Letters to the Editor

THE LANGUAGE OF HI-FI

Your balanced and sensible leader in the August issue came as balm to my inflamed spleen after also reading in one of your considerably less distinguished contemporaries that a highly respected preamplifier "sounded boring" and "made the music sound as if played by amateurs". Surely the nadir of lunacy in the use of subjective language! One gets the impression that these terminological outrages are being perpetrated on gullible readers by a new breed of journalistic *wunderkind*, who would probably be hard pressed to define a decibel. The reasons for this development are beyond me — probably it is either an effort to conceal technical incompetence or because it makes saleable copy; or a mixture of both.

Of course, I am not against the use of subjective language. What I am against is the increasing tendency to use language of imprecise meaning. To misquote Gertrude Stein "a volt is a volt is a volt" and I hope no one is going to question that or challenge that a volt measured in hi-fi equipment is any different from any other. But when someone says vis-à-vis the performance that the "information retrieval efficiency was low" (yes, really — I didn't make it up) then like the late and quite unlamented Hermann Goering, I reach for my axe. If I as an experienced professional engineer cannot understand it, then heaven help the poor layman.

We commentators in engineering journalism have a heavy responsibility and should never resort to language that is capable of alternative interpretation or is open to doubt: and if there is a slight doubt, then it should be clearly defined or explained. At the risk of being accused of pedantry, I will go further and say that every observed phenomenon in reproduced sound is measurable and may be expressed in quantitative terms. Some subtle effects perhaps may be harder to measure than others; but I am with Galileo and Lord Kelvin. Inventing new words is not the way out.

May I finish with another observation, and a warning against another tendency not confined to the popular hi-fi press? This is the lack of a sense of proportion and a failure to appreciate the realities of the technical side of audio. I have just been reading with interest an article in a well-known technical publication. The writer discusses with great insight, the technical desiderata for a pickup input stage; then spoils it all by proudly declaiming in the final paragraphs that the improvements result in a reduction of the t.h.d. to 0.0004%. Marvellous. Then if someone is able to make a gramophone record and cartridge capable of the same order of inherent *Dt* we might just be able to notice the difference.

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AURAL SENSITIVITY TO PHASE

I fear that Mr Moir (Letters, July 1977 issue) has misunderstood the point which I was trying to make in my letter on the audibility of polarity reversals (Letters, May 1977). Far from the distortion of one stage in the amplifier chain being cancelled by a complementary distortion in a subsequent stage, as suggested by Mr Moir as an explanation for the effects I discussed, I was at pains in my letter to make clear that this was *not the case*. All subsequent stages in the chain, including the transducer, were shown not to be responsible for the effect in question. (In the case of the loudspeaker, this was done by listening from both front and back of the dipolar electrostatic panels, thus introducing a polarity reversal in the acoustic waveform, which was found to reverse the effect.) The change in quality of the signal was due entirely to its *own* asymmetry, not to subsequent distortion. This confirms the earlier work cited in my letter.

An even more vivid demonstration of this effect can be obtained by linearly combining two sinusoidal oscillator signals, one a "fundamental" frequency of around 400Hz and the other an adjustable-level "second harmonic" of around 800Hz. If the second harmonic is allowed to drift *slowly* in phase relative to the fundamental a very pronounced cyclic change in the sound quality of the signal will be heard, and it is instructive to listen to it while observing the asymmetric waveform on an oscilloscope. No such effect appears to occur if the 800Hz signal is shifted to the third harmonic, i.e. 1200Hz; the waveform is now always symmetric with respect to polarity reversals. With a fourth harmonic, however, the effect is again subtly audible if the level is suitably chosen.

Towards the end of his letter, Mr Moir in fact seems to support my argument, by agreeing that *on good signals* a polarity reversal *is* indeed subtly audible. This strikes me as being an important conclusion! Even more than just standardizing the absolute polarity of the whole audio chain, as I suggested, it would seem that the non-linear-phase errors inherent in the use of pressure and/or velocity microphones in recordings, which are reproduced indiscriminately via either pressure or velocity transducers, also requires serious investigation.

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Mr Driscoll, responding in the July issue to my letter of last February, asserts of himself "My grasp of basic principles is not so uncertain that I could believe Coleman's claim that "tone bursts which differ in the framing of phase" (I wrote "OR 'phase' ") of the sine wave with respect to the burst envelope have spectra of different shapes." My claim can easily be checked, and is correct. Where does that leave his "grasp of basic principles"?

If the members of a regular sequence of tone bursts are well separated, so that they are heard as separate bursts, it is enough to calculate the Fourier transform or spectrum of any one of them. If a particular burst consists of the sinusoid $\sin(2\pi f_0 t+\epsilon)$ gated on for $2n$ periods centred about the time $t=0$ then its transform is $K\sqrt{(f-f_0)^{-2}+(f+f_0)^{-2}+(f^{-2}-f_0^{-2})\cos 2\epsilon}\,\sin(2\pi nf/f_0)e^{j\phi(f)}$ where $\phi(f)=\epsilon-\tan^{-1}((f-f_0)\sin 2\epsilon/(f+f_0+(f-f_0)\cos 2\epsilon))+\pi/2$ and K is is independent of both f and ϵ. If the burst is not a whole number of periods long the expression becomes more complicated.

This spectrum peaks at $f=f_0$ and the width of the peak, taken between neighbouring zeros, is f_0/n, inversely proportional to the burst length, and compatible with the requirements of the acoustic uncertainty relationship. Its shape, i.e. the variation of its modulus with f, clearly does change when the value of ϵ changes, and in addition the reference phase $\phi(f)$ of the component of frequency f depends in a non-linear fashion on both f and ϵ. If the centre of the burst occurs, not at time $t=0$, but at $t=T$, then $\phi(f)$ contains a further additive term $-2\pi fT$. If $\epsilon=\pi/2$ the spectrum of the burst decays at frequencies far from f_0 as f^{-1}, whereas if $\epsilon=0$ it decays as f^{-2}. This is understandable since in the latter case the burst has discontinuities of slope at its ends, but in the former has amplitude discontinuities, which will splash the spectrum out much further, a point about which I warned Mr Driscoll in my February letter. He doesn't have to take my word for these statements — presumably one of his brighter students could check the calculations, or he could ask one of the enterprising loudspeaker manufacturers who have set themselves up with minicomputers, f.f.t. programmes, and graphics terminals to let him see for himself what a sinewave toneburst spectrum really looks like, in phase as well as in amplitude.

It is all too easy for those acquainted in principle with Fourier transforms to mention the use of transfer functions and Fourier transforms for calculating network responses to signals of finite duration, leaving the impression that this is essentially a trivial extension of normal a.c. calculations. It is not, and exposure to the specific Fourier transforms of a few simple signals, such as tone bursts, can go a long way towards driving the point home.

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CONFUSION ABOUT DISTORTION?

In a letter in your August issue Mr Greenbank quotes an earlier correspondent who states: ". . . 'loss of information' occurs during amplifier 'latch-up' — when, as we all know, 100% intermodulation distortion occurs." This statement is symptomatic of a general confusion which has resulted from harmonic distortion, intermodulation distortion, "latch-up", "clipping", "slew-rate limiting", and transient intermodulation distortion all being regarded as "distortion".

The use of distortion as a generic term is probably responsible for it being generally unnoticed that the above list may be the