LETTERS TO THE EDITOR

MORE COMMENTS ON "ON THE AUDIBILITY OF MIDRANGE PHASE DISTORTION IN AUDIO SYSTEMS"*

Dr. Shanefield's letter in the June issue of the *Journal*,[1] discussing the above paper, raises questions of great interest to me, my first contribution on the subject of the effect of phase shift on sound quality having been published in *Wireless World* as far back as 1956 April.

Before World War I, I was active in the design of amplifiers for sound-film recording and reproduction, but during the war part of my time was spent in the development of radar system amplifiers for waveform reproduction, and hence, requiring close control of the phase–frequency characteristics. Returning to audio problems after the war, I thought that I would achieve fame and fortune by designing cinema amplifiers with special attention to the effects of phase shift. However, on comparing the experimental zero-phase-shift amplifier (a convenient but incorrect term) with the standard amplifiers designed without reference to the effect of phase shift, it was found to our great dismay that the sound quality provided by the zero-phase-shift amplifier was no different from that using the standard units. As we could see no explanation of the findings, further consideration of the problem was postponed, but at intervals since that date I have returned to the problem. Some of the findings have been published, chiefly in *Wireless World*, but were briefly discussed in the contribution by Moir, Heyser, and Preis listed as Ref. [5] in the above letter.

At present there is no well-founded evidence that differential time delays have any observable effect on sound quality, provided they do not greatly exceed the limits set by CCIF. Indeed, these limits seem to be unnecessarily tight in my experience. We do not find that phase effects are significantly more obvious if headphones are used.

If special test-tone combinations are used, then effects can become obvious if the relative phases of the components are varied. In many examples this is due to the change in the ratio of peak to rms signals that may occur as the relative phase of the components is changed. This and distortion cancellation are, I believe, the explanation of Furindle's findings.

Proving that differential time delays are of no consequence is impossible, but one can consider the situation that would arise if they were important. The location of the individual instrument in an orchestra could only be determined by a long experimental investigation; indeed, the instruments would then have to be fixed to the floor. There would only be one seat in the house in which the music would sound as the composer intended and he would have to provide details of the hall and the seat to be used for listening to the composition.

Lipshitz mentions the problem of ensuring that the frequency response stays constant as the position of the tweeter is varied in the two-loudspeaker type of experiment. I would suggest that this is virtually impossible, for the effect of moving one unit with respect to the reference unit is to introduce "comb filter" effects into the response curve. The peak spacing would be a function of the loudspeaker spacing and the peak amplitude a function of the directivity of the two units being employed. Our experiments have produced no evidence that the comb filter effects are of any significance in this type of experiment.

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[1] D. Shanefield, S. P. Lipshitz, M. Pocock, and J. Vanderkooy, *J. Audio Eng. Soc. (Letters to the Editor)*, vol. 31, pp. 447–448 (1983 June).

COMMENTS ON "CHANNEL CODINGS FOR DIGITAL AUDIO RECORDINGS"

The audio field, as we are all aware, is fast evolving. In particular, the application of digital techniques has emerged only during the last decade. The customary audio technician, normally not familiar with details of these techniques, will certainly benefit from survey or tutorial papers on digital coding, such as modulation systems and the error correction used in optical disk and magnetic tape media.

The above paper[1] is an example of a paper intended as partly tutorial. After reading this paper I felt that some additional comment might be useful.

To start with the end of the paper, I note that the reader will miss in the references the important and pioneering work of IBM researchers in the field of modulation systems based on run-length-limited sequences. Tang and Bahl [1] gave in 1970 the basis of block codes, such as EFM used for the modulation system in the Compact Disc. Franaszek [2], [3] further developed the theory and gave examples of simple embodiments of encoders and decoders. An example of such embodiment appears in the work of his co-workers [4] who describe in 1977 a run-length-limited code with run-length constraints $(d, k) = (2, 7)$, with the feature that the decoder needs only eight observation

[1] T. T. Doi, *J. Audio Eng. Soc.*, vol. 31, pp. 224–238 (1983 Apr.).

positions so that error propagation is restricted to only four data bits. The new code described by Doi, HDM-2, has the same run-length constraints but needs (with more complex hardware) fifteen channel bits for decoding, resulting in worst case error propagation of eight data bits.

The author exaggerates somewhat the importance of the choice of a channel code. His Fig. 1 shows the packing density of magnetic storage for computers with the applied channel code as a parameter. The reader might conclude that the enormous increase in packing density results from the application of new channel codes. Nowhere in the paper is there a quantitative comparison of the maximum achievable linear packing density of different codes, except for (p. 225): "Soon direct coding of MFM was tried and the packing density increased five times. The next step was the adoption of codes of higher density ratios such as 3PM and HDM-2, with the packing density doubling or tripling."

A very well written and documented paper by White [5] gives a plot of packing density versus time, but with the correct parameter: technology such as heads and materials.

The reader who wants to choose a recording code is referred to the work of Mackintosh [6] who gives a survey of channel codes and compares the maximum achievable packing density as a function of the code. He found little difference between codes as regards the maximum packing density achievable by each. He further concludes: "The system designers should choose any code for their particular system, and the system should then be designed around the code."

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[3] P. A. Franaszek, "Run-Length-Limited Variable-Length Coding with Error Propagation Limitation," U.S. patent 3,689,899 (1972).

[4] J. S. Eggenberger and P. Hodges, "Sequential Encoding and Decoding of Variable Word Length Fixed Rate Data Codes," U.S. patent 4,115,768 (1977).

[5] R. M. White, "Disk-Storage Technology," *Sci. Amer.*, pp. 112–121 (1980 Aug.).

[6] N. D. Mackintosh, "The Choice of a Recording Code," *Radio Electron. Eng.*, vol. 50, no. 4, pp. 177–193 (1980).

AUTHOR'S REPLY

The paper was written to introduce the progress of channel codings in the field of digital audio. It was not my intention to describe the wide range of studies in various other fields (except for a very brief description of computer peripherals), and it is true that many important works were therefore not introduced. I thank Mr. Immink for drawing attention to important work and additional references that might benefit readers with deeper interests.

I disagree with the comment, "the system designers should choose any code for their particular system, and the system should then be designed around the code," for it implies that a study on channel coding is useless. Yet most "practical" system designers know that the selection of channel coding is important for the system, and that is the reason so many people devote effort to the study of channel codings. In analog signal transmission a transformer is used to match the impedance between transmission line and generator (or load). Similarly, channel coding is a method for matching the characteristics of a digital signal with various constraints of the channel (or of the recording). This case is complex because many parameters must be considered simultaneously, and a practical system must be designed with many peculiar restrictions. Some systems must run the tape exactly at a speed of 30, 15, or 7.5 in/s. Some are limited in signal-to-noise ratio but the jitter is small; others have terrible jitter but greater signal-to-noise ratio. Some have severe problems in optical skew tolerance, others with intersymbol interference caused by nonlinearity of thin film heads.

It is impossible to discuss in general which channel coding is best for recording, because the evaluation is useful only in view of the restrictions. This limitation is clearly stated in the paper (p. 225): "But it was impossible to choose the best code for magnetic recording, because the choice depends on the characteristics of the recording media and the design of the decoding circuits." If readers gained the impression that the progress in packing density shown in Figs. 1–3 was achieved only by the selection of channel codings, that was not my intention. Choice of channel coding is one of the methods that system designers have used to improve the packing density, but it is not possible to evaluate the conribution of channel coding alone.

In comparing HDM-2 and the code proposed by Franaszek, I believe the hardware is less complex in HDM-2. From my study, error propagation in general is not directly related to constraint length, that is, sometimes a code with shorter constraint length has a longer average error propagation. But I have never directly compared HDM-2 and the Franaszek code in terms of error propagation.

In closing, I thank Mr. Immink for this useful discussion, which reminds me of our long, serious, and pleasant discussion leading to the Compact Disc format. This present correspondence has also been very enjoyable.

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