Applications of Blumlein Shuffling to Stereo Microphone Techniques*

MICHAEL A. GERZON, *AES Fellow*

Oxford OX4 1XX, UK

Between 1931 and 1934 Blumlein proposed using a pair of identical ear-spaced microphones pointing in the same direction with a "shuffling" network to create stereo. Although Blumlein shuffled stereo techniques have never been used commercially, they have unique advantages. Various implementations and uses are described, including improved results from dummy-head and Theile sphere microphones and uses with boundary-layer, cardioid, and shotgun microphones.

0 INTRODUCTION

Among the many ideas in his famous stereo patent [1], Blumlein described a remarkable stereo microphone technique that, as far as the author is aware, has never been used commercially in the more than 60 years since it was invented. While the author published a series of articles on this technique more that 20 years ago [2], it is still almost unknown to the audio community as a whole.

This technique, which was termed the "Blumlein difference technique" in [2], but which I shall term "Blumlein shuffling" in the present paper, has not been unused because it is no good. Rather, the problem has been that it involves feeding a microphone array into quite a sophisticated signal-processing system, the Blumlein shuffler, in order to work, and it has not been easy to do this signal processing on commercially available equipment. Even the much simpler signal processing required for the MS microphone technique [3], [4] delayed that system's widespread use for many years, and the only widely used microphone technique involving very sophisticated postmicrophone signal processing has been the sound-field microphone [5], [6], which uses quite elaborate frequency-dependent matrixing to achieve improved polar-pattern accuracy and spacial coincidence of microphone outputs.

Any microphone technique that requires complicated postmicrophone signal processing is likely to be widely used only if equipment becomes available commercially, allowing the signal processing to be done even by users who do not fully understand the theory of operation. Such equipment has not been, and is not, available for the Blumlein shuffling technique.

However, because this technique has some unique advantages and possibilities not shared by any other stereo microphone technique, the author feels that it is important to make the audio community more widely aware of its uses, and to suggest some simple practical methods of implementing it that may be accessible to those wishing to experiment with it. It is the writer's hope that a greater awareness of Blumlein shuffling may cause suitable equipment to become commercially available, making it accessible to the general recording engineer.

Before outlining the technique itself, we list some of the unique things of which it is capable. One application is to improve stereo localization quality and accuracy from recordings made with a binaural dummy-head or the Theile sphere microphone [7], or other similar microphone techniques using ear-spaced omnidirectional microphones separated by a baffle. This was the original application of Blumlein in [1].

To obtain stereo from a closely spaced pair of boundary-layer microphones is another application. A third application is to obtain a wide stereo pickup with an exceptionally high degree of rejection of unwanted sounds from the side and the rear, using available high-quality microphones; thus obtaining stereo directivity factors larger than is possible by means of conventional stereo microphone techniques. The most extreme example of

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the use of Blumlein shuffling is a practical stereo shotgun microphone technique, allowing the pickup of true stereo at great distances from the sound stage, with a wide stereo presentation and sharp stereo images.

The main disadvantage of the Blumlein shuffling technique is that it has poor mono compatibility, and so is mainly useful either for media where mono compatibility is of secondary importance (such as CDs), or as a subsidiary part of an overall mix (such as an "atmosphere" pickup in TV drama or the "backline" pickup in a live rock recording).

In outline, Blumlein shuffling uses two identical microphones (which may be, but need not be, separated by a baffle), spaced apart by a distance typically on the order of 200 mm, both pointing in the same direction. The outputs of these microphones are then postprocessed by a Blumlein shuffling network that converts low-frequency phase differences between the two microphone outputs into amplitude differences suitable for stereo loudspeaker reproduction.

The Blumlein shuffler should not be confused with the shelf-filter shuffler [8]–[10] used for widening conventional stereo at frequencies below 700 Hz, since such shelf-filter shuffling is designed not to convert phase differences into amplitude differences, but simply and solely to increase stereo width by a factor of between 1.6 and 2.5 at low frequencies.

Rather, Blumlein shuffling is an early approximation to the matrixing used to convert binaural dummy-head recordings into transaural signals [11], [12] intended for reproduction via a stereo pair of loudspeakers. With the technology of Blumlein's era in 1931, full binaural-to-transaural conversion was not feasible, except only at low frequencies below around 700 Hz, and Blumlein shuffling was the best approximation available with the technology of that era.

This paper first describes the principle of the Blumlein shuffler, how it converts phase differences to amplitude differences, and the kind of adjustments available, in particular for stereo width. The application of Blumlein shuffling to various microphone types, ranging from omnidirectional to shotgun microphones, is discussed, with calculations of the directivity factor (rejection of unwanted sounds) and stereo stage width, and some comments on subjective aspects. The paper then describes an alternative method of Blumlein shuffling, which can be implemented by means of a passive network after some microphone preamplifiers with electronically balanced outputs.

1 PRINCIPLES OF BLUMLEIN SHUFFLING

1.1 Shuffling

The basic concept of shuffling, devised by Blumlein [1] in 1931 but given this name publicly only in [8] by the EMI team who worked with him, is shown in Fig. 1. A stereo signal in left–right form is converted by an MS, or sum-and-difference, matrix into the form of a sum, or M, signal and a difference, or S, signal, and the two signals M and S are subjected to the action of nonidentical filters and then converted back by a second MS, or sum-and-difference, matrix into left–right form.

In this paper we shall not confine the term shuffling to a system implemented as in Fig. 1, but to any stereo-in–stereo-out linear processing that has the same *effect* on signals as does Fig. 1. While implementation as in Fig. 1 is often convenient, there are other methods of implementation, such as that presented in [10].

It is convenient to adopt gain and polarity conventions to define the MS matrix as follows:

$$M = 0.7071(L + R) \qquad S = 0.7071(L - R) \quad (1)$$

where L and R are the input left and right signals and M and S are the respective sum and difference signals. The reason for adopting the factor $0.7071 = 2^{-1/2}$ is simply that the inverse MS matrix has exactly the same form, that is,

$$L = 0.7071(M + S) \qquad R = 0.7071(M - S). \quad (2)$$

Other gain and polarity conventions could be adopted, but we find this the most convenient one to use, despite its complicated factor, because 1) the MS conversion matrices to and from the MS mode are identical, meaning that there is no need for additional gain adjustments in conversion between MS and LR modes, and 2) the total energy $L^2 + R^2$ of a stereo input signal is preserved by MS matrixing, that is,

$$L^2 + R^2 = M^2 + S^2 \quad (3)$$

so that the total energy of a signal can be studied equally well in either mode.

The simplest use of signal processing in the MS mode is the *stereo width control* devised by Blumlein [1], shown in Fig. 2. Here the gain of the S signal is varied but that of the M signal kept fixed. For gains *w* greater than 1 for the S signal, this causes an increased width of presentation for sounds panned across the sound stage, and for *w* between 0 and 1 it gives a reduced width. Width control is one of the most practically useful adjustments of stereo images, and so it is surprising that this ancient 1931 technique is still not incorporated in many mixing consoles and is rare in consumer equip-

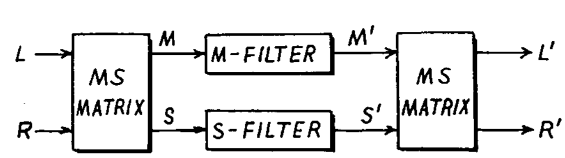

Fig. 1. Schematic of a shuffler.

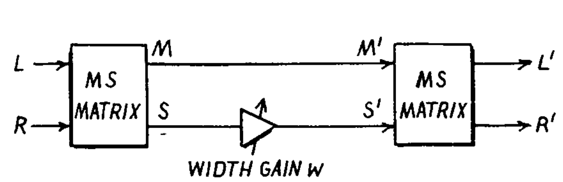

Fig. 2. Width control for amplitude stereophony.

ment. This is presumably because there is a legacy of thinking of "stereo" as being two independent channels, dating back to the early consumer stereo era, rather than thinking of it as interrelated channels to be signal processed as a whole.

The filters in a shuffler, as in Fig. 1, affect the relative gains of the M and S signals, which affect the width, but they affect the relative *phases* of the M and S signals as well. There has been a tendency to concentrate purely on the effect of the relative gains in the M and S signal paths, whereas often the effect on the relative phases can be more significant.

To take an extreme example, if the gains of the M and S signal paths are unchanged, with value 1, but the phase of the S signal path is changed by 180°, that is, its polarity is inverted, then the output left and right signals L′ and R′ have the form

$$L' = 0.7071[M + (-S)] = R$$

$$R' = 0.7071[M - (-S)] = 0.7071(M + S) = L \tag{4}$$

which has the effect of interchanging the left and right channels.

Smaller phase changes also have significant effects. For example, a 90° phase shift in the S signal path eliminates all separation of a conventional amplitude-panned stereo input, giving left and right output signals that differ only in phase, that is, amplitude differences between the input left and right channels are converted to phase differences. In a similar way, a phase shift in the S signal path can convert phase differences between the input channels to amplitude differences at the output. Even smaller phase shifts in the S channel will, in general, have the effect of reducing the separation of amplitude-panned signals, as noted in [10], for example, and also of increasing amplitude differences of input signals differing only in phase.

Thus in considering shuffling, it is important to be aware not only of the amplitude responses of the shuffling filters, but also of their phase responses.

For input stereo signals incorporating both amplitude and phase differences, the effect of phase shifts in the S signal path is in general complicated, requiring a detailed calculation of the complex gains of signals to determine the amplitude and phase differences at the output. Nevertheless, as was already known to Blumlein [1] in 1931, phase shifts in the S signal path can have the effect of converting input phase differences into output amplitude differences and vice versa. This effect of shuffling has often been little understood, and it is obscured if one concentrates only on the amplitude response of filters in the M and S signal paths.

The generally complicated behavior of shuffling means that it can be designed for a number of different applications. By minimizing or eliminating phase differences between the M and S channels and simply varying the relative gains with frequency, one can have a frequency-dependent width control for altering the reproduced width of input stereo signals that are panned by amplitude, for example, with panpots or with coincident directional stereo microphone pairs. This frequency-dependent width control is the principle behind what Griesinger called *spatial equalization* [9], also used by Clark et al. [8] and the author [10], whereby lower frequencies, below around 600 or 700 Hz, are reproduced with a greater relative width gain w (equal to between 1.6 and 2.5) than higher frequencies. Such an increase of lower frequency width is found to give an increased sense of spaciousness and some compensation for the different stereo localization psychoacoustics of lower and higher frequencies. However, in such frequency-dependent width adjustment applications to amplitude stereophony, phase shifts in the difference channel are undesirable, as explored in [10], [13].

The original application of Blumlein [1] processed input signals differing mainly in phase, not amplitude, and in his application, as we shall see, phase shifts in the filters were not merely desirable, but absolutely essential. In [14] Griesinger quotes the present author as having claimed that elimination of phase shifts was desirable in this application—something that is simply not true.

In this paper we adopt the term Blumlein shuffling to mean shuffling in which phase shifts in the S signal path relative to the M signal path are used to convert phase differences in the input stereo signals into amplitude differences in the output stereo signals. This is to be contrasted with those kinds of shuffling in which phase shift differences between the S and M signal paths are desirably minimized, designed to adjust the width at different frequencies of amplitude-panned stereophony. Griesinger's cited paper [14] appears to confuse these two quite distinct kinds of shuffling, and rules applicable to the latter width-adjustment type of shuffling are inappropriate to the former Blumlein shuffling, which converts phase into amplitude differences.

1.2 Blumlein Shuffling for Time Differences

If one has two identical microphones spaced apart laterally and pointing in the same direction, then provided the spacing is small compared to the sound source distance, the signals arriving at the two microphones will differ in time of arrival but not in amplitude. Blumlein realized in [1] that such time-of-arrival differences could, at lower frequencies, be converted into amplitude differences for feeding to loudspeakers.

Consider a signal arriving with amplitude gain 1 in two stereo channels L and R, but with a time of arrival in L of T before arrival in R. Then it is convenient to consider the left channel as having a time advance of $\frac{1}{2}T$ and the right channel a time delay of $\frac{1}{2}T$, since this makes all mathematics more symmetrical.

The Blumlein shuffler for such time-delayed signals consists of an RC boost in the S signal path relative to M, as shown in Fig. 3, with equalization time constant τ, corresponding to a +3.01-dB boost frequency (or frequency constant),

$$F = \frac{1}{2\pi\tau} . \tag{5}$$

The complex gains at angular frequency ω of the L and R signals are, respectively,

$$L = \exp(\tfrac{1}{2}j\omega T) \qquad R = \exp(-\tfrac{1}{2}j\omega T) \tag{6}$$

whose M and S signals have respective complex gains,

$$M = \sqrt{2}\cos(\tfrac{1}{2}\omega T) \qquad S = \sqrt{2}\, j \sin(\tfrac{1}{2}\omega T) \tag{7}$$

by using $\exp(jk) = \cos k + j \sin k$ for real k, where $j = \sqrt{-1}$.

At low frequencies, for which $\tfrac{1}{2}\omega T$ is small, Eqs. (7) become approximately

$$M = \sqrt{2}\,[1 - \tfrac{1}{2}(\tfrac{1}{2}\omega T)^2 + \cdots]$$

$$S = \sqrt{2}\, j\,[(\tfrac{1}{2}\omega T) - \tfrac{1}{6}(\tfrac{1}{2}\omega T)^3 + \cdots) \tag{8}$$

by power series expansion, so that at the lowest frequencies, M has an approximately flat frequency response, whereas S has a 90° phase shift j, as well as a 6.02-dB-per-octave bass cut.

The bass boost in the Blumlein shuffler is intended to compensate both for this bass cut in the S signal path and for the 90° phase shift. The equalization of this bass boost has gain

$$1 + (j\omega\tau)^{-1} \tag{9}$$

in the S signal path and 1 in the M signal path, giving complex gains

$$M' = \sqrt{2}\cos(\tfrac{1}{2}\omega T) = \sqrt{2}\,(1 + \cdots)$$

$$S' = \sqrt{2}\, j[\sin(\tfrac{1}{2}\omega T)][1 + (j\omega\tau)^{-1}]$$

$$= \sqrt{2}\left(\frac{\tfrac{1}{2}T}{\tau} + \tfrac{1}{2}j\omega T + \cdots\right) \tag{10}$$

so that at low frequencies, M′ has gain $\sqrt{2}$, S′ has gain $\sqrt{2}(\tfrac{1}{2}T/\tau)$, and the output channel gains are

$$L' \cong 1 + \frac{\tfrac{1}{2}T}{\tau} \qquad R' \cong 1 - \frac{\tfrac{1}{2}T}{\tau}\,. \tag{11}$$

What this has shown is that the Blumlein shuffler has the effect at low frequencies of converting a time delay T between input stereo channels L and R into amplitude gains, as in Eqs. (11), which depend on the time constant

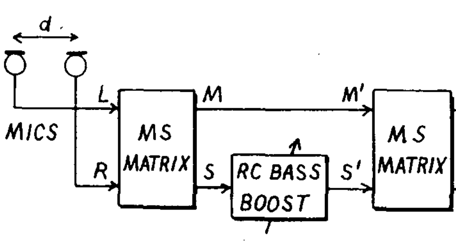

Fig. 3. Basic Blumlein shuffler for a pair of laterally spaced identical microphones pointing in the same direction, converting interchannel delays into amplitude differences.

τ of the shuffling filter.

From Eqs. (11) we can draw various conclusions. First, adjustment of the time constant τ of the Blumlein shuffler acts, at low frequencies, as a width control for signals arriving with a pure time difference. Thus in Blumlein shuffling, width control is achieved not by adjustment of the amplitude gain of the S signal, but by adjustment of the time constant τ of the S signal filter. This observation is important in the following.

Second, the overall energy gain at low frequencies according to Eqs. (11) is not equal to 2, as in the input signals, but it is equal to

$$2\left[1 + \left(\frac{\tfrac{1}{2}T}{\tau}\right)^2\right].$$

This results in an increased gain at low frequencies by a factor of

$$1 + \left(\frac{\tfrac{1}{2}T}{\tau}\right)^2 \tag{12}$$

as compared to the high-frequency energy gain, which equals 2, since at high frequencies the shuffling filter has gain 1 and virtually no phase shift, so leaves the input signals unaltered.

The reason for using a shuffler filter whose high-frequency gain is 1 is that any other choice would cause "comb-filter" coloration effects at high frequencies, whereas the choice of 1 preserves the input signals and simply preserves the time delay between the two stereo channels.

Third, it will be seen from Eqs. (11) that a single Blumlein shuffler converts, at low frequencies, different interchannel time delays T into different channel amplitude gains, so that time-delay stereophony is converted at low frequencies to amplitude stereophony.

However, what is not clear without further analysis is at what frequency a frequency ceases to be "low." The general mathematical formula for the complex gains of the left and right output channels is

$$L' = \exp(\tfrac{1}{2}j\omega T) + \frac{\sin(\tfrac{1}{2}\omega T)}{\omega\tau}$$

$$R' = \exp(-\tfrac{1}{2}j\omega T) - \frac{\sin(\tfrac{1}{2}\omega T)}{\omega\tau} \tag{13}$$

from which it is possible to compute as a function of frequency the relative amplitude gains and phases of the two output channels as a function of frequency for any given values of T and τ.

Roughly speaking, the "transition frequency" between the low-frequency amplitude stereophony region and the high-frequency time-delay stereophony region occurs at that angular frequency for which

$$\frac{|L'|^2 - |R'|^2}{|L'|^2 + |R'|^2} \cong \tfrac{1}{2}\left[\frac{T/\tau}{1 + (\tfrac{1}{2}T/\tau)^2}\right]. \tag{14}$$

Generally speaking, one is likely to choose the time constant τ such that

$$\tau \geqslant {}^1\!/_2\,|T| \tag{15}$$

so as to avoid overwide low-frequency stereo images according to Eqs. (11). Under these conditions one finds approximately that the transition frequency is almost entirely dependent on T, being around

$$F_{\mathrm{transition}} = 0.3/T\ . \tag{16}$$

In other words, the smaller the time delay T between the stereo channels, the higher the frequency at which Blumlein shuffling is effective at converting time delays into amplitude stereophony. According to Eq. (14), the transition frequency is slightly increased, to about

$$0.39/T \tag{17}$$

when T is such that the low-frequency image is panned right over to one side or the other of the stereo stage, but Eq. (16) is not too far out even here.

Thus for a time delay of 0.3 ms, quite a typical value for ear-spaced microphones with an angle of arrival of about 30°, the transition frequency between amplitude and time-delay stereophony is at around 1 kHz according to Eq. (16), but is reduced to around 500 Hz for sounds arriving from the sides and increased to 2 kHz for sounds arriving from azimuth angles of around $\pm 15°$ from due front.

Since conventional amplitude stereophony starts giving a poor directional illusion in the region of 1–2 kHz, these figures suggest that a typical microphone spacing of about 200 mm (roughly ear spacing) is about ideal for use with Blumlein shuffling. For smaller spacings the amplitude stereophony region extends to higher frequencies, which is OK except that the amount of bass boost introduced by the shuffling increases bass noise and the effects of gain mismatches between microphones unacceptably [2]. Conversely, a wider microphone spacing, with larger time delays, reduces the transition frequency between amplitude and time-delay stereophony to lower values than those found in human hearing.

It is of interest to speculate whether Blumlein [1] was aware of the optimality of near earspacing of microphones for use with Blumlein shuffling since he did suggest this spacing.

1.3 Application Problems

The preceding theory means that one might consider the use of Blumlein shuffling with a pair of identical microphones spaced apart by a distance d, pointing in the same direction, as shown in Fig. 3, where the sum signal is unaltered and the difference signal given an RC bass boost below a frequency chosen according to the desired width of stereo presentation in the bass. As we saw in Section 1.2, the boost frequency acts as a width control. However, too high a boost frequency causes excessive width, as well as a high degree of bass boost with attendant amplification of all unwanted error signal components in the difference channel, including both noise and errors due to slight mismatches between the two microphones.

The problem of bass boost of errors means that in practice we have to limit the bass boost. In practice, we have found that we agree with Griesinger's figure in [14] of limiting the S signal path bass boost to around 15 dB. Using conventional RC filtering circuity means in effect putting an additional RC bass cut with a cutoff frequency of about 0.18 times the frequency F at which the RC boost occurs.

The effect of this additional RC cut is that the stereophony at extremely low frequencies reverts to time-delay stereophony, but with an effective microphone spacing around 5.6 times as large as the actual microphone spacing, due to the gain boost. Thus for an 180-mm microphone spacing, at extremely low frequencies, the effective microphone spacing is increased to 1 m. At middling frequencies up to the transition frequency of Eq. (16) the results are those of amplitude stereophony, and at higher frequencies the results are again time-delay stereophony, but now with a spacing of only 180 mm.

However, this extreme-bass RC bass cut degrades the effect of the amplitude stereophony at midfrequencies, since it introduces additional, and now unwanted, phase shifts in the S signal path. If instead one uses a phase-compensated high-pass filter, as shown in Fig. 4, to prevent excessive bass boost in the difference channel, where the M signal path now contains a unity-gain all-pass network whose phase response is matched to that of the high-pass filter in the S signal path, then the effect of this extra phase shift is minimized. The author has previously described [2], [15] how such phase compensation can be achieved (see also [16]), and showed in [15] (in another application) how this increases the effective bass separation by about an octave for a given degree of limitation of response. See also [16] for a detailed discussion of the benefits of using all-pass phase compensation in the M channel when putting high-pass filtering in the S channel.

A disadvantage of all-pass phase compensation in the sum channel is that it alters the phase response in the bass, which is known [17] to have audible effects on bass quality when used in digitally phase-equalized sound reproduction systems [17] of the kind likely to become common in the future. However, in digitally implemented shufflers it is possible to devise phase-linear

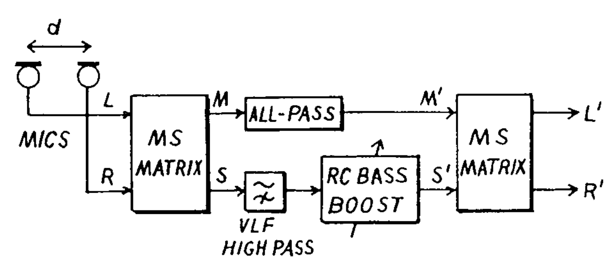

Fig. 4. Addition of a very low-frequency bass cut to limit bass-boost amplitude in Blumlein shuffler, also showing all-pass phase compensation (optional) for this cut in sum channel.

high-pass filters for the difference channel, so that the sum-channel all-pass is a pure time delay with no phase distortion.

Besides the bass boost problem, we noted in Eq. (12) that the effect of Blumlein shuffling was also to introduce a degree of bass boost in the frequency response, depending on the reproduced stereo position, which in general makes the sound slightly bass heavy. This can be corrected by means of a slight bass cut in both output channels, but is better done by introducing a slight bass cut in the M signal path, as shown in Fig. 5.

The bass cut filter also introduces small phase shifts so that the time constant of the *RC* boost in the difference signal path needs to be slightly altered to maintain the correct phase difference for Blumlein shuffling to work correctly. The bass cut filter in the sum channel is a shelf filter, typically with a gain of around -1.0 dB at lower frequencies and 0 dB at higher ones. Since the amount of bass cut required depends on the stereo position (being 0 dB at the center of the stereo stage and -3 dB at the edge), the value chosen is a compromise to give an "average" flat frequency response, averaged over a range of directions. As we shall see later, this average varies with the characteristics of the microphones.

Fig. 5 shows a Blumlein shuffler, incorporating a shelf-filter bass cut in the sum channel, an *RC* Blumlein shuffling bass boost in the difference channel, and a high-pass filter to prevent excessive bass boost in the difference channel (limiting boost to around 15 dB), preferably with an all-pass filter in the sum channel to compensate for any phase shift in the difference-signal high-pass filter. The detailed design of these filters depends on the application and will be detailed later in the paper.

1.4 Analysis of Arrival at Microphones

We now look at the effect of Blumlein shuffling from another point of view—that of the polar diagrams of the resulting signal pickup, as was done in [2], and also by Blumlein in (unpublished) work at EMI in the period up to 1934. Fig. 6 shows two identical microphones, spaced apart laterally by a distance d, responding to a sound arriving at an azimuth angle θ (measured anticlockwise) from due front. Writing the speed of sound in air as c (it typically equals around 340 m/s), then, relative to the center of the microphone array (see Fig. 6), the sound arrives at the left microphone with a time *advance* equal to

$$\tfrac{1}{2}T = \left(\frac{\tfrac{1}{2}d}{c}\right)\sin\theta \qquad (18a)$$

and at the right microphone with a time *delay* of

$$\tfrac{1}{2}T = \left(\frac{\tfrac{1}{2}d}{c}\right)\sin\theta\ . \qquad (18b)$$

Equations (18) remain true for sounds not in the horizontal plane, provided that θ is the angle of arrival measured from the median plane of the two microphones.

We suppose that the two microphones have identical polar characteristics, and thus assume that the polar diagram in the horizontal plane has the (generally complex) gain $g(\theta)$ as a function of arrival angle θ. Then the gain, as a function of direction in the horizontal plane, of the outputs of the two microphones is

$$\mathrm{L} = g(\theta)\exp\left[j\omega\left(\frac{\tfrac{1}{2}d}{c}\right)\sin\theta\right]$$

$$\mathrm{R} = g(\theta)\exp\left[-j\omega\left(\frac{\tfrac{1}{2}d}{c}\right)\sin\theta\right] \qquad (19)$$

where ω is the angular frequency of the sound, and $g(\theta)$ is the polar gain pattern at that frequency.

As before, the effect of Blumlein shuffling using an S signal path *RC* bass-boost equalization with time constant τ, as in Eq. (9), is to give, at sufficiently low angular frequencies ω, outputs L′ and R′,

$$\mathrm{L}' = g(\theta)\left[1 + \left(\frac{\tfrac{1}{2}d}{c\tau}\right)\sin\theta\right]$$

$$\mathrm{R}' = g(\theta)\left[1 - \left(\frac{\tfrac{1}{2}d}{c\tau}\right)\sin\theta\right] \qquad (20)$$

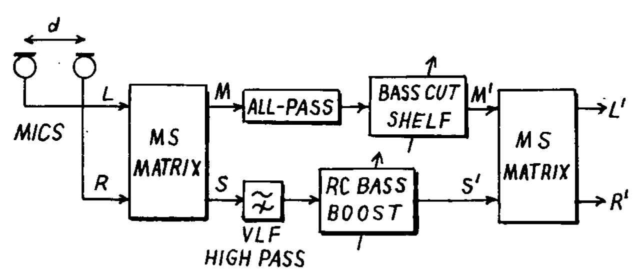

Fig. 5. Blumlein shuffler, incorporating shelf-filter equalizer in M signal path to restore flatness of average bass response.

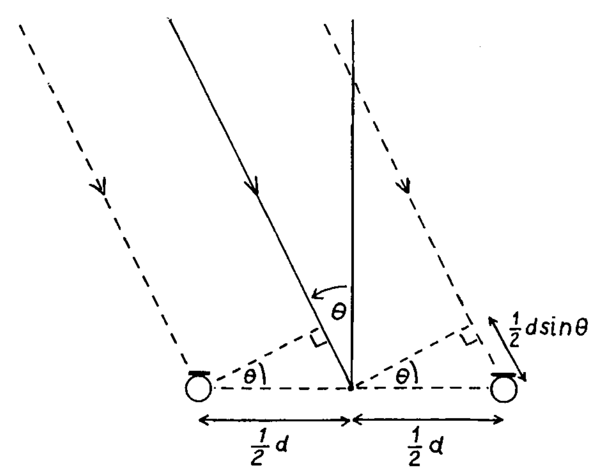

Fig. 6. Geometry of sound arrivals at a pair of laterally spaced identical microphones spaced apart by a distance d, from an azimuthal direction θ (measured anticlockwide from due front).

using Eqs. (11). This has sum and difference polar diagrams

$$\begin{aligned} \mathrm{M}' &= \sqrt{2}\, g(\theta) \\ \mathrm{S}' &= \sqrt{2}\, g(\theta) \left[\left(\frac{{}^{1\!/\!2}d}{c\tau} \right) \sin\tau \right] . \end{aligned} \tag{21}$$

In the simplest case, where the microphones are omnidirectional and $g(\theta) = 1$, Eqs. (20) describe the polar diagrams of back-to-back subcardioid, cardioid, or hypercardioid microphones, since $\sin\theta$ is the polar diagram of a leftward-facing figure-of-eight microphone. The two microphone characteristics are back-to-back cardioids if

$$\frac{{}^{1\!/\!2}d}{c\tau} = 1 \tag{22}$$

and the width of the stereo image is widened or narrowed proportional to $1/\tau$.

The reproduced stereo position of arriving sounds does not at all depend on the identical polar diagrams $g(\theta)$ of the microphones, which only affect the amplitude and phase responses of the sound. From Eqs. (20) or (21), if the time constant τ of the Blumlein shuffling is adjusted to satisfy Eq. (22) or, equivalently, if its frequency constant, Eq. (5), is adjusted to equal

$$F = \frac{c}{\pi d} = \frac{108.2}{d} \tag{23}$$

where d is measured in meters, F in hertz, and the speed of sound c in air is assumed to be 340 m/s, then sounds arriving from the sides (that is, the azimuths $\pm 90°$) are reproduced at the left and right extremes of the stereo stage, and sounds with $\theta = \pm 45°$ are reproduced (at low frequencies) with a stereo separation of 15.3 dB, which gives a fairly wide stage width in stereo reproduction. The gain of the ratio of difference to sum,

$$\frac{\mathrm{L}' - \mathrm{R}'}{\mathrm{L}' + \mathrm{R}'} = \frac{\mathrm{S}'}{\mathrm{M}'}$$

in this case equals

$$\frac{\mathrm{S}'}{\mathrm{M}'} = \sin\theta \tag{24}$$

so that, for example, it equals one-half for $\theta = 30°$, giving a 9.54-dB stereo channel separation.

Decreasing the time constant of the Blumlein shuffling increases the reproduced width, with, for example,

$$F = \sqrt{2}\frac{c}{\pi d} = \frac{153}{d} \tag{25}$$

giving an infinite stereo separation for azimuths $\pm 45°$, an increase in stage width of $\sqrt{2}$, and an antiphase crosstalk of -15.3 dB for azimuths of $\pm 90°$. For omnidirectional microphones the polar diagrams are the back-to-back hypercardioids at low frequencies, having nulls 135° off axis. We shall regard the shuffling satisfying Eqs. (22) or (23) as being a standard "reference" value. Smaller time constants and higher frequency constants have the effect of introducing antiphase sound pickups at azimuth angles around 90°, whereas larger time constants and lower frequency constants do not fill up the stereo stage.

However, if the polar diagrams $g(\theta)$ of the original microphones are highly directional, pointing toward the front, then sounds from larger azimuths will be attenuated because they arrive off axis, so that in this case the antiphase reproduction of larger azimuths is not so important, and greater widths (smaller time constants) can be used with Blumlein shuffling, allowing a wide reproduced stage for a narrow original angular sound stage. This is particularly useful for the case when the original microphones are gun microphones used at a great distance from the sound source.

1.5 Baffles

Blumlein shuffling can also be used, as was the case with Blumlein's preferred technique in [1], with an acoustic baffle placed between the microphones. At the lower frequencies where Blumlein shuffling is effective, the baffle provides relatively little acoustic obstruction, so that the difference at the two microphones remains a phase difference, whereas at the frequencies at which the baffle provides acoustic obstruction and creates amplitude differences, the effect of the Blumlein shuffling is no longer great. Thus by use of a baffle with Blumlein shuffling, amplitude differences are produced between the output stereo channels over the whole frequency range, with the baffle taking over at frequencies at which the Blumlein shuffling becomes ineffective at converting phase differences into amplitude differences.

However, the optimum form of baffle is not at all clear. Blumlein originally used a rigid circular disk placed in the median plane between the microphones, a technique that still remains popular among some European broadcasters such as the BBC (although they omit the shuffling), whereas other forms of baffle include a dummy head, a solid sphere with the microphones surface-mounted [7], and specialized proprietary designs like the SASS microphone [18], [19]. With the reservations noted later, such baffled microphone arrays are almost always improved by use with appropriate Blumlein shuffling to restore the stereoism of lower frequencies.

The key word in the last sentence is "appropriate." For good results it is important to match the lower frequency stereo effect to that obtained over loudspeakers at higher frequencies. There are two main pitfalls here: 1) not equalizing the bass shelf boost caused by the boost of the difference signal, for example, by means of a shelf bass cut in the S channel as in Fig. 5, and 2) using an inappropriate value of the microphone separation d in calculating the equalizer time or frequency constant of

Eqs. (22) or (23).

It is not true for baffled microphones that the acoustically appropriate distance d between the microphones is their actual physical distance apart. The general effect of a baffle is to increase the effective acoustic separation at low frequencies. In general, the problem of computing the effective low-frequency acoustic separation d of baffled microphones has not been solved, and this may have to be determined by acoustical measurement procedures too complicated to be described here. However, the answer is known for microphones mounted on a solid sphere, using methods developed by Lord Rayleigh in the last century [20], where it turns out, as noted by Cooper [21], that the effective acoustical distance d between microphones mounted on opposite sides of a hard sphere is $1\frac{1}{2}$ times the diameter D of the sphere, that is,

$$d = 1\tfrac{1}{2}D \tag{26}$$

for a hard sphere.

2 PRACTICAL MICROPHONE TECHNIQUES

2.1 Subjective Aspects

The writer had assumed when writing [2] in 1970 that the results of using Blumlein shuffling on a pair of identical forward-facing microphones without any baffle would not be acceptable, due to the lack of high-frequency amplitude differences, and there proposed using two sets of spaced microphones, each with their own Blumlein shuffling, with a crossover network ensuring that the array with smaller spacing took over at higher frequencies where the array with larger spacing failed to have good separation. (Blumlein suggested something similar in [1].)

It was therefore something of a surprise to discover that the subjective quality of the stereo produced by Blumlein shuffling of an ear-spaced pair of identical microphones was much better than expected, despite the poor amplitude separation at high frequencies and the small interloudspeaker time delays (typically below 0.5 ms).

The reason why Blumlein shuffling works better than it should without baffles is not fully understood, but appears to be related to the failure of amplitude stereophony to give good directional illusions in the region around 1–2 kHz. It appears that the departure from amplitude stereophony given by Blumlein shuffling of ear-spaced microphones actually gives *better* localization in the region around 1–2 kHz than does amplitude stereophony. In particular, the computed interloudspeaker phase and amplitude differences as a function of frequency (reported in Tables 1–6) seem to conform quite well with the values for optimum localization computed by Cooper and Bauck [22]. While Blumlein shuffling of microphones spaced apart by around 180 or 200 mm is not claimed to give perfect stereophony, it is at least as good as many other highly regarded near-coincident microphone techniques, and is a great improvement, in both spaciousness and localization quality, over the same microphone arrays without Blumlein shuffling.

British recording engineer Tony Faulkner has proposed what he termed a "phased array" stereo microphone technique (not to be confused with phased arrays in the beam-forming and steering sense), which comprises two identical forward-facing figure-of-eight microphones spaced apart side by side by about 180 mm. While in some situations such "phased arrays" do give a certain spatial quality, the application of Blumlein shuffling brings them to life with a much more dramatic directional effect and spaciousness, without the loss of whatever virtue they may have possessed without any shuffling.

Another factor that may help explain the subjective performance of nonbaffled Blumlein-shuffled identical microphone pairs is that, as we saw in Section 1.4 and Eq. (16), the frequency at which shuffling restores amplitude stereophony is increased toward the center of the stereo stage, being above 2 kHz over the center quarter of the stage. Thus if recordings are made of sound stages with a number of sound sources near the center of the stage, then such images will be quite well handled by Blumlein shuffling up to 3 kHz or better. Also, at the highish frequencies for which Blumlein shuffling starts failing near centerstage, the ears tend to increase the perceived stage width for a given amplitude ratio between loudspeaker outputs. Thus a variety of psychoacoustic phenomena may be conspiring to improve the subjective quality of stereoism without a baffle.

From a subjective point of view, when used with dummy-head or sphere baffles, it is found to be very important to incorporate a shelf-filter bass cut to compensate for the boost in bass energy caused by the Blumlein shuffling, since this is otherwise found to cause an unpleasant heavy bass quality that ruins the tonal balance. With care it is possible to equalize the bass such that there is an insignificant tonal difference between the unshuffled and Blumlein shuffled outputs from the microphones, which is generally desirable. This is possible despite the fact that most sound directions in fact have an unflat energy response according to Eq. (12) since the ears generally seem reasonably happy if the *average* response over all directions is flat, as the author also noted in [13]. Our experience is that care to equalize bass to be on average flat within a small fraction of a decibel is well worthwhile from a quality point of view, and to avoid introducing unwanted variables that confuse the comparison of different microphone techniques.

One thing well worth experimenting with is decreasing the time constant τ of the Blumlein shuffling filter, that is, increasing its +3.01-dB frequency constant F so as to give increased width. For the outer images in a stereo stage, this increase of width is confined to frequencies below around 700 Hz, although it affects frequencies up to 1.5 or 2 kHz near stage center. Thus the effect is similar to that of spacial equalization [8]–[10] in increasing the width only of lower audio frequencies. This can yield a dramatic increase in the sense of being "enveloped" by ambient and environmental sounds in a recording, and has been found to be particularly effective with recording arrays using baffles, such as many BBC sound effects recordings. However, it is important to use a greater degree of shelf-filter bass cut for higher frequency constants, since otherwise the large bass boosts give a very heavy bass.

2.2 Analysis of Shuffled Arrays

In order to describe the use of Blumlein shuffling with individual microphone arrays and characteristics, we need to specify the frequency constant F to be used for a given spacing and width and the associated bass cut filter in the M signal path to ensure flat average response. We also need to specify the performance of the shuffled array, including such parameters as image width and directivity factor, as well as total energy gain in different directions (important if microphone directivity is being used to exclude unwanted sounds in the recording location). The details of this analysis will be given elsewhere, but here we summarize the information we shall tabulate for different microphone systems for stereo. In order to help make comparisons, we also include data for a few existing microphone techniques not involving shuffling.

For microphones without baffles spaced apart by a distance d, with the frequency constant F given by Eqs. (22) or (23) (so that the sounds from azimuths $\pm 90°$ come from the left or right loudspeakers at low frequencies), Table 1 gives the stereo separation (that is, amplitude ratio) of the two output channels when a simple RC boost is used in the difference channel for shuffling. This separation is shown for azimuths 0–90° and for frequencies as a multiple of F. By way of example, $F = 601$ Hz for a spacing of 180 mm, $F = 721.5$ Hz for a spacing of 150 mm, $F = 541$ Hz for a spacing of 200 mm, and $F = 361$ Hz for a spacing of 300 mm between microphones. Table 1 shows clearly that Blumlein shuffling gives reasonable amplitude stereophony for sound near the center of the stage up to quite high frequencies, but that separation suffers at the larger azimuths at much lower frequencies. However, it should be remembered that interchannel phase differences are also important in determining the stereo effect. The corresponding interchannel phase differences in degrees are shown in Table 2. Phase differences near 90° at low frequencies have the effect of increasing the perceived stereo width and separation. Tables 1 and 2 have been computed from Eqs. (13) with

$$\tfrac{1}{2}T = \tau \sin \theta \, . \tag{27}$$

The nominal "acceptance angle," that is, the total width 2θ of the azimuthal arrival stage between sounds emerging at low frequencies, from the left channel only and the right channel only is 180° for Eqs. (22) or (23) and the figures tabulated in Tables 1 and 2. We also consider the case with acceptance angles of 120 and 90°, for which we find, respectively.

$$F = \left(\frac{2}{3^{1/2}}\right)\left(\frac{c}{\pi d}\right) = \frac{125}{d} \tag{28}$$

and

$$F = 2^{1/2}\frac{c}{\pi d} = \frac{153}{d} \, . \tag{29}$$

Tables 3 and 4 show the separation in decibels and the phase difference in degrees for various azimuth angles and frequencies for the case of Eq. (28) with 120° low-frequency acceptance angle, Tables 5 and 6 show the

Table 1. Interchannel separation in dB for various azimuthal arrival angles and frequencies of Blumlein shuffled microphones with no baffles, as a function of frequency, for low-frequency acceptance angle of 180°.

Azimuthal Angle (deg)	Frequency										
	0	$\frac{1}{8}F$	$\frac{1}{4}F$	$8^{-1/2}F$	$\frac{1}{2}F$	$2^{-1/2}F$	F	$\sqrt{2}F$	$2F$	$\sqrt{8}F$	$4F$
0 or 180	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00
15 or 165	4.60	4.60	4.59	4.57	4.54	4.48	4.37	4.15	3.73	3.01	1.85
30 or 150	9.54	9.50	9.36	9.19	8.87	8.28	7.27	5.71	3.54	0.85	−1.58
45 or 135	15.31	15.00	14.18	13.28	11.88	9.92	7.44	4.48	1.08	−2.15	−1.27
60 or 120	22.88	20.85	17.56	15.21	12.52	9.57	6.36	2.81	−1.11	−2.98	1.31
75 or 105	35.22	23.98	18.18	15.16	12.08	8.88	5.47	1.67	−2.42	−2.25	2.14
90	∞	24.07	18.01	14.95	11.83	8.60	5.15	1.28	−2.82	−1.80	2.11

Table 2. Phase lead in degrees of left channel over right for microphone technique of Table 1.

Azimuthal Angle (deg)	Frequency										
	0	$\frac{1}{8}F$	$\frac{1}{4}F$	$8^{-1/2}F$	$\frac{1}{2}F$	$2^{-1/2}F$	F	$\sqrt{2}F$	$2F$	$\sqrt{8}F$	$4F$
0 or 180	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
15 or 165	0.0	4.0	7.9	11.2	15.9	22.4	31.6	44.5	62.4	87.1	120.9
30 or 150	0.0	9.5	19.0	26.6	37.2	51.3	69.7	93.2	123.1	164.0	−133.0
45 or 135	0.0	19.9	38.0	51.0	66.6	84.5	105.1	130.3	165.5	−135.0	−36.1
60 or 120	0.0	43.1	68.0	81.0	94.2	108.9	126.9	152.2	−165.1	−82.2	37.2
75 or 105	0.0	79.9	93.6	100.9	109.7	121.1	137.7	164.1	−145.7	−47.8	84.3
90	90.0	96.0	102.0	107.0	114.3	124.9	141.0	167.9	−138.7	−36.3	100.4

data for a 90° acceptance angle.

By comparison, Table 7 gives the separation in decibels for various azimuthal sound arrival angles for several common coincident-microphone techniques, namely, the Blumlein technique (90° angled figures of eight), the wide MS technique with M having a cardioid ($1 + \cos\theta$) polar diagram and S a $\sqrt{3}\sin\theta$ polar diagram (with an acceptance angle of 120°), the narrow MS technique with the M polar diagram $1 + \cos\theta$ and the S polar diagram $\sin\theta$ (which has 180° acceptance angle), and back-to-back cardioids (with M polar diagram 1 and S polar diagram $\sin\theta$). Also listed in Table 7 are the acceptance angles. Table 8 shows the total energy response of the two stereo channels as a function of direction for these coincident-microphone techniques, including above and below, normalized for 0 dB at the front, plus the computed directivity factor (see Julstrom [23]), that is, the ratio of frontal energy gain to average energy gain integrated over all directions in the sphere of directions. This measures the relative rejection of sounds arriving from a random direction. Also shown in Tables 7 and 8 is the 120° angled coincident cardioid

Table 3. Interchannel separation in dB for various azimuthal arrival angles and frequencies of Blumlein shuffled microphones with no baffles, as a function of frequency, for low-frequency acceptance angle of 120°.

Azimuthal Angle (deg)	Frequency										
	0	$\frac{1}{8}F$	$\frac{1}{4}F$	$8^{-1/2}F$	$\frac{1}{2}F$	$2^{-1/2}F$	F	$\sqrt{2}F$	$2F$	$\sqrt{8}F$	$4F$
0 or 180	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00
15 or 165	5.36	5.35	5.33	5.31	5.26	5.16	4.98	4.63	4.01	2.96	1.42
30 or 150	11.44	11.35	11.09	10.76	10.17	9.16	7.61	5.47	2.74	0.34	−2.13
45 or 135	19.91	18.86	16.70	14.85	12.52	9.80	6.76	3.36	−0.43	−2.97	0.53
60 or 120	∞	24.07	18.01	14.95	11.83	8.60	5.15	1.28	−2.82	−1.80	2.11
75 or 105	25.27	21.15	16.57	13.81	10.82	7.62	4.07	−0.05	−3.88	0.08	0.99
90	22.88	19.94	15.96	13.35	10.45	7.27	3.71	−0.51	−4.08	0.76	0.38

Table 4. Phase lead in degrees of left channel over right for microphone technique of Table 3.

Azimuthal Angle (deg)	Frequency										
	0	$\frac{1}{8}F$	$\frac{1}{4}F$	$8^{-1/2}F$	$\frac{1}{2}F$	$2^{-1/2}F$	F	$\sqrt{2}F$	$2F$	$\sqrt{8}F$	$4F$
0 or 180	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
15 or 165	0.0	4.7	9.4	13.3	18.8	26.5	37.3	52.2	72.8	100.8	139.2
30 or 150	0.0	12.4	24.4	34.0	46.7	63.1	83.4	108.2	140.0	−173.7	−97.3
45 or 135	0.0	32.8	56.7	70.8	85.7	101.8	120.7	145.9	−174.3	−99.2	14.3
60 or 120	90.0	96.0	102.0	107.0	114.3	124.9	141.0	167.9	−138.7	−36.3	100.4
75 or 105	180.0	134.9	124.5	123.9	127.1	135.4	150.9	−179.5	−112.7	1.5	152.8
90	180.0	142.1	130.0	128.3	130.7	138.4	153.9	−175.2	−103.0	14.3	169.9

Table 5. Interchannel separation in dB for various azimuthal arrival angles and frequencies of Blumlein shuffled microphones with no baffles, as a function of frequency, for low-frequency acceptance angle of 90°.

Azimuthal Angle (deg)	Frequency										
	0	$\frac{1}{8}F$	$\frac{1}{4}F$	$8^{-1/2}F$	$\frac{1}{2}F$	$2^{-1/2}F$	F	$\sqrt{2}F$	$2F$	$\sqrt{8}F$	$4F$
0 or 180	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00
15 or 165	6.67	6.66	6.62	6.57	6.47	6.28	5.92	5.27	4.18	2.54	0.43
30 or 150	15.31	15.00	14.18	13.28	11.88	9.92	7.44	4.48	1.08	−2.15	−1.27
45 or 135	∞	24.07	18.01	14.95	11.83	8.60	5.15	1.28	−2.82	−1.80	2.11
60 or 120	19.91	18.04	14.88	12.54	9.80	6.68	3.06	−1.32	−4.17	1.85	−0.75
75 or 105	16.21	15.15	12.94	11.03	8.55	5.52	1.79	−2.93	−3.19	3.00	−2.14
90	15.31	14.38	12.37	10.56	8.16	5.15	1.37	−3.46	−2.56	2.94	−2.06

Table 6. Phase lead in degrees of left channel over right for microphone technique of Table 5.

Azimuthal Angle (deg)	Frequency										
	0	$\frac{1}{8}F$	$\frac{1}{4}F$	$8^{-1/2}F$	$\frac{1}{2}F$	$2^{-1/2}F$	F	$\sqrt{2}F$	$2F$	$\sqrt{8}F$	$4F$
0 or 180	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
15 or 165	0.0	6.1	12.1	17.1	24.1	33.8	47.3	65.7	90.3	123.1	168.5
30 or 150	0.0	19.9	38.0	51.0	66.6	84.5	105.1	130.3	165.5	−135.0	−36.1
45 or 135	90.0	96.0	102.0	107.0	114.3	124.9	141.0	167.9	−138.7	36.3	100.4
60 or 120	180.0	150.6	137.9	135.0	136.2	143.2	158.9	−167.5	−85.1	37.4	−159.8
75 or 105	180.0	159.7	148.2	144.6	144.8	151.3	168.0	−150.9	−48.6	85.9	−95.8
90	180.0	161.6	150.7	147.1	147.2	153.6	170.9	−144.7	−36.7	102.4	−72.9

microphone technique. From Table 7 the width of the stereoism given by these microphone techniques can be determined (the "Blumlein" technique gives the widest stereo and the "narrow" MS technique the narrowest for frontal stage sounds), and their ability to reject sounds away from the front, both as a function of direction and for random arrival directions.

Tables 1–6 provide a comparison of the localization behavior of Blumlein shuffled microphones with no baffles, by comparing the technique with Fig. 7 for the familiar coincident-microphone techniques. It will be seen that the low-frequency localization of the shuffled microphones is comparable in width to the widest of the coincident-microphone techniques, giving a generally wider presentation of front-stage material, especially for the 120° and 90° acceptance angles. Table 8 may be compared with the calculations presented in the following section for the directional energy response for various Blumlein shuffled techniques.

2.3 Unbaffled Blumlein Shuffled Techniques

Table 9 shows similar information as Table 8, that is, the total energy gain as a function of direction, and the directivity factor computed for the low-frequency behavior of Blumlein shuffling applied to a spaced pair of identical forward-facing microphones with various polar diagrams, all gains being normalized to 0 dB at front center azimuth. We show the cases with 180° and 120° acceptance angles.

The microphone polar diagrams $g(\theta)$ that we consider here are

1) Omnidirectional, $g(\theta) = 1$
2) Cardioid, $g(\theta) = 1 + \cos\theta$
3) Figure-of-eight, $g(\theta) = \cos\theta$
4) 135°-null hypercardioids (sometimes termed "supercardioids"), $g(\theta) = 0.7071 + \cos\theta$
5) 120°-null hypercardioids, $g(\theta) = \frac{1}{2} + \cos\theta$
6) 107.46°-null hypercardioids, $g(\theta) = 0.3 + \cos\theta$.

In calculating the directivity factor one has to take into account the fact that the polar diagrams at low frequencies are *second-order* polar diagrams, involving second spherical harmonics of direction as in Eqs. (20), and the integration of the energy gain over the sphere of directions must take into account the way these polar diagrams vary with direction above and below the horizontal plane [2], [24]. We omit the mathematics here, just giving the results of calculations. Some general theoretical results on the directivity factor of higher order directional microphones are presented in [25] using rather abstract techniques.

Table 9 shows the energy gain in decibels at low frequencies of these microphones used with Blumlein shuffling in the difference channel for the two acceptance angles of 180° [Eq. (22) or (22)] and 120° [Eq. (28)] as a function of direction normalized to 0 dB for front sounds. It also lists the directivity factors of these techniques. Table 10 gives the same information for the same Blumlein shuffled microphones techniques at high frequencies, where the polar diagram in stereo is the same as that of the original mono microphones used, since shuffling has no effect at high frequencies.

It will be seen from Table 9 that if the microphones used are cardioids or hypercardioids facing forward, the Blumlein shuffled microphones give an excellent rejection of sounds to the sides and back of the microphone array. The use of hypercardioids with nulls 135° off-axis is giving over 15-dB reduction of pickup over the rear 120° of arc as compared to the front—which provides excellent rejection of, for example, audience noise from a broad rear sector. This rear-noise rejection is far better than with conventional stereo microphone techniques, and yet is not at the expense of a wide stereo image.

While the use of 135°-null hypercardioids gives the best rear-stage sound rejection, the highest stereo directivity factor is for the 107°-null hypercardioid in the case of a 180° acceptance angle. In this case the *stereo* directivity factor at low frequencies is an astonishingly high 3.25, which is actually higher than that for a mono cardioid. The use of a 120°-null hypercardioid, however, gives better rear-stage rejection and a directivity factor that still exceeds 3 for a 180° acceptance angle.

From Table 8 only the "narrow" MS technique has a directivity factor comparable to that of many of the

Table 7. Interchannel separation in dB for various azimuthal arrival angles for coincident-microphone techniques described in text.

Direction of Arrival, Azimuth (deg)	Microphone Technique				
	Blumlein	Wide MS	Narrow MS	180° Cardioids	120° Cardioids
0	0.00	0.00	0.00	0.00	0.00
15	4.77	4.03	2.30	4.60	2.65
30	11.44	8.73	4.77	9.54	5.42
45	∞	15.68	7.66	15.31	8.47
60	11.44*	∞	11.44	22.88	12.04
75	4.77*	17.00*	17.61	35.22	16.54
90	0.00*	11.44*	∞	∞	22.88
105	−4.77*	8.27*	17.61*	35.22	34.00
120	−11.44*	6.02*	11.44*	22.88	∞
135	−∞	4.24*	7.66*	15.31	31.35
150	−11.44	2.71*	4.77*	9.54	17.46
165	−4.77	1.32*	2.30*	4.60	8.06
180	0.00	0.00*	0.00*	0.00	0.00
Acceptance angle	90.00	120.00	180.00	180.00	240.00

* Antiphase.

Blumlein shuffled techniques, and this is at the expense of a very narrow reproduced stereo stage, unlike in the Blumlein shufffled techniques.

Also of interest is the shuffled "phased array" (forward figure-of-eight) case, which has a reasonable directivity factor while rejecting sounds side-on to the microphone array.

In general the use of Blumlein shuffling does not cause a large loss of directivity factor as compared to the microphones used for mono with the same polar diagrams, and also it degrades only slightly the rejection of side and rear sounds. By contrast, the MS technique often significantly degrades rear-stage rejection (due to rear pickup by the S signals) and the directivity factor.

2.4 Use with Gun Microphones

The use of Blumlein shuffling is particularly beneficial where a stereo gun microphone is required. Because gun microphones have polar diagrams that are very frequency dependent, crossed pairs of gun microphones give very poor stereo images, with the image width varying greatly with frequency in an irregular manner. While it has been proposed to use a gun microphone with a figure-of-eight S signal using the MS technique, the figure of eight has substantial pickup at the sides and rear, meaning that much of the directivity is lost, and the irregular polar diagram of the shotgun M signal still gives stereo imagery that is confused.

However, if two matched gun microphones, spaced apart about 200 mm, are pointing in the same direction, then it is possible to combine a good stereo image quality with a directivity factor virtually equal to that of the original gun microphone by using Blumlein shuffling.

Because the beam widths of the gun microphones are rather narrow, sounds more than, say, 45° off axis will be sharply attenuated, making the use of a narrow acceptance angle such as 90° or even as little as 60° acceptable by increasing the frequency constant F of the Blumlein shuffler. Thus sounds nearly on axis will be presented with a good stereo width as well as a fairly flat frequency response. Sounds further off axis will have a less good frequency response because of the typical performance of the polar diagram of a shotgun microphone, but its stereo image will still remain fairly sharp.

One bonus of the narrow acceptance angle of a shotgun is that the frequency range over which the amplitude stereophony is reasonable will generally be larger than for normal microphones, since the relative time differences at the two microphones are smaller, thanks to Eq. (16). In addition the mono sum of the shuffled microphone will be subject to comb-filter effects at a higher frequency than for ordinary microphone types, thanks to the smaller arrival angles and time differences, and may be acceptable in many applications provided that the source of prime interest is almost on axis with very little time delay between microphones.

Thus Blumlein shuffling is particularly appropriate for use with shotgun microphones, giving reasonable stereo at a greater distance than any other known stereo microphone technique. The disbenefit is the potential for poor mono compatibility, although care in use can minimize this problem as just noted.

A hand-held stereo shotgun microphone will tend to give a rather larger stereo image movement than existing stereo microphones if the microphone is "wobbled around." This is the audio equivalent of the fact that "camera shake" is more serious for telephoto lenses than for wide-angle lenses, and so care should be taken in pointing the microphone array accurately.

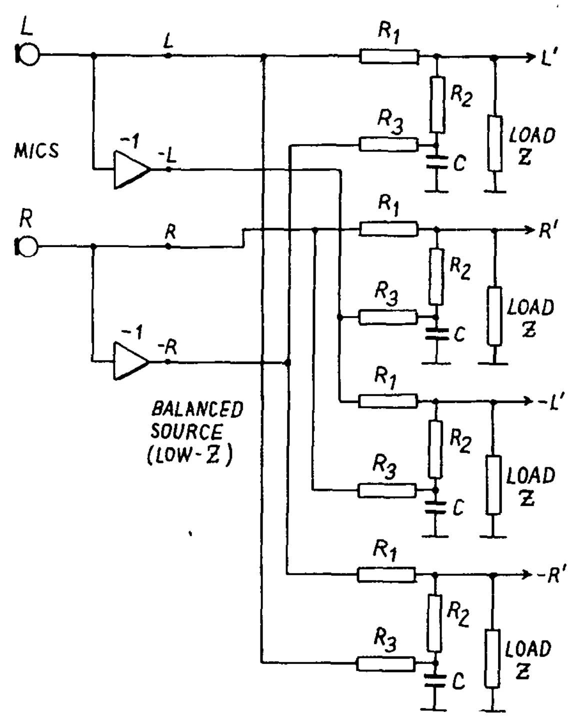

Fig. 7. Schematic of circuit for passive Blumlein shuffling from electronically balanced microphone amplifier output into unbalanced or balanced input with load impedance Z. − L′ and − R′ output circuits are not required for feeding an unbalanced input.

Table 8. Directional energy pickup pattern of stereo microphone techniques of Table 7, in dB relative to front.

Stereo Microphone Technique	Direction, Azimuth (deg) 0	45	60	90	120	135	180	Above/Below	Directivity Factor
Blumlein	0.00	0.00	0.00	0.00	0.00	0.00	0.00	−∞	1.500
Wide MS	0.00	0.43	0.51	0.00	−2.04	−4.02	−∞	−6.02	1.714
Narrow MS	0.00	−0.69	−1.25	−3.01	−6.02	−8.34	−∞	−6.02	2.400
180° cardioids	0.00	1.76	2.43	3.01	2.43	1.76	0.00	0.00	0.750
120° cardioids	0.00	−0.08	−0.25	−1.09	−3.01	−4.53	−9.54	−3.52	1.688

2.5 Calculating the Shelf Equalizer

In Fig. 6 we noted that the gain of bass energy of Blumlein shuffling should be compensated by a shelf-filter bass cut in the M signal path. The precise form of this filter depends on the microphone characteristics to which it is applied, although a bass cut of around 1 dB is a broad compromise that should be reasonably good for many cases. This filter is computed as follows.

If the shuffling filter is applied only to the difference signal path, then the gains of due front sounds, which have 0 S signal, are unchanged by the shuffling. The energy gain averaged over the sphere of directions is inversely proportional to the directivity factor (by the definition of the latter) if the forward gain is constant. Thus the low-frequency energy gain of the shelf filter required to flatten the average energy response is simply the high-frequency directivity factor (which can be read from Table 10) divided by the low-frequency directivity factor (which can be read from Table 9). The last column of Table 9 shows the low-frequency gain in decibels of the shelf-filter equalizer calculated in this manner.

It will be noted from the last column of Table 9 that wider acceptance angles have the effect of requiring a greater degree of bass cut, and the degree of bass cut required increases further for even smaller acceptance angles. For example, with shuffled omnidirectional microphones, for a 90° acceptance angle a shelf-filter cut of −2.22 dB is required, and for a 60° acceptance angle a cut of −3.68 dB is required. The cuts required for other polar diagrams are rather less than this for such small acceptance angles.

Provided that they give a flat perceived response to start with, baffled omnidirectional microphones, such as the Theile sphere microphone, need the same calculated bass shelf-filter cut in decibels as calculated for unbaffled omnidirectional microphones, although the actual frequency constants for surface microphones on a hard

Table 9. Directional energy pickup pattern of various Blumlein shuffled unbaffled microphone techniques, with acceptance angles indicated in parentheses, shown in dB relative to front at low frequencies.

Shuffled Stereo Microphone Technique	Direction, Azimuth (deg) 0	45	60	90	120	135	180	Above/ Below	Directivity Factor	Low-Frequency Shelf-Filter Gain (dB)
Omnidirectional (180°)	0.00	1.76	2.43	3.01	2.43	1.76	0.00	0.00	0.750	−1.25
Cardioid (180°)	0.00	0.39	−0.07	−3.01	−9.61	−14.93	−∞	−6.02	2.308	−1.14
Figure of eight (180°)	0.00	−1.25	−3.59	−∞	−3.59	−1.25	0.00	−∞	2.500	−0.79
135°-null hypercardioids (180°)	0.00	0.13	−0.58	−4.65	−15.89	−∞	−15.31	−7.66	2.732	−1.07
120°-null hypercardioids (180°)	0.00	−0.13	−1.09	−6.53	−∞	−15.44	−9.54	−9.54	3.068	−0.99
107.5°-null hypercardioids (180°)	0.00	−0.46	−1.79	−9.73	−13.82	−8.32	−5.38	−12.74	3.250	−0.89
Omnidirectional (120°)	0.00	2.22	3.01	3.68	3.01	2.22	0.00	0.00	0.692	−1.60
Cardioids (120°)	0.00	0.84	0.51	−2.34	−9.03	−14.47	−∞	−6.02	2.143	−1.46
Figure of eight (120°)	0.00	−0.58	−3.01	−∞	−3.01	−0.58	0.00	−∞	2.368	−1.03
135°-null hypercardioids (120°)	0.00	0.59	0.00	−3.98	−15.31	−∞	−15.31	−7.66	2.596	−1.29
120°-null hypercardioids (120°)	0.00	0.33	−0.51	−5.86	−∞	−14.98	−9.54	−9.54	2.872	−1.28

Table 10. Directional energy pickup pattern of mono microphones and Blumlein shuffled unbaffled microphones at high frequencies, shown in dB relative to front.

Mono Microphone Polar Diagram	Direction, Azimuth (deg) 0	45	60	90	120	135	180	Above/Below	Directivity Factor
Omnidirectional	0.00	0.00	0.00	0.00	0.00	0.00	0.00	0.00	1.000
Cardioid	0.00	−1.38	−2.50	−6.02	−12.04	−16.69	−∞	−6.02	3.000
Figures of eight	0.00	−3.01	−6.02	−∞	−6.02	−3.01	0.00	−∞	3.000
135°-null hypercardioid	0.00	−1.63	−3.01	−7.66	−18.32	−∞	−15.31	−7.66	3.497
120°-null hypercardioid	0.00	−1.89	−3.52	−9.54	−∞	−17.20	−9.54	−9.54	3.857
107.5°-null hypercardioid	0.00	−2.22	−4.22	−12.74	−16.26	−10.08	−5.38	−12.74	3.992

sphere will be about two-thirds of those computed for unbaffled omnidirectional microphones spaced the same distance apart, thanks to Eq. (26).

Conversely, it can be shown that for highly directional shotgun microphones, Blumlein shuffling has a relatively smaller effect on the directivity factor, so the amount of shelf-filter equalization in that case may be relatively small. In any case, equalization with such microphones is best determined by ear, since variations in polar response make the "best" equalization in any case more problematic.

2.6 Boundary-Layer Microphones

Blumlein shuffling may also be applied to ear-spaced boundary-layer microphones, which should point in the same direction if they are directional types, and which may be separated by a baffle. The properties and equalizations for such Blumlein shuffled boundary-layer microphones are exactly the same as for the same polar patterns in free space. The author has obtained excellent recordings with a pair of boundary-layer microphones spaced 180 mm side by side, taped to a floor in front of musicians. A similar technique is well suited to picking up stage performers in theatrical productions, either for recording or even for public address pickup in stereo. The polar characteristics used may be omnidirectional or forward-facing cardioids to minimize rear pickup.

The only difference in theoretical performance between Blumlein shuffled boundary-layer microphones and their free-space equivalents is that all directivity factors are doubled, since the energy integration is only over a half-sphere. This doubling of directivity factors is one reason why boundary-layer microphones are often found to give a very clear sound in difficult conditions.

3 PRACTICAL IMPLEMENTATIONS

3.1 When to Process?

Location recording with Blumlein shuffling techniques does not absolutely require the availability of a Blumlein shuffling network, since the raw microphone outputs can be recorded and processed off tape. However, this loses the advantage of being able to monitor the final effect on location. This option has become feasible with the advent of digital recording media. With analog recording media, small mismatches and noise problems tended to be magnified at low frequencies.

It is important that the two microphones have accurately matched gain and polar characteristics, to better than 1 dB, at lower frequencies so that the shuffling does not magnify these differences and produce a lopsided sound balance. One can accurately match the two microphones by placing them immediately next to one another and adjusting the gains of the microphones so that the difference of their two outputs is nulled, that is, is made as close to zero as possible. This use of subtraction to accurately match microphone gains is one of the most sensitive methods available and can be judged by ear. Preferably the difference signal should be monitored through a bass boost so that any potential problems with Blumlein shuffling can be identified when performing the gain matching.

Another advantage of postprocessing is that the original "mono" signals are available for other uses, and that a digital postprocessor can be used to optimize the phase response of the S signal path to maximize separation, which optimally uses phase-linear high-pass filtering as discussed earlier, which is most easily done in a digital processor.

Alternatively, a purpose-built microphone processor can be constructed following the block diagram of Fig. 5, using a 15-dB shelf boost in the difference S channel to perform the shuffling, and with a shelf filter (with central shelf frequency approximately equal to the frequency constant of the S signal filter) with a bass cut of between 0 and, say, 3 dB in the sum M signal path. Also an optional first-order all-pass filter may be placed in the M signal path to phase compensate for the very low-frequency "flattening off" at + 15 dB of the S signal equalizer, to improve the stereo separation. The degree of cut of the M signal bass equalizer and the frequency constant of the S signal shuffling equalizer should be calibrated so that both settings can be repeated and calculated values can be used according to the measured microphone separation, polar diagrams, and desired angular coverage, according to the equations and tables of this paper. Such a processor will incorporate two MS matrices as well as the adjustable equalizers.

3.2 Experimental Processing

Before committing to the design and construction of a dedicated Blumlein shuffling processor, the reader may wish to experiment. One way of doing this is to use two commercially available MS matrix boxes and to put graphic equalizers in the sum and difference signal paths. While such equalizers are not ideal for this purpose, since they are incapable of exactly duplicating the exact desired shuffling filters, if carefully adjusted they can give good results. Adjustment is best done so that the amplitude response of the equalizers is measured and made to conform to the ideal response to within a fraction of a decibel over most of the audio range. While the phase response will be slightly in error due to frequency response errors below the audio band, this seems to work well in practice, although being tedious to adjust.

For example, for a sphere microphone 200 mm in diameter with surface-mounted pressure microphones, and for an acceptance angle of 180°, the frequency constant F of the equalizer corresponds to a distance $d = 1\frac{1}{2} \times 200$ mm $= 300$ mm, by Eq. (26), giving a frequency constant of $F = 361$ Hz. The sum signal shelf filter should have a high-frequency gain of 0 dB and a low-frequency gain of -1.25 dB. From Table 9, row 1, last column, the gain should be about halfway, that is, -0.625 dB at the frequency $F = 361$ Hz. Since the S signal path should have a relative gain of $+3.01$ dB at this frequency, its actual gain will be $3.01 - 0.62 = 2.39$ dB. At half the frequency, where its relative gain should be $+6.99$ dB, its actual gain will be about 6.99

$-1 = 6$ dB, since the shelf filter in the M channel has a gain of about -1 dB at that frequency.

The result of equalizing a Theile sphere microphone by a graphic equalizer shuffling arrangement of this kind has no perceptible effect on the bass tonal quality, and it can be difficult to hear any change upon switching between unshuffled and shuffled signals. However, more extended listening reveals that bass sounds are more correctly aligned in direction with the treble, and the overall result has lower listening fatigue and is generally more natural. Moreover, the effect of this shuffling does not significantly alter the good sense of distance of this microphone technique, whereas more careless shuffling can be very damaging, particularly if the equalization is inaccurate or the shelf filtering in the sum channel is omitted.

The effect of using a larger acceptance angle (higher frequency constant F) is to create a more dramatic stereoism and sense of space, but it is more difficult to guarantee tonally neutral results, and care in setting up the bass equalization in the sum channel is vital.

3.3 A Simple Passive Shuffler

Another way of experimenting with shuffling is to place a passive processing network between a microphone preamplifier with an electronically balanced output and the input of a tape recorder or mixer. Since the Blumlein shuffler boosts certain signal components by up to 15 dB, such a passive network must introduce an (at least) 15-dB attenuation. This is often not a problem if, for example, a professional + 4-dB standard piece of equipment feeds a domestic -10-dB standard piece of equipment.

Although the passive shuffler is less easy to adjust, its advantage is that it only requires a few resistors and capacitors to implement, so it is ideal both for preliminary evaluations at low cost and also for a simple portable processing kit that can be taken on location for use with existing equipment.

The circuit involved, shown in Fig. 7, makes use of the fact that equipment such as microphone amplifiers having properly electronically balanced outputs provides both the signal and its polarity inverse with gain -1, which is shown schematically in the input stages of Fig. 7 as a gain of -1. (*Warning*: Many items of equipment for balanced use do not have such properly electronically balanced outputs.) These signals may be recovered from the pin 2 "hot" terminal and pin 3 "cold" terminal of *XLR*-type output connectors or from the tip and ring of balanced 1/4-in (6.3-mm) jack connectors by a simple wired lead.

These positive and negative signals are then cross-fed to the simple three-resistor one-capacitor networks shown in Fig. 7, which in turn feed the inputs of another piece of equipment with a load impedance, as shown schematically in Fig. 7 as Z. For unbalanced inputs, only the L_{out} and R_{out} parts of the circuit are used, but for electronically balanced inputs, a second pair of three-resistor one-capacitor networks are used as shown, fed symmetrically from the opposite-polarity terminals.

This network will not be analyzed in detail here since the analysis should not be too difficult for any competent circuit engineer. By considering the effect of the circuit first for L = R (that is, an M signal) by imagining the L and R outputs from the microphone amplifier connected together, it can be shown that the network has the effect of a simple first-order bass cut shelf-filter network, and by considering the effect for L = $-$R (that is, an S signal) by imagining the L and $-$R outputs from the microphone amplifier connected together, the circuit is then found to have the effect of a first-order shelf filter with gain 1 at low frequencies and a smaller gain at high frequencies equal to that of the M signal shelf.

By such an analysis it is also found that the relative equalization of the M and S signals is not affected by the resistive load impedance Z, which only has the effect of changing the overall equalization, which is a shelf bass cut. For load impedances Z greater than 40 kΩ the resistor values R_1 = 12 kΩ, R_2 = 3 kΩ, and R_3 = 15 kΩ are recommended, giving a high-frequency loss of 13.98 dB and a shuffling with a 13.98-dB bass boost. Table 11 shows the frequency constant F and the associated microphone spacing d for an 180° acceptance angle for various preferred values of capacitor C. In general C is proportional to $1/d$. The source-load impedance of the three-resistor-one-capacitor network is 2.4 kΩ at low frequencies and 7.2 kΩ at high frequencies. Of course, all resistor and impedance values can be scaled upward or downward as long as the capacitor value is varied in inverse proportion. Wider acceptance angles can be achieved by making the capacitor C smaller, by a factor 0.866 for a 120° acceptance angle, by 0.707 for a 90° acceptance angle, and by 0.500 for a 60° acceptance angle.

Alternatively, one can use the values R_1 = 10 kΩ + 810 Ω, R_2 = 2.7 kΩ, and R_3 = 12 kΩ + 1.5 kΩ, which also gives a 13.98-dB high-frequency loss and which should be used with the capacitor C values shown in Table 12 for various frequency constants F and microphone spacings d for a 180° acceptance angle. This passive circuit can be made adjustable for different spacings or acceptance angles by providing a switched range of capacitor values, which can be made fairly fine in adjustment through series-connected chains of capacitors, using make-before-break switching to avoid switch noise.

High-precision components should preferably be

Table 11. Values of frequency constant F and microphone spacing d (for 180° acceptance angle) for various values of capacitor C for R_1 = 12 kΩ, R_2 = 3 kΩ, and R_3 = 15 kΩ in Fig. 7.

C (nF)	F (Hz)	d (mm)
150	707.4	153.0
220	482.3	224.4
330	321.5	336.6
470	225.8	479.4
680	156.0	693.6
1000	106.1	1020.0

used, with resistor tolerances better than 2 or 1% and capacitors matched to better than 2%. This circuit does not provide any adjustment of the overall bass-cut shelf equalization, which is fixed at around − 1 dB, although reducing the load impedance Z, for example, by a parallel resistor increases the bass cut. Nevertheless, it is a reasonable compromise for initial experiments and straightforward location use. This circuit does not incorporate any all-pass phase compensation of the M signal for the curtailment of the S signal bass boost to + 13.98 dB. This kind of refinement requires the use of active circuitry.

Provided that the user has access to suitable electronically balanced output signals from the microphone amplifiers, this simple pasive network makes Blumlein shuffling available to a wide range of users without requiring dedicated equipment, and most studios and broadcasting organizations should have little difficulty in trying it out in practical and even location recording situations.

This circuit illustrates that shufflers do not need to use MS matrices, but can be implemented from input signals providing the stereo in both polarities via a symmetrical cross-feed arrangement. Many other such implementations are possible.

3.4 The Microphones

In a practical recording setup, mounting the two identical microphones is generally easy since they can be mounted side by side on any stereo bar permitting the required spacing of around 180 or 200 mm. By mounting the stereo bar on a "pistol grip," it is even practical to have a hand-mounted Blumlein shuffled array, which is particularly useful with cardioid, hypercardioid, or shotgun type microphones. In general the use of Blumlein shuffling will make the microphone system more sensitive to wind and handling noise (since second-order microphone characteristics have a high degree of proximity effect [2]), so that care should be taken in shielding against wind noise and in mounting arrangements when stand or handling transmitted noise is a problem.

As noted earlier, accurate microphone gain matching is essential. Preferably microphones with stable gain should be used in prematched pairs and operated from the same phantom power supply to avoid gain mismatches. Failing this, nulling of the difference signal can be used to preset the microphone gains as described earlier, but this only works if the microphone gains remain stable during the duration of use after gain matching. Care should also be taken to ensure that two microphones should have substantially identical polar diagrams (so that models with high variability are avoided, as should the use of microphones made before and after a significant production change). It should also go without saying that the directions of the microphones should be accurately aligned, particularly in the case of hypercardioid, figure-of-eight, and shotgun microphones.

Because of the high proximity effect of second-order microphone characteristics [2], in general, Blumlein shuffling used with directional microphones is not suitable for use very close to a sound source (unless additional shuffling is used which has been designed to compensate for the proximity effect—and calculating such shuffling is a specialist's task). In general, a distance of at least 2 m is advisable, except in the case of omnidirectional microphones, when a smaller distance may be usable.

Table 12. Values of frequency constant F and microphone spacing d (for 180° acceptance angle) for various values of capacitor C for R_1 = 10.81 kΩ, R_2 = 2.7 kΩ, and R_3 = 13.5 kΩ.

C (nF)	F (Hz)	d (mm)
150	786.0	137.7
220	535.9	202.0
330	357.3	302.9
470	250.8	431.5
680	173.4	624.2
1000	117.9	91.8

4 FUTURE DEVELOPMENTS

The most urgent needed development with Blumlein shuffling is the availability of a dedicated commercial processor with controls calibrated for use with arrays with known characteristics (microphone spacing d, acceptance angle, sphere diameter D, and polar diagram), possibly with a nomogram or internal calculator program to set up parameters according to these characteristics. Such a processor might also incorporate the subtraction nulling facilities required to ensure accurate microphone gain matching. Preferably such equipment would be battery operated so that it could be used for ENG (electronic news gathering) applications with stereo shotgun microphones.

Griesinger [14] has described a commercially available processor that incorporates Blumlein shuffling, but is intended primarily for home playback use and does not incorporate facilities (such as bass shelf-filter cut) that we believe are essential for optimum use.

For processing in the digital domain we believe that the incorporation of a phase linearizer for the high-pass filter required to limit the bass boost in the S channel is desirable, although there is inevitably a time delay (latency) through such equalizers which can create operational problems when mixing with other microphones. Therefore a simpler all-pass phase compensation should be available as an alternative when this is a problem. Digital processing also provides the opportunity to use a more refined shuffling filter than the RC type for the 6-dB per octave boost with 90° phase shift, possibly with a phase response closer to 90° at lower frequencies than the RC type. This would give less interchannel phase difference at the outputs and better channel separation at low frequencies.

We have not in this paper calculated the stereo performance of Blumlein shuffling for the Theile sphere microphone [7], although the theory for doing this is well known [20], [22], and we encourage others to do this and publish the results. The optimum shuffler characteristic for the sphere or dummy head may differ in detail from that ideal for unbaffled microphones, and further studies are needed to optimize results in various applications of Blumlein shuffling. Whether or not such optimization would yield significant improvements is still an open question. Certainly it is our experience that getting quite subtle details right does yield significantly better sounding results, and further refinements of an apparently minor nature, affecting frequency response by a fraction of a decibel and phase by only a few degrees, may prove to be important.

5 CONCLUSIONS

In this paper we have examined the use of the Blumlein shuffling technique, which converts intermicrophone time-delay differences into stereo amplitude differences, with pairs of identical microphones pointing in the same direction and spaced apart laterally by about 200 mm.

Not only does Blumlein shuffling produce an acceptable true stereo with reasonable imaging quality, it also allows stereo microphone techniques that combine wide stereo imaging with a high rejection of sounds from unwanted directions.

The technique can be used with almost any known microphone type, including boundary-layer and shotgun microphones, and in both applications it yields unique results unavailable with any other microphone technique, such as a true shotgun stereo microphone with very high directivity.

This paper has described the principles and also emphasized some of the more subtle adjustments, such as bass equalization of the sum channel by around 1 dB and phase compensation at very low frequencies, which can improve the subjective naturalness of the technique.

A detailed analysis of the microphone performance of Blumlein shuffling has been given both in terms of mathematical expressions and in the form of numerous numerical tables that should help in the selection of the best technique for many applications.

In the absence of dedicated processors for Blumlein shuffling, a simple practical passive circuit has been given that can be used at least for experimental recordings, and even for regular use once evaluated.

Blumlein shuffling can also be used to improve the loudspeaker results from both dummy-head recordings and other head-size baffled microphone techniques such as the Theile sphere microphone, yielding a more spacious and more convincing localization quality in the bass. However, it has been found that accurate equalization is required here for best results.

The technique has been found to have two main problems—poor mono compatibility and unsuitability for use close to a source, and it is also more liable to wind and handling noise than many other techniques. However, none of these problems lead to a lack of usability, and with care even mono compatibility problems can be minimized.

In conclusion, the Blumlein shuffling method is commended to recording engineers as allowing good stereo results in a wide range of situations where existing stereo techniques cannot cope well, such as cases of high ambient noise or use at a long distance from the sound source, and also for applications such as recording and public address of theatrical productions. Blumlein shuffling is of particular use in providing a wide stereo stage from a relatively narrow range of incident directions. Arguably the oldest stereo microphone technique, perhaps the time has come for it to become a part of the armory of techniques used in commercial recording and broadcast work, where its unique virtues extend the range of situations that can be handled by the recording engineer.

6 ACKNOWLEDGMENT

Thanks are due to two persons, Jerry Bruck with whose help I was able to try shuffling with the Schoepps version of the Theile sphere microphone, and Sean Davies, who made me aware of Blumlein's 1934 unpublished work on Blumlein shuffling with directional microphones.[1] I also appreciate a continuing correspondence from Duane Cooper since the early 1970s, where many aspects of the acoustics of the head and solid-sphere baffles were discussed in relation to loudspeaker reproduction.

7 REFERENCES

[1] A. D. Blumlein, British patent 394, 325 (1931 Dec. 14) reprinted in *J. Audio Eng. Soc.*, vol. 6, pp. 91–98 and 130 (1958 Apr.); and in J. Eargle, Ed., *Stereophonic Techniques* (Audio Engineering Society, New York, 1986), pp. 32–40.

[2] M. Gerzon, "Ultra-Directional Microphones," *Studio Sound*, vol. 12, pp. 434–437 (1970 Oct.); pp. 501–504 (1970 Nov.); pp. 539–543 (1970 Dec.).

[3] G. Bore and S. F. Temmer, "'M–S' Stereophony and Compatibility," *Audio Mag.*, p. 19 (1958 Apr.); reprinted in J. Eargle, Ed., *Stereophonic Techniques* (Audio Engineering Society, New York, 1986), pp. 202–205.

[4] W. L. Dooley and R. D. Streicher, "M–S Stereo: A Powerful Technique for Working in Stereo," *J. Audio Eng. Soc.*, vol. 30, pp. 707–718 (1982 Oct.).

[5] M. A. Gerzon, "The Design of Precisely Coincident Microphone Arrays for Stereo and Surround Sound," Proceedings of the 50th Convention of the Audio Engineering Society, *J. Audio Eng. Soc. (Abstracts)*, vol. 23, pp. 402, 404 (1975 June).

[1] I am grateful to S. W. Davies, who has seen notes [26] in the course of preparing a history of professional recording in Britain for informing me of the existence and nature of contents of these notes, which compute the directional pattern produced by Blumlein shuffling of omnidirectional and figure-of-eight microphones, well before similar calculations by the author in 1970 [2].

[6] K. Farrar, "Soundfield Microphone," *Wireless World*, vol. 85, pp. 48–50 (1979 Oct.); pp. 99–103 (1979 Nov.).

[7] G. Theile, "On the Naturalness of Two-Channel Stereo Sound," in *Television Sound Today and Tommorrow, Proc. AES 9th Int. Conf.* (Detroit, MI, 1991 Feb. 1–2), pp. 143–149 and *J. Audio Eng. Soc.*, vol. 39, pp. 761–767 (1991 Oct.).

[8] H. A. M. Clark, G. F. Dutton, and P. B. Vanderlyn, "The 'Stereosonic' Recording and Reproducing System" *J. Audio Eng. Soc.*, vol. 6, pp. 102–117 (1958 Apr.); reprinted in J. Eargle, Ed., *Stereophonic Techniques* (Audio Engineering Society, New York, 1986), pp. 81–96.

[9] D. Griesinger, "Spaciousness and Localization in Listening Rooms and Their Effects on the Recording Technique," *J. Audio Eng. Soc.*, vol. 34, pp. 255–268 (1986 Apr.).

[10] M. Gerzon, "Stereo Shuffling. New Approach—Old Technique," *Studio Sound*, vol. 28, pp. 122–130 (1986 July).

[11] D. H. Cooper and J. L. Bauck, "Prospects for Transaural Stereo," *J. Audio Eng. Soc.*, vol. 37, pp. 3–19 (1989 Jan./Feb.).

[12] D. Griesinger, "Binaural Techniques for Music Reproduction," in *The Sound of Audio, Proc. AES 8th Int. Conf.* (Washington, DC, 1990 May 3–6), pp. 197–207.

[13] S. P. Lipshitz, D. Griesinger, and M. A. Gerzon, "Comments on 'Spaciousness and Localization in Listening Rooms and Their Effects on the Recording Technique," and Author's Reply, *J. Audio Eng. Soc. (Letters to the Editor)*, vol. 35, pp. 1013–1014 (1987 Dec.).

[14] D. Griesinger, "Equalization and Spatial Equalization of Dummy-Head Recordings for Loudspeaker Reproduction," *J. Audio Eng. Soc.*, vol. 37, pp. 20–29 (1989 Jan./Feb.).

[15] M. A. Gerzon, "More Comments on 'A Dynamic Noise Reducer for Sum-Difference Multiplex Systems,'" *J. Audio Eng. Soc. (Letter to the Editor)*, vol. 26, pp. 229–230 (1978 Apr.).

[16] D. S. McCoy, "Distortion of Auditory Perspective Produced by Interchannel Mixing at High and Low Audio Frequencies," *J. Audio Eng. Soc.*, vol. 9, pp. 13–17 (1961 Jan.); reprinted in J. Eargle, Ed., *Stereophonic Techniques* (Audio Engineering Society, New York, 1986), pp. 130–135.

[17] P. G. Craven and M. A. Gerzon, "Practical Adaptive Room and Loudspeaker Equalizer for Hi-Fi Use," presented at the 92nd Convention of the Audio Engineering Society, *J. Audio Eng. Soc. (Abstracts)*, vol. 40, p. 436 (1992 May), preprint 3346.

[18] B. Bartlett and M. Billingsley, "An Improved Stereo Microphone Array Using Boundary Technology: Theoretical Aspects," *J. Audio Eng. Soc. (Engineering Reports)*, vol. 38, pp. 543–552 (1990 July/Aug.).

[19] M. Billingsley and B. Bartlett, "Practical Field Recording Applications: An Improved Stereo Microphone Array Using Boundary Technology," *J. Audio Eng. Soc. (Engineering Reports)*, vol. 38, pp. 553–565 (1990 July/Aug.).

[20] Lord Rayleigh, *Theory of Sound*, vol. 2 (Dover, New York, 1945) Section 334, pp. 272–282.

[21] D. H. Cooper, "Problems with Shadowless Stereo Theory: Asymptotic Spectral Status," *J. Audio Eng. Soc.*, vol. 35, pp. 629–642 (1987 Sept.).

[22] D. H. Cooper and J. L. Bauck, "On Acoustical Specification of the Natural Stereo Imaging," presented at the 65th and 66th Conventions of the Audio Engineering Society, *J. Audio Eng. Soc. (Abstracts)*, vol. 28, pp. 368, 370 (1980 May), preprint 1616; p. 552 (1980 July/Aug.), preprint 1649.

[23] S. Julstrom, "An Intuitive View of Coincident Stereo Microphones," *J. Audio Eng. Soc.*, vol. 39, pp. 632–649 (1991 Sept.).

[24] M. A. Gerzon, "Periphony: With-Height Sound Reproduction," *J. Audio Eng. Soc.*, vol. 21, pp. 2–10 (1973 Jan./Feb.).

[25] M. A. Gerzon, "Maximum Directivity Factors of *n*th Order Transducers," *J. Acoust. Soc. Am.*, vol. 60, pp. 278–280 (1976 July).

[26] A. D. Blumlein, unpublished notes in EMI archives (1934 Nov. 1 and 4).

THE AUTHOR

Michael A. Gerzon was born in Birmingham, England, in 1945 and received an M.A. degree in mathematics at Oxford University in 1967, where he did postgraduate work in axiomatic quantum theory.

His interests in audio stemmed from interests in music, sensory perception, and information theory. Since

1967 he has been active in sound recording live music, recording artists as diverse as Emma Kirkby, Michael Tippett, Pere Ubu, and Anthony Braxton, and has recorded music for around 15 LP and CD releases.

Arising from this interest, since 1971, he has earned his living from consultancy work in audio and signal processing. He was one of the main inventors of the Ambisonic surround sound technology, working in the 1970s and early 1980s with the British National Research Development Corporation.

With Dr. Peter Craven, he co-invented the Sound Field microphone. He also developed mathematical models for human directional psychoacoustics for use in the design of directional sound reproduction systems, and was made a fellow of the Audio Engineering Society in 1978 for this work. He was awarded the AES gold medal in 1991 for his work on Ambisonics. With Dr. Craven, he also developed the basis of the noise-shaped dither technologies now widely used for resolution enhancement of CDs, and continues to work actively in this area.

He has published over 90 articles and papers in the field of audio and signal processing, including numerous papers on stereo and surround sound systems. He has so far been granted 9 British patents and corresponding applications internationally. He has published papers on practical and theoretical aspects of linear and nonlinear signal processing and systems theory, digital reverberation, room equalization, data compression, spectral analysis, and noise-shaping and dither technologies.

He is currently working on digital signal processing algorithms for professional digital audio with ks Waves, on multiloudspeaker directional reproduction technologies with Trifield Productions Ltd., and on dither and buried data technologies for XtraBits.

He has many current research interests, including the development of general methods for designing complex signal processing systems based on the methods of *-algebras and category theory, image data compression, and advanced mathematical modeling of auditory perceptual effects relevant to audiophile quality.

Other interests include writing poetry, the history of contemporary musics, and the mathematical and conceptual foundations of theoretical physics.