Trial by three tones

Testing audio amplifiers by the triple-tone method is simple, uses a similar signal to that encountered when playing music and produces informative data. Ivor Brown describes the method and presents results

In the field of audio systems, and particularly with amplifiers, there is a long-standing problem of what measurements should be made to make a quantitative assessment that is in agreement with subjective listening tests. It is well known that amplifiers found to be very linear in the laboratory do not always perform well in listening tests.

Most tests use repetitive waveforms that have a defined spectrum; the relative amplitudes of those frequency components in the output waveform that are not present in the input signal can then be used to assess the performance of the amplifier. The simplest test is to apply a single sinusoid to the amplifier and to find the harmonics in its output. Taking the square root of the sum of the squares of the amplitudes of the harmonics and relating the result to the fundamental leads to the total harmonic distortion (THD) figure. It is now accepted that THD has little relevance to audio performance. However, a poor THD does suggest that all is not well, while a low figure is encouraging, in that the amplifier may be capable of good subjective results.

A slightly more complex approach is to use two sinusoids and to find their intermodulation products, which are generally larger and more numerous than the harmonics. The same distortion mechanisms are involved, so little more information is gained about the amplifier.

Some have employed pulse waveforms. Hirata[1] proposes an input composed of asymmetrical positive and negative pulses, arranged so there is no DC component in the signal. Any DC present in the output waveform is due to non-linearities in the amplifier and forms the basis for the assessment. The 'no–DC' requirement for the input waveform is critical, and not easy to satisfy if low-distortion amplifiers are being tested; Belcher[2] uses computer-generated pseudo-random binary waveforms. The process involves digital frequency shifting and comb filters to isolate the distortion components generated within the amplifier. It is a complex system, but good agreement with listening tests is claimed.

Amplifiers in the 1990s

What are the critical areas of amplifier performance that should be investigated? The days of worrying about reproducing the full audio frequency range are past, as is the concern over obvious distortions present in the output signal; yet most systems cannot create an effective illusion of a live performance, which surely should be a principal aim. Two important qualities necessary for the illusion are clarity and definition. www.keith-snook.info

Until the arrival of the CD, all sources of reproduced music contained appreciable noise, which served to mask low-level distortions produced by amplifiers. Low levels of noise, being steady and not related to the music, are easily tolerated by most listeners. Without noise, such distortions can be detected, not as definite sounds, but as a lack of clarity. In stereo and other

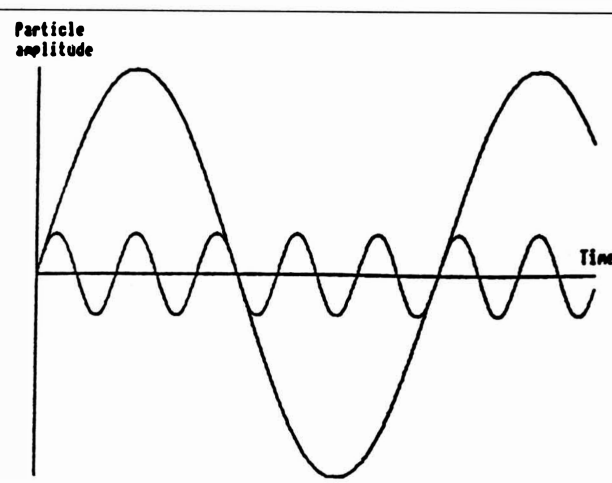

Fig. 1. For two sinusoidal waves with different frequencies to have the same peak particle velocity, relative amplitudes must be inversely proportional to frequencies.

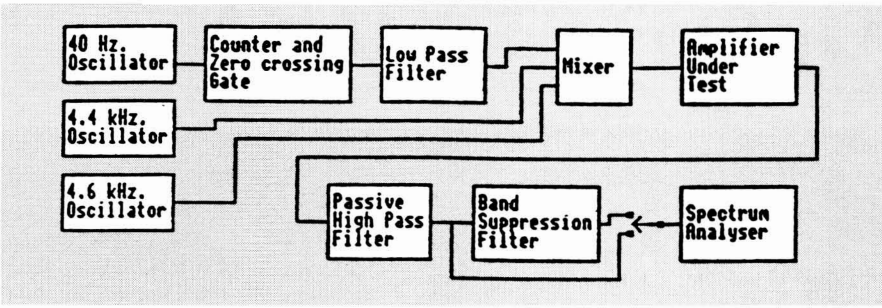

Fig. 2. Triple-tone test system.

multi-channel systems, this will be experienced as a loss of definition, with the spaces between the performers filled with a variable background derived from, but not simply related to, the programme material. It follows, therefore, that we should measure distortion components down to lower levels than has been the practice. (The fact that audio masking occurs is evidenced by the literature accompanying some CD re-issues, apologising for distortion and/or spurious studio noise that was only found when the master tapes were digitised.)

Some basic acoustics

The power in a plane sound wave is given by the product of the square of the RMS particle velocity and the specific acoustic impedance of the medium, air in our case. **Figure 1** shows the particle amplitudes of two sinusoidal sound waves with different frequencies. They have the same gradient at the zero crossings, and hence the same peak particle velocity. To achieve this, their relative amplitudes have to be inversely proportional to their frequencies. The specific acoustic impedance is resistive and independent of frequency for a plane wave, so the power in the waves is the same. The amplitude of a 100Hz sound will be 100 times, or 40dB, greater than a 10kHz one of the same power.

This is a simplified situation; most sound waves met in audio reproduction are not plane and the ear does not have a flat frequency response, but these considerations do not alter the basics. In fact, the situation is worse when non-plane waves are considered. If a 100Hz sinusoid produces high-order harmonics 70dB below the fundamental, they are only about 30dB below a 10kHz signal of equal energy. High-frequency sounds are not effectively masked by large-amplitude, low-frequency ones; consider how the piccolo or triangle can be heard through a loud orchestral climax.

This was discussed by Wigan[3] in 1961. He applied a weighting factor to each harmonic up to the tenth, so increasing the significance of high-order ones, but that number of harmonics is surely too low for current amplifiers. This is a lengthy process that requires the measurement of very low level harmonics.

Triple-tone method

Figure 2 is a block diagram of the test system. The first signal is a low-frequency (40Hz), large-amplitude sinusoid, gated at its zero crossings to form tone bursts. The other two signals are equal-amplitude continuous sinusoids and come from low-distortion oscillators. They are smaller than the low-frequency signal and their frequencies are close together (nominally 4.4 and 4.6kHz). All three are added to make the input signal to the amplifier under test. Using two high-frequency signals enables harmonic and intermodulation products to be investigated.

A low frequency is chosen for the first signal so that the variable current taken from the amplifier's power supply during each cycle is likely to cause the rail voltages to alter, the effect being enhanced as the signal is in bursts. After gating, it is passed through a low-pass filter to reduce the high-order harmonics produced by the switching process. Another advantage of gating is that it lowers the dissipation in the amplifier for a given peak output. Most amplifiers are not intended to operate continuously at high power levels, music signals having a high peak-to-mean power ratio.

The low-frequency signal sweeps the two small signals over most of the amplifier's working range, changes in their amplitude and waveform being observed as harmonic and intermodulation products. The cross-over region is critical for class-AB amplifiers, which are by far the most common type; the gated 40Hz allows the small signals to remain in this region during its off periods, hence giving it extra weighting.

A passive, high-pass RC filter effectively removes the 40Hz signal and its low-order harmonics, leaving the two high-frequency signals and distortion products to be displayed on a spectrum analyser, with each fundamental at a level of approximately −6dB. If the fundamentals do not appear in a given spectrum they can be suppressed and the analyser sensitivity increased. To show the fine detail of the spectra only a narrow range of frequencies can be displayed, using a slow sweep rate and narrow bandwidth.

In this instance, the 40Hz bursts have equal on and off periods of eight cycles, with a peak level at the output of 15V. The corresponding steady-state sinewave power is 14W in 8Ω, so the output voltage sweeps about two thirds of a 30W amplifier's dynamic range. Each high-frequency signal is about 30dB below the peak output level; RMS voltages of the individual signals at the output are 9.94V at 40Hz and 0.34V at 4.4 and 4.6kHz.

Spectra are centred on multiples of 4.5kHz, with an analyser bandwidth of 3Hz. When the 4.4 and 4.6kHz compo-

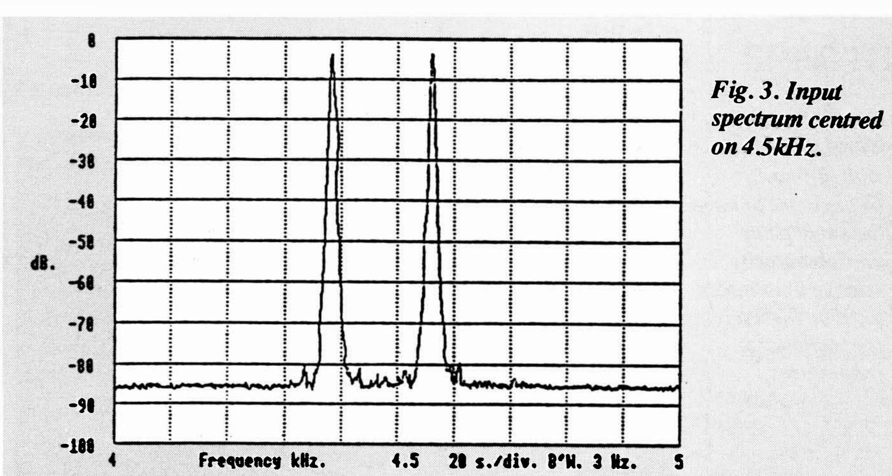

Fig. 3. Input spectrum centred on 4.5kHz.

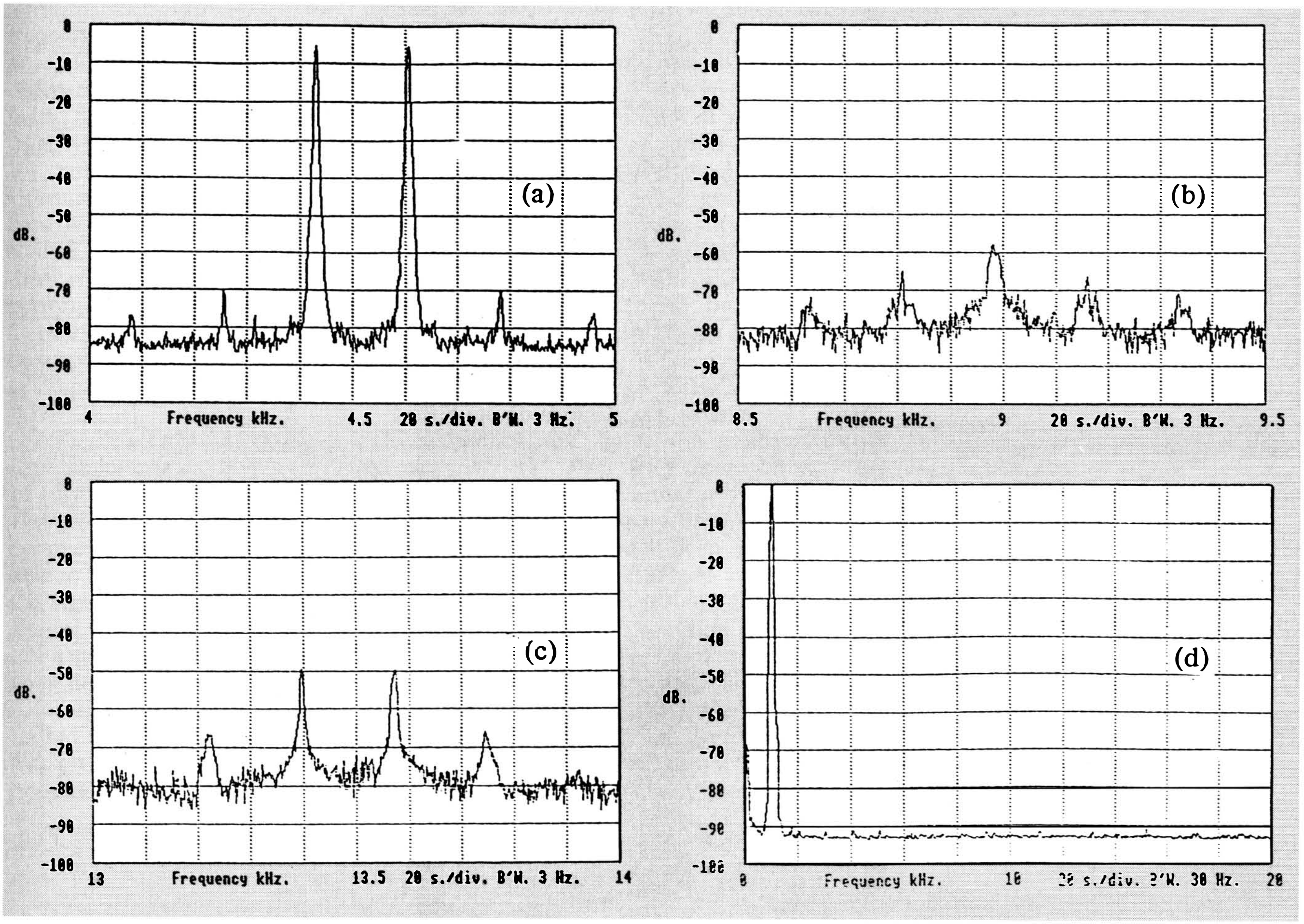

Fig. 4. Triple-tone testing of amplifier 1 centred on (a) 4.5kHz; (b) 9.0kHz; (c) and 13.5kHz; and a check spectrum for 1kHz 15V peak output (d).
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nents are displayed on the analyser at −6dB, its noise level of around −85dB is some −109dB below the peak output signal, enabling components of 0.0004% of the peak to be revealed. For higher-frequency spectra with the suppression filter in circuit, this figure is reduced by 10dB.

Input waveform

For all distortion measurements it is important to check the spectrum of the input signal and to be certain that no significant distortion is being introduced by the equipment. **Figure 3** shows the input spectrum centred on 4.5kHz with small side components 50Hz away from the main signals. The higher frequency spectra have no components above the noise level and are not shown.

Results

Spectra labelled (a) (b) and (c) are from triple-tone tests and are centred on 4.5 9.0, and 13.5kHz respectively, with the analyser sensitivity increased by 10dB for spectra (b) and (c). The frequencies of the two small signals (f_1 and f_2). and the frequency scaling on the spectra depend on manual setting up, so will not be highly accurate. In this context, this is of no importance.

Amplifier 1. Many component distributors offer an encapsulated module that will provide an output of 30W into 8Ω at 1kHz, with less than 0.02% distortion. Considering these figures alone, it seems suitable as a good-quality domestic power amplifier. **Figure 4** shows the results of testing one of these modules.

(a) Intermodulation components at ($2f_1$-f_2); ($3f_1$-$2f_2$); etc. There are no specific peaks to relate to the 40Hz signal, but there is a rise in the noise level near the peaks, indicating many small intermodulation products.

(b) Second harmonics at $2f_1$ and $2f_2$ are present, with the intermodulation sum component at (f_1+f_2) and others at ($3f_1$-f_2) etc. Again, the noise level rises near the peaks.

(c) Third harmonics are present with peaks at ($2f_1$+f_2) and ($2f_2$+f_1) between them. These intermodulation products are only some 54dB (0.2%) below the signals they are derived from.

All these components come from only three input signals. How many more will occur with a complex music waveform? It is worth noting that when the 40Hz signal is made continuous, most of the intermodulation components disappear. The weakness of the design clearly lies in the cross-over region.

(d) As a check, the spectrum for a 1kHz 15Vpeak output is also included. There are no apparent problems, at this level the specification was handsomely satisfied, with all distortion components at −90dB or better.

Amplifier 2. Figure 5 shows the results from a well considered amplifier designed some twenty years ago. It has a single supply rail and feeds the load via a large output capacitor. Both channels were driven.

(a) Significant components are present at multiples of 40Hz away from the fundamentals. There is no evidence of intermodulation products from the two high-frequency signals.

(b) Shows second harmonics and the

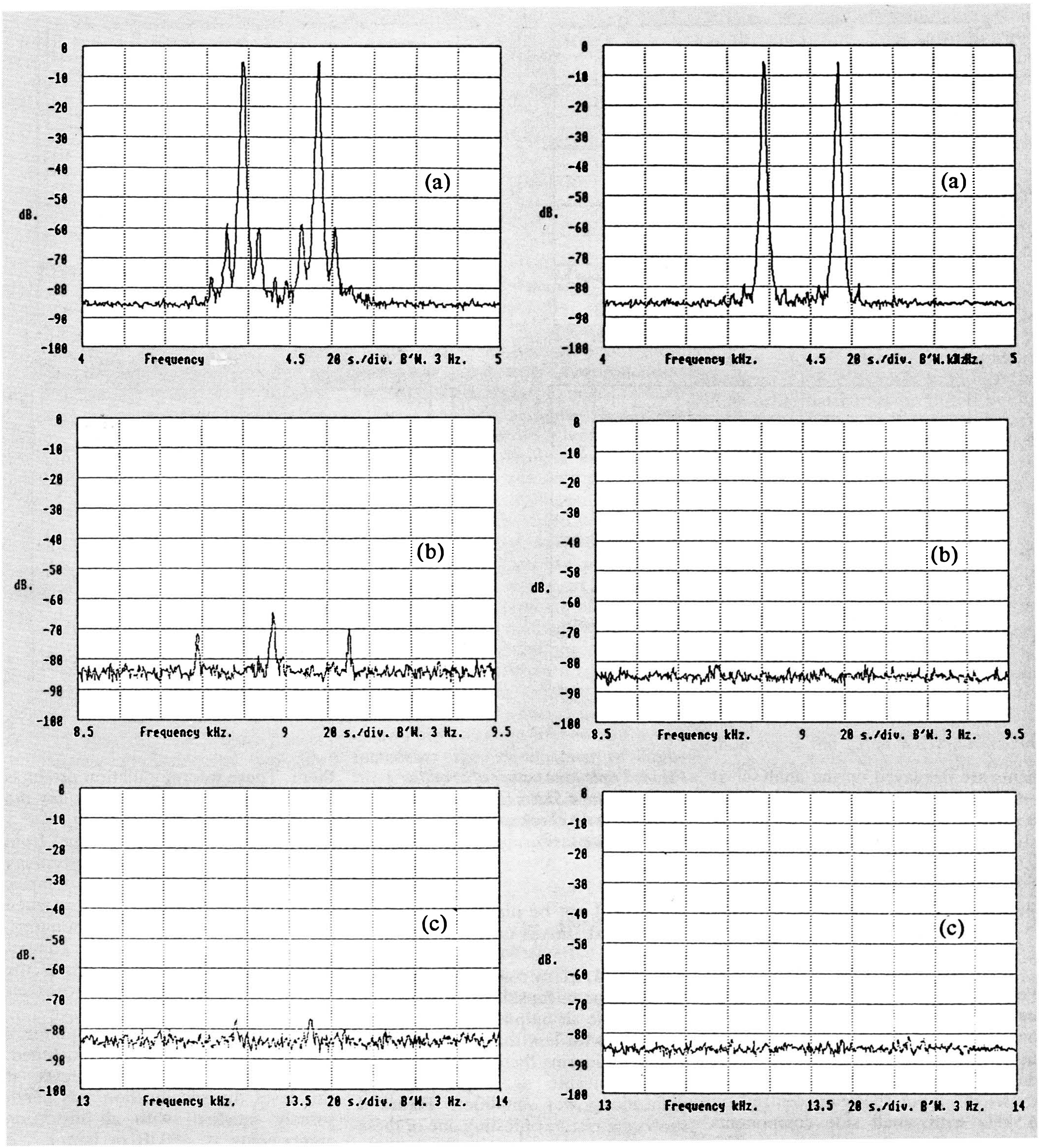

Fig. 5. Triple-tone testing of amplifier 2 centred on (a) 4.5kHz; (b) 9.0kHz; (c) and 13.5kHz.

Fig. 6. Triple-tone testing of amplifier 3 centred on (a) 4.5kHz; (b) 9.0kHz; (c) and 13.5kHz.

sum component.

(c) Only very small peaks at $(2f_1+f_2)$ and (f_1+2f_2) are present.

The cross-over region is better managed than in the first amplifier, but variation of the supply rail voltage with signal does cause problems.

A "classic" valve design and a class-A transistor design, both of about the same vintage as amplifier 2, were also investigated, their spectra showing similar 40Hz-related features. Is it a coincidence that all three have a single unbalanced power supply?

Amplifier 3. This is my experimental mosfet design, previously described in this publication[4,5]. Spectra in **Fig. 6** are for one channel of a stereo pair, with both channels driven and fed from a common, unstabilised power supply. The reservoir capacitors were 2200μF; an appreciably lower value than would normally be used.

Apart from an increase in low-frequency noise, shown by the rougher base line to the spectra, they are little different from those for the input waveform. The noise can be lowered somewhat with larger reservoir capacitors.

Other amplifiers. A number of amplifiers belonging to Brunel University students have also been tested; most of them commercial products. They outperformed amplifiers 1 and 2, but all had defined distortion components in their spectra. None were quite so clean as amplifier 3.

Conclusions

This triple-tone method is a by-product of my work on mosfet amplifiers. Equipment that has to be made is not complex and the rest is readily available in audio laboratories. The composite signal "exercises" the amplifier in a similar manner to music signals, and has a simple, well defined spectrum. It does not directly investigate transient distortion mechanisms, but these should not be a problem with careful design. (Antoniazzi and others[6] describe a method for determining whether transient effects are a problem.)

Frequencies and relative amplitudes of the signals were chosen because they fitted in reasonably well with acoustic theory, suited the equipment available and gave informative results. It may be that more meaningful results would be obtained if these were changed. Some amplifiers tested produced worse-looking spectra with different amplitudes and frequencies, but a standard condition is necessary for comparisons to be valid.

As a university lecturer, my main concern is electronic circuits, so I have neither the time, nor resources to perform laboratory and subjective listening tests on either a wide range of amplifiers, or costly ones. My mosfet amplifier has now been heard by a considerable number of people, who confirm that it does have the qualities of clarity and definition. From the number of significant distortion components in the spectra, their frequencies and amplitudes, it may be possible to derive a figure of merit for amplifiers that agrees with subjective assessments. I must leave that to other interested parties.

Triple-tone testing, as a development tool in the laboratory, has already proved very useful, and should be more so if the sensitivity of the system can be increased further. ■

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