LETTERS TO THE EDITOR

MORE COMMENTS ON "A DYNAMIC NOISE REDUCER FOR SUM-DIFFERENCE MULTIPLEX SYSTEMS"[1]

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In the above paper Cabot suggested the use of a variable high-frequency blend ("hi-blend") to counteract the noise in the difference channel in FM stereo reception. His proposal has the same effect on the stereo signal as filtering only the difference signal with a variable RC low-pass filter. Unfortunately RC filters not only produce attenuation of the $A - B$ difference channel, which has the desired effect of reducing noise, but they also introduce phase shifts in $A - B$ relative to the sum signal $A + B$, which has the undesirable effect of decreasing the channel separation much more than would be the case with attenuation without phase shift. Fig. 1 shows the crosstalk in decibels produced for a given difference channel attenuation for the two cases, (1) attenuation caused by an RC low-pass filter, and (2) attenuation without phase shift. In practice the poor crosstalk of RC circuitry means that if one could find a means of eliminating the phase shift, one would increase the frequency at which a given loss of stereo width occurs by about one octave without affecting the noise attenuation. This very substantial improvement can be used either to reduce the audible effects of the hi-blend, or else to give a greater noise reduction for the same audible effect.

A means of avoiding phase shifts of the low-passed $A - B$ signal relative to $A + B$ is to use phase compensation, that is, to pass the $A + B$ signal through an all-pass network having the same phase response as the low-pass filter in the $A - B$ channel. Suitable filters are the RC all-pass filter with response

$$S(\omega) = \frac{1 - j\omega\tau}{1 + j\omega\tau} \tag{1}$$

and the double RC low-pass filter with response

$$L(\omega) = \{1/(1 + jw\tau)\}^2 \tag{2}$$

where the time constant τ equals $1/2\pi F$, with F being the -6 dB point of the filter $L(\omega)$. These two filters have precisely the same phase responses, as does the double RC high-pass filter

$$H(\omega) = -\{j\omega\tau/(1 + j\omega\tau)\}^2 . \tag{3}$$

Note that

$$S(\omega) = L(\omega) + H(\omega). \tag{4}$$

Thus a way of achieving phase-compensated hi-blend is to sum-and-difference matrix the original stereo signals A and B, pass $A + B$ through $S(\omega)$, $A - B$ through $L(\omega)$, and to sum-and-difference back to stereo. However, the circuitry for this is complex, and also suffers from the disadvantage that the degree of hi-blend is not continuously variable.

A simpler means of achieving phase-compensated hi-blend, and of making it variable as well, is to pass both of the signals A and B through all-pass filters $S(\omega)$, and also to derive the difference signal $A - B$ to give it a variable gain $\frac{1}{2}g$, and to pass it through the *high*-pass filter $H(\omega)$. One then subtracts the filtered $A - B$ from the all-pass filtered A and adds it to the all-passed B, giving output signals

$$A' = S(\omega)A - \tfrac{1}{2}gH(\omega)(A - B) \tag{5}$$

$$B' = S(\omega)B + \tfrac{1}{2}gH(\omega)(A - B).$$

Then when $g = 0$, A' and B' are ordinary stereo signals, and when $g = 1$, their difference signal has effectively been filtered by $S(\omega) - H(\omega)$, which equals $L(\omega)$ from Eq. (4) so that one has phase-compensated hi-blend. For intermediate values of g between 0 and 1, one has fully phase-compensated partial hi-blend.

Thus by controlling the value of the single-gain g, the degree of hi-blend may be controlled either automatically by a level-sensing circuit as suggested by Cabot, or by a manual user control. If one wishes to avoid "pumping"

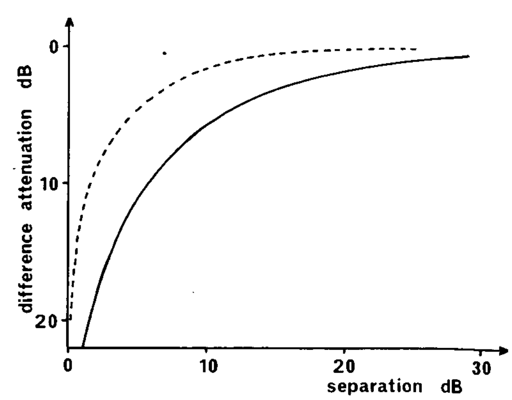

Fig. 1. Stereo crosstalk versus attenuation of difference channel. Dashed line—RC low-pass filter attenuation; solid line—attenuation without phase shifts.

[1] R. C. Cabot, "A Dynamic Noise Reducer for Sum-Difference Multiplex Systems," *J. Audio Eng. Soc.*, vol. 25, pp. 95-98 (March 1977).

and loss of subtleties of ambience, the latter option is to be preferred, although there is no reason why both manual and automatic options should not be available to the user. Besides the improved stereo width and/or increased noise suppression offered by the phase-compensated scheme, an additional advantage is that the absence of interchannel phase differences prevents any unpleasant "phasiness" effects which can make the stereo image sound less focused and more fatiguing to listen to.

A similar scheme to the above, but with the roles of low-pass and high-pass filters interchanged, can be used for phase-compensated attenuation of the difference signal in the bass for disc-cutting to prevent excessive vertical groove modulation. Again one has the advantage of the stereo width holding up to an octave lower than would be the case for simple RC blend. A third application, also in the bass, would be to CD-4 disc cutting, where a phase-compensated high-pass filtering of only the difference

$$Q = (\mathrm{LF} - \mathrm{LB}) - (\mathrm{RF} - \mathrm{RB})$$

of the two subcarrier modulations would reduce intercarrier beat distortion without excessive loss of corner-to-corner separation and insignificant effect on localization. For example, an attenuation of Q by as much as 10 dB still gives $-$ 13.7-dB corner-to-corner separation; this compares with less than 6-dB separation (for example, from LF to LB) if the bass of both subcarrier modulations is filtered.

We note that the idea of phase-compensated filtering of the sum-and-difference channels is over twenty years old (see [1]–[3]) and has been used also in the UMX system [4]. The main novelties in the preceding are the idea of having variable phase-compensated filtering and the method of implementation without sum-and-difference matrixing.

REFERENCES

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[2] D. S. McCoy, "Distortion of Auditory Perspective Produced by Interchannel Mixing at High and Low Frequencies," *J. Audio Eng. Soc.*, vol. 9, pp. 13-18 (Jan. 1961).

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[4] D. H. Cooper and T. Shiga, "Discrete-Matrix Multichannel Stereo," *J. Audio Eng. Soc.*, vol. 20, pp. 346-360 (June 1972).