I sacrifice 3dB of signal-to-noise since impedances are not perfectly optimum and distortion at 2dB below clip is only 0.03% over 20-20kHz.

But this is the way thoughts are going in Japan.

*Tim de Paravicini,*
*London NW6.*

# ADVANCED PRE-AMPLIFIER DESIGN

In reply to Mr Williamson (Letters, April), I think there are mainly two points to be made. One, that any pre-amplifier should have adequate signal handling capacity in excess of the performance of any pickup cartridge both dynamically and in pure consideration of the amplitude of signals. Second, that as far as I am concerned the two pickup cartridges which are capable of giving peaks in excess of 200mV are the Ortofon SL15 with appropriate transformer and the Decca London cartridge.

The reference to signal peaks of 80 cm/s observed on gramophone records came from the book "Hi-Fi Systems" by G. King where there is a graph illustrating the velocities measured on gramophone records at various frequencies.

I nominate my favourite charity as the Musicians Union!

*A. J. Watts,*
*SGS-ATES (United Kingdom) Ltd,*
*Aylesbury,*
*Bucks.*

# LONG WAVES FOR AMATEURS?

I am normally in favour of amateur radio but a statement in your March 1977 issue (p. 78) that the USA may request a frequency allocation in the l.f. band for amateurs fills me with anger. How can anyone be so wickedly irresponsible or unappreciative of the value of long wave channels!

Just in case the unique feature of long wave transmission has slipped anyone's mind I would point out that the long wave channels are the only ones capable of giving reliable, fade-free global communication without resorting to the use of satellites.

In my opinion it shows a serious lack of appreciation of the potentialities of these frequencies to allow anyone to use them just for low power local broadcasting, and hence it is quite wrong to allow more than one transmitter on each channel unless the carriers are synchronised and they are radiating the same programme.

*H. G. May,*
*Barton-on-Sea,*
*Hants.*

# AUDIBILITY OF PHASE EFFECTS

In view of the continuing controversy in these columns over the audibility (and hence undesirability) of non-linear-phase shifts in an audio signal — i.e., phase shifts which leave the harmonic structure unaltered but distort the signal waveform — the following recent observations of mine may be of interest to your readers. In particular, they may enable readers who have built the *Wireless World* Dolby B noise reducer to verify some of these effects for themselves.

Having completed the noise reducer kit from Integrex Ltd., I was somewhat surprised to find that, listening to the built-in calibration signal at the monitor output (with the input selector in the auxiliary position), I could hear a subtle but distinct difference between the apparent purity of the (approximately 456Hz) tone with the record/play button in and the sound with the button out. Reference to the circuit diagram shows that the only change introduced by this switch is the insertion of a unity-gain polarity inverting stage into the output circuit. Further investigation showed that the gain of this stage was indeed unity (within 0.02dB) and its harmonic distortion very low (of the order of 0.02% t.h.d.). So it clearly was not the culprit. It was at this point that I measured the calibration oscillator t.h.d. and found that this was 2.66%, comprised of 2.57% second harmonic, 0.62% third harmonic, 0.25% fourth harmonic and approximately 0.16% higher-order harmonic distortion. The pronounced second harmonic distortion, like all even-order harmonic distortions, rendered the waveform asymmetrical; this asymmetry was sufficient to be just barely visible on an oscilloscope.

Here, then, was the explanation of the change in sound quality observed before. It is known from recent work [1, 2, 3] that the inner ear does not respond symmetrically to compression and rarefaction, and at lowish frequencies (below say 1kHz) where the rate of neuron firings can be modulated by the audio *waveform*, the ear performs to a certain extent at least like an asymmetrical waveform detector, responding more to one signal polarity than to the other. In this connection reference should be made to the publications cited in references [1, 2] and [3], and in particular to the work of J. H. Craig and L. A. Jeffress. By switching from "record" to "playback", and hence inverting the slightly asymmetrical calibration waveform, the fact that the ear treats compressions and rarefactions unequally resulted in an audible difference in the tonal quality. Of course, this polarity reversal of the asymmetrical signal is equivalent to a phase shift of the harmonics relative to the fundamental, and so this result has direct relevance to the current discussions on the audbility of phase distortion. The letter by M. A. Gerzon [4] should also be consulted for corroborative evidence.

The above explanation has subsequently been confirmed by introducing polarity reversals at other points in the reproduction chain, with the same effect. The audible effect of the polarity reversal in the Dolby noise reducer could be exactly counterbalanced by another polarity reversal later in the chain. In this way, it was possible to rule out transducer asymmetry as a contributory cause. The audibility of the polarity reversal has also been confirmed by friends on whom I have repeated the experiment.

The audibility of the polarity reversal depends to a great extent on having the volume level just right – neither too loud nor too soft. This also agreed with the earlier experiments cited. The change is audible on both headphones and loudspeakers, but for convenience the former were used primarily in my tests.

I would like to invite readers who have constructed the *Wireless World* Dolby B circuit to try this experiment themselves. Of course, I cannot vouch that the distortion of their calibration oscillators will be the same as mine and so produce the desired asymmetry! It should be emphasized that the change is *subtle*, and some perseverance may be required in order to hear the tonal difference. (Experiment also with the volume level.) The noise reduction should be switched "off." (Switching it "on" exaggerates the difference in the right-hand channel, by pre-emphasizing the higher harmonics when in the "record" mode and de-emphasizing them when in the "playback" mode. The left-hand Dolby side-processor loop is not performing its normal function when the calibration oscillator is on, and so the left-hand channel does not display this further effect. Thus it may be found helpful initially to monitor the right-hand channel output with the noise reduction switched "on," to serve as an aid in learning what to listen for. The change under these circumstances is, however, *not* a simple polarity reversal.)

At first sight, all the above would seem to bear only on the audibility of polarity reversals of non-sinusoidal waveforms. As such, it strongly suggests that an effort should be made to standardize the polarities of the whole recording/reproduction chain from microphone, through record or tape, to loudspeaker. This suggestion has been made before, for example by D. S. Stodolsky[1]. It also serves as a warning to those who conduct A/B comparison tests on audio components without taking into account the possible relative polarity reversals which such components can introduce. For example, some power amplifiers are inverting from input to output, whereas others are non-inverting. Some of the alleged differences between components compared A/B may be due to such oversights.

Our observation *does*, however, indeed bear directly on the vexed question of the audibility of non-linear-phase shifts for the following reasons. Non-linear-phase distortion results in waveform distortion, and hence can change the symmetries of the signal waveform. As shown above, such symmetry changes *can* be detected by the ear, and so such phase distortion must be classed as undesirable, whatever the component is which introduces it. So, to conclude, it is my belief that phase distortion is audible under suitable circumstances, that more effort should be devoted to obtaining bounds on the allowable phase distortion on programme material (by means of properly conducted experiments with source signals which have not been phase-distorted by the audio chain), and that in principle the goal of phase-linearity (at least over the bulk of the audio band) is a desirable one which is worth pursuing, especially in transducers.

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## References


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Further letters on the audibility of phase effects, and also letters on transient intermodulation distortion in amplifiers, will be published in a later issue.