T phase-shift network. The frequency of oscillation is determined by the component values used in the feedback network. R2, R3 and C4 form one branch; and C2, C3 and R4 (in series with the fine frequency adjustment, RV1) form the other branch of the twin-T network. The gain of the amplifier stage (TR1) is made adjustable by means of RV2. The output amplitude is made adjustable by means of VR1 and the output signal (approximately 150mV RMS) is made available at the two front-panel-mounted 2mm sockets, SK1 and SK2. With careful adjustment the output signal THD can be as little as 1%.

You will need

Perforated copper stripboard (9 strips, each with 25 holes)
ABS case with integral battery compartment
9V PP3 battery clip
9V PP3 battery
Miniature DPDT toggle switch (S1)
2 2-way miniature terminal blocks (ST1 and ST2)
1 red 2mm panel-mounting socket (SK1)

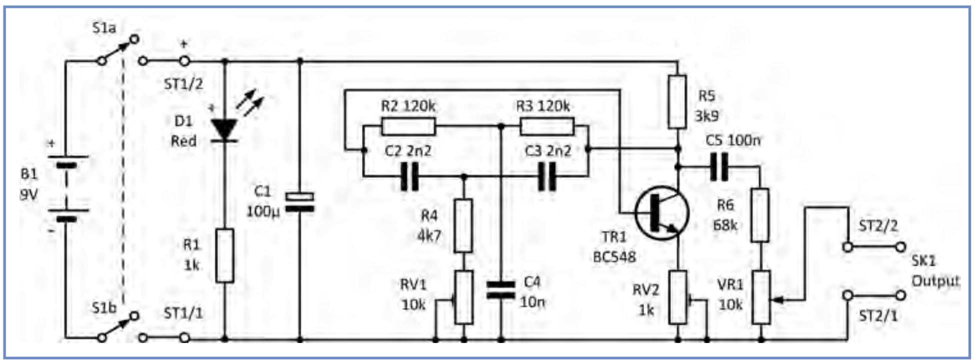


Fig.6.23. Complete circuit of the handy test signal source

1 black 2mm panel-mounting socket (SK2)
1 BC548 transistor (TR1)
1 5mm red LED (D1)
1 1kΩ resistor (R1)
2 120kΩ resistors (R2 and R3)
1 4.7kΩ resistor (R4)
1 3.9kΩ resistor (R5)
1 68kΩ resistor (R6)
1 10kΩ miniature multi-turn pre-set resistor (RV1)
1 1kΩ miniature multi-turn pre-set resistor (RV2)
1 10kΩ linear variable potentiometer (VR1)
1 100µ 16V radial electrolytic (C1)
2 2.2nF disc ceramic capacitors (C2 and C3)
1 10nF disc ceramic capacitors (C4)
1 100nF disc ceramic capacitors (C5)

Assembly is straightforward and should follow the component layout shown in Fig.6.24. Note that the '+' symbol shown on D1 indicates the more-positive (anode) terminal of the LED. The pin connections for the LED and transistor are shown in Fig.6.25. The reverse side of the board (*not* an X-ray view) is also shown in Fig.6.24. Note that there's a total of 23 track breaks to be made. These can be made either with a purpose-designed spot-face cutter or using a small drill bit of appropriate size. There are also eight links that can be made with tinned copper wire of a suitable diameter or gauge (eg, 0.6mm/24SWG). When soldering has been completed it is very important to carry out a careful visual check of the board, as well as an examination of the track side of the board, looking for solder splashes and unwanted links between tracks. The internal and rear panel wiring of the test signal source is shown in Fig.6.26.

Setting up

Setting up is reasonably straightforward, but ideally you will need an oscilloscope and a digital frequency meter to calibrate the source and ensure that the output signal is undistorted. It is still possible to set the circuit up without these test instruments. Connect the oscilloscope to the output (if you don't have an oscilloscope you can just connect the output to an amplifier and use a speaker to monitor the signal). Next, set RV2 to minimum (zero resistance between TR1 emitter and 0V) and switch on. The output waveform (along with some noticeable distortion) should then be displayed on the 'scope (or heard from the loudspeaker).

Slowly increase RV2 until the oscillation stops, then back off the adjustment until it just starts again. At this point the output waveform will be a reasonably pure sinewave with an amplitude of about 150mV. Next, switch off momentarily and then switch on again. Check that oscillation restarts. If not, repeat the process but back off the setting of RV2 a little further until oscillation starts reliably. If you have a digital frequency meter available this can be connected to the output and RV1 can be adjusted for an output of exactly 1kHz.

If you only have an oscilloscope available you can set RV1 to produce an output waveform with a period of exactly 1ms. This will be less accurate than using a digital frequency meter, but you should still be able to produce an output within about 50Hz of the nominal 1kHz. If you don't have either an oscilloscope or a digital frequency meter it is possible to use a keyboard musical instrument to set the output frequency since the B5 key (two octaves above middle-C) should produce sound at approximately 988Hz. By comparing the sound from a loudspeaker driven by the test signal and an amplifier from the keyboard instrument you should be able to produce a signal that is very close to 1kHz.

Next month

In next month's *Teach-In 2018* we will be looking at radio frequency (RF) tests and measurements. We will be introducing a selection of RF test instruments and measurement techniques and our practical project will feature a sensitive RF 'sniffer' that can be used to check for radiated signals over a very wide frequency range.

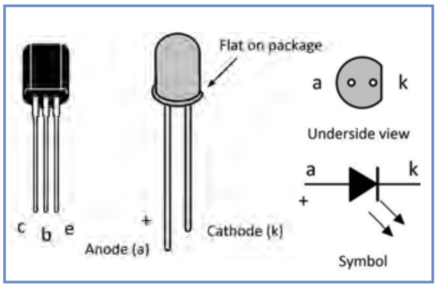


Fig.6.25. (left) Transistor TR1 (BC548) and (middle/right) diode D1 pin connections

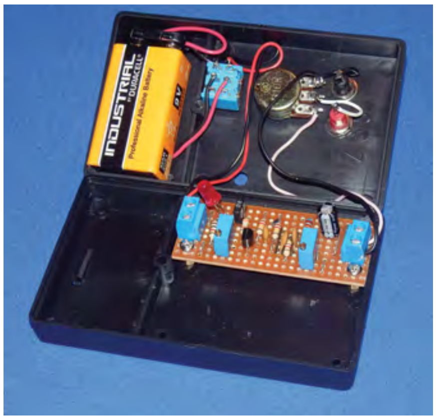


Fig.6.26. Internal wiring of the handy test signal source

Fig.6.27. External appearance of the test signal source

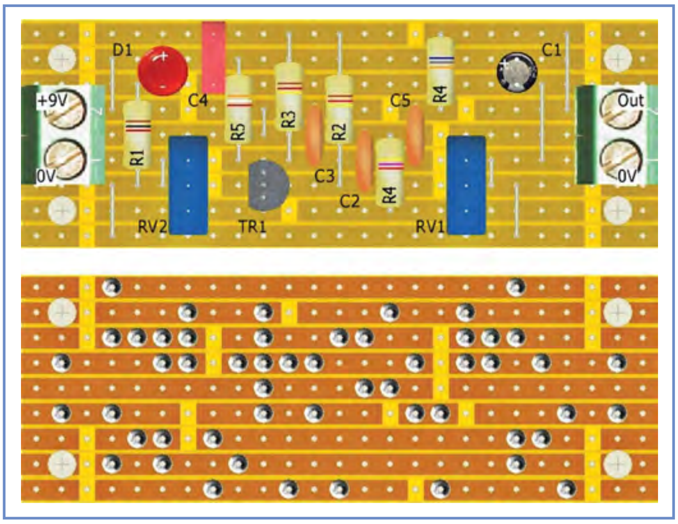


Fig.6.24. (top) Stripboard layout of the handy test signal source and (bottom) underside of the stripboard showing track breaks

Fig.6.28. Using the signal source to test a high-quality audio amplifier at its full rated output power