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Stereo Shuffling:
New Approach – Old Technique

Michael Gerzon introduces an approach for experimentation

Although many recording engineers and studios don't realise it they already have the equipment to produce a marked improvement in the stereo quality of many of their recordings. Not digital effects using reverberation, delayed echoes or the like but a technique that has been known but almost unused for over 30 years. This is the stereo 'shuffler'.

What is a shuffler, and how come you didn't know you had one? Last question first – a shuffler can be produced by unconventional connections between the inputs and outputs of many mixers (my initial experiments were with an £80 mixer (!) but it should work at any price level) along with a stereo graphic equaliser. But to get the best out of this, and in the absence of a dedicated commercial shuffler, it is important to understand what you are doing.

The basic idea of the shuffler goes back to Alan Blumlein's invention of modern stereo in 1931[1]. (His British Patent 394,325 repays detailed study as perhaps still the best source text on how stereo works.) Blumlein conceived stereo not just as a left (L) and right (R) speaker signal but also in terms of a sum signal M (= L+R) and a difference signal S (=L–R). The letters M and S stand for 'mid' and 'side' signals (as in the M-S microphone technique): M is the signal containing information about the middle of the stereo stage, whereas S only contains information about the sides – since S=0 for a central signal.

Given M and S, the original left and right signals can be recovered by a second sum-and-difference operation, via 2L=M+S and 2R=M–S. By thinking in terms of the sum and difference signals, Blumlein was not merely able to devise the MS microphone technique (which was rediscovered and named by Laurisden in Denmark in the 1950s) but was able to modify the stereo effect of other recordings. In particular, Blumlein was able to modify the width of the stereo images of coincident-microphone recordings by increasing (or decreasing) the gain of the S signal relative to M before recovering the left and right signals (Fig 1). An increase in the relative gain of S increased width, whereas a decrease of S gain decreased width. In view of the fact that width control was known in 1931, it is strange that it is still not available on most modern stereo equipment.

One of Blumlein's many discoveries' was that increased width could yield stereo images beyond the left and right speakers. This useful discovery would permit one to pan sounds over a wider stage than normally used in today's studio. There is no reason why panpots should not be designed to cover such an increased stage width – yet I am unaware of a single mixer in which this is actually done.

This is not to say that width control is without problems – which we shall discuss in more detail further on – however, these problems can often be solved by a more sophisticated process called 'shuffling', also based on Blumlein's work. Blumlein noted that one could not merely alter the gain of the difference signal S, but one could alter this gain in a frequency-dependent way by using an equaliser. By this means he showed how one could improve the directional quality of particular stereo microphone techniques (including one pseudo-dummy-head technique rediscovered at the BBC a few years ago). The process of equalising the difference and sum signals differently before recovering left and right is termed 'shuffling'. In effect, shuffling is a frequency dependent width control.

The first systematic commercial use of shuffling was in EMI's *Stereosonic* system in the mid 1950s[2], in which the bass width of recordings made with coincident crossed figure-of-eight microphone pairs was increased relative to the treble width. The reason why EMI used shuffling was that research had revealed that stereo images at bass frequencies reproduced more narrowly than at treble frequencies for a given intensity ratio in the two speakers, and the increased bass width attempted to compensate for this. This didn't work adequately with the actual recording techniques EMI used at that time, so they dropped it.

From time to time, shuffling (or processes achieving identical results to shuffling) has been revived – for example, various domestic hi-fi products under the Realistic name have shuffler stereo 'enhancement' circuits built in – but in my opinion these are poorly implemented, giving an exaggerated bass-heavy quality. Various American authors have revived or rediscovered shuffling in recent years, notably Richard Kaufman[3] and David Griesinger[4].

Kaufman has proposed using the system illustrated in Fig 2 for shuffling. This system requires the construction of special sum-and-differencing circuits. Although such circuits are quite simple, they have to be constructed by the user, since they are not available as standard products (except for special purpose MS microphone processors). Nevertheless, Fig 2 is a useful way of understanding shuffling, and we shall use this for our basic descriptions of what it does. Other methods of achieving the same result are often easier to implement and use but their theory is harder to understand.

Essentially, in order to widen the stereo image at a given frequency, one increases the gain at that frequency in channel 2 of the graphic equaliser, possibly slightly decreasing the gain at that frequency in channel 1 in order to retain a flat frequency balance in the resulting overall sound. Similarly, to decrease the width at a given frequency, one reduces that frequency's gain in channel 2, increasing it slightly in channel 1 to maintain the overall frequency balance.

To change the overall width, one similarly increases or decreases the overall gain in channel 2 of the graphic equaliser of Fig 2. Moreover, the processing system of Fig 2 is a powerful technique of reducing various problems with stereo. For example, reducing the width at low frequencies makes vinyl records easier to cut, since low frequency S signals at a high level are hard to cut. Noise from FM stereo reception can be reduced by cutting the width around 7kHz, since S channel noise around this frequency contributes most to the perceived noise. Stereo mics picking up thumps from transmitted floor vibrations can often yield more thump-free recordings by selective bass filtering of the S channel.

Besides such problem-reducing roles, shuffling also has uses in enhancing overall stereo quality. One use, with coincident microphone recordings, is to use shuffling to render imperfect frequency-dependent images, due to microphone imperfections, more sharp by compensating for the width variations in the image. This can be particularly useful in the extreme bass, where conventional microphones such as cardioids tend to become more omnidirectional at the very lowest frequencies. An enhancement of extreme-bass width (typically below 100Hz) can sometimes compensate for this. In a similar way, providing that the capsules are sufficiently coincident, treble irregularities of microphone polar diagrams can be partially compensated for. Also, as described further below, some of the phase anomalies caused by small spacings of a few cm between microphones can also be partially compensated by suitable shuffling.

It has been suggested by Griesinger[4] that the sense of spaciousness of recordings can be improved by increasing the bass width below about 600Hz. He suggests that, if the S gain is increased relative to that of M at low frequencies by from 4 to 8dB, this is effective in creating an impressive reconstruction of the sense of space of the recording location when used with coincident microphone techniques. My own experiments using coincident and near-coincident (spacings of about 5 to 7cm) microphones yield the same frequency and S channel boost for best results in many cases, however, unlike Griesinger, I believe that this enhancement should be tried on a case-by-case basis, rather than as a blanket processing technique. This is because the technique is not uniformly effective with all locations and microphone techniques, and also there are some unwanted side effects on stereo image quality whose seriousness varies considerably between recordings.

In doing any shuffler processing, it is important to monitor over loudspeaker systems optimised for good stereo imaging – this unfortunately excludes most studio monitoring systems. Something like the classic miniature BBC *LS3/5A* speaker is still hard to beat for accurate monitoring of stereo images – both Griesinger and myself seem to have settled on it as a reference for work in this area, although it is worth trying results with other speaker types. If one doesn't have precision monitoring of stereo imaging (*eg* if one uses speakers optimised for spaciousness of reproduction rather than precise images), then one doesn't know if the shuffling is producing an authentic improvement in a recording or just compensating for the anomalies of the specific monitoring system used. For location recording work, it is advisable to make recordings without shuffler processing, postponing shuffling to post-tape processing in a familiar accurate monitoring environment.

An understanding of some of the problems that arise in the use of width and shuffler systems will speed up the process of finding optimum shuffler and width settings in a given case.

Normal panpot stereo and also stereo from truly coincident perfectly cardioid microphones, produce stereo images by placing individual sounds in the two speaker channels in identical phase but with differing amplitudes. If sounds are in opposite phase (*ie* with one channel phase-inverted relative to the other) on the two stereo channels with one louder than the other, then the sound image tends to have a more diffuse quality but can often be located *beyond* that stereo loudspeaker which is louder. While it is wonderful to have stereo images located beyond the loudspeakers, such images can cause problems.

Firstly, if the 'beyond the speaker' images are bassy sounds, then they can cause vinyl records to be hard to cut at high levels due to the large S signal produced. Secondly, because such beyond-the-speakers images have larger S than M signals, such sounds can be drastically reduced in level when reduced to mono, giving an unbalanced mono mix. For cassette, compact disc and video media, however, these problems may not be too important. Thirdly, the beyond-the-speakers images do have an unconvincing localisation quality. This often results in unstable, fuzzy or confusing images, although at other times the images can be quite dramatic in effect.

If beyond-the-speaker images are produced by increasing the width of an already-made recording or submix, then additional problems can arise if this recording does not consist of amplitude-difference stereo images to start with. The four cases when other types of stereo images with phase as well as amplitude differences can occur are:

- recordings made with microphones slightly spaced (by a few centimetres) from one another;
- recordings made with coincident hypercardioid, figure-of-eight or MS techniques, in which antiphase images can occur;
- when stereo studio effects involving phasing of channels are used, *eg* autopanning systems, some stereo reverb devices and some stereo synthesiser outputs;
- recordings with time differences in the two stereo channels due to analogue tape azimuth error or digital converter timing differences.

Consider by way of example, a recording made with ORTF technique, in which two cardioid microphones, angled about 110° apart, are spaced apart by ear distance spacing – about 17cm – as illustrated in Fig 3. A sound arriving from due left will arrive at the left microphone about 0.5ms before it arrives at the right microphone (since sound travels in air at about 340 m/s) and will be picked up about 20dB down on the right channel compared to the left (because of the cardioid directionality patterns of the microphones). At low frequencies, the phase difference of the sound at the two channels is small but at 1kHz, the sound has to travel half a wavelength between left and right microphones. As a result, at low frequencies (and also at 2kHz, 4kHz, 6kHz, etc) the sound arrives in phase at the two microphones, and a width control increase will indeed widen the stage, reducing crosstalk below 20dB for moderate width increases. However, at 1kHz (and also at 3kHz, 5kHz, 7kHz, etc) the sound arrives at the right microphone out of phase compared to the left microphone, so that a width control increase will actually reduce the crosstalk below 20dB at those frequencies.

As a result, any attempt to use width increase with ORTF technique will indeed widen lower frequencies but it will have the effect of alternately narrowing and widening higher frequencies, resulting in a possibly confused and degraded stereo imaging. Thus, with any spacing of microphones, width control should be confined to those lower frequencies at which the sounds from all directions arrive at both microphones more-or-less in phase. In practice, this would mean confining width control to frequencies below about $^{5.4}/_{d}$ kHz where d is the distance between the microphones in cm, or $^{2.1}/_{d}$ kHz where d is the distance between the microphones in inches. An exception to this rule is when the microphone spacing is so large (*eg* more than 2m) that the sound arrivals at the microphones are effectively incoherent, and the microphone signals can be treated as effectively independent signals.

It has to be said that on many recordings where width increase over a wide frequency band 'shouldn't' work, it does seem to give an enhanced wider image. This seems to be unpredictable, so should be tested on a case-by-case basis. The effect of other sources of interchannel phase differences, on width control, such as tape azimuth errors, can be similar to the effects described above with spaced microphones.

We have just seen a reason with particular kinds of microphone techniques, and with interchannel tape azimuth errors, why shuffling should be confined to bass frequencies (*eg* below about 320Hz for ORTF technique). Experimenting with width control and shuffling yields other good reasons why width control should be confined to the lower frequencies. Given that beyond-the-speaker images have anomalous localisation low frequencies do seem to localise quite reliably beyond the loudspeakers. So, if we increase the width only of lower frequencies (say below 600Hz or so), those low frequency signal components that do localise well are widened, while those at higher frequencies are retained in their usual easy-to-localise positions. Here the different frequency components of sounds are no longer in precisely the same position.

We shall, for convenience, adopt David Griesinger's term 'spatial equalisation' to describe width increase below about 600Hz. Actually, the effect of spatial equalisation isn't as simple as just described. First, as observed in the fifties by Clark, Dutton and Vanderlyn[2], and as rediscovered by Griesinger, many stereo reproduction systems give narrower reproduction of bass than treble, so that spatial equalisation might be expected to give sharper images in those cases. This, however, depends on the assumption that wideband sound localisation is simply the sum of the separate narrowband effects, something that is not confirmed by all research (*eg* that of Bower[5,6]). This suggests that the ears actually do some sort of cross-referencing of localisation data at different frequencies to produce the perceived imaging effect.

Listening shows that often, with a single sound that has lowish-frequency fundamentals and high frequency overtones, the subjective effect of increased bass width is often (but not always!) that the high frequency overtones are also pulled out to or near to the position of the fundamental frequencies. In such cases, the images can remain sharp. Unfortunately, high frequency sounds without low frequency components (*eg* cymbals) are not shifted. This can have strange effects on the stereo image – for example, on one recording with a drum kit spread across the right half of the stereo stage, the kick drum is moved over from the left side of the drum kit to the right side by spatial equalisation! Similar anomalies can change the distribution of instruments across the stage in an orchestral recording.

One thing that spatial equalisation (bass width increase below 600Hz) undoubtedly does is give all the sense of increased spaciousness that normal width increase can give but without the gross anomalies that the latter can give at higher frequencies. It appears[4] that the sense of spaciousness in stereo recordings largely depends on the directional handling of bass frequencies. This is very evident on, say, stereo recordings of audience applause in a live acoustic. The use of spatial equalisation can give a sense of being enveloped in the audience, even though there is no obvious change in the high-frequency imaging. The lower frequency components of the clapping seem to be enough to create a sense of being almost *there* among the audience.

On many recordings, spatial equalisation does not only improve the sense of spaciousness but can also improve the stereo imaging heard by listeners sitting away from the stereo seat.

In general when reprocessing recordings using spatial equalisation, one should avoid aiming at a grossly spectacular effect as this will probably prove to be unnatural and tiring on casual or repeated listening. One should listen carefully to what happens to the positioning of different instruments to make sure that the altered positioning is acceptable – sometimes it won't be. In preparing panpot recordings with the intention of using spatial equalisation, the original mix should either be monitored (under appropriate conditions) through the spatial equalisation, or if not, instruments with lowish or mid-frequency fundamentals should be panned rather more narrowly in the stereo image than is finally intended. One can, of course, experiment with using bass widening only on parts of the mix – *eg* on stereo reverb and on sounds intended to be placed beyond the speakers only.

Spatial equalisation seems most beneficial on stereo recordings on which the sense of space is inadequate. On recording techniques capturing a good sense of space (*eg* spaced omnis and – in good acoustics – Blumlein crossed figure-of-eights) the processing is often superfluous, and can sometimes even give exaggerated spaciousness. If one uses crossed cardioids, one cannot normally capture a good sense of space, especially as this technique seems to lend itself best to relatively close placement to the musicians. Spatial equalisation seems often to give excellent results with crossed cardioids, often giving a good sense of the acoustics of the recording venue.

With live recordings made with a crossed cardioid stereo pair, it is actually not true that precise coincidence of the two microphones gives best results (contrary to the case for many other directional characteristics). A degree of spacing can actually improve the stereo image quality. What one should certainly not do is space the microphones as in Fig 4 – this might liven up the sound but it degrades stereo imaging. Fig 5 indicates the optimum kind of spacing – this is similar to the 'crossed-over' ORTF technique, except that the optimum spacing for imaging is only about 5cm (2in). This spacing, widely used in cheap Japanese stereo electrets, was first commended to me by Tony Faulkner.

Remarkably, for normal stereo listening configurations, it turns out that the 5cm spacing produces roughly the same phase/amplitude relationships between the two ears of a listener in the stereo seat as does a live sound from the same apparent direction up to about 2kHz – and in this respect is better than true coincidence. Such 5cm-spaced crossed-over cardioids, angled about 115° to 120°

apart, seem to be an optimal cardioid technique for stereo imaging accuracy. The use of bass-widening up to 600Hz with this technique seems to give a much better sense of space than the use of ORTF technique, and without the latter's 'phasiness' anomalies.

There is another reason why spatial equalisation matches this cardioid recording technique particularly well – the existence of phase shifts between sum and difference channels in the shuffler circuits described in this article. Although not mentioned so far for simplicity, such phase shifts occur because equalisers not only alter the amplitude of signals but also their phase. This generally degrades the localisation of amplitude stereo and is usually a defect. It is possible (as realised by Vanderlyn[2] as early as 1957) to 'phase compensate' the equalisers to match the phase in the M and the S channels but this generally involves more complex circuitry, so will not be discussed further here.

Griesinger's results that bass widening sounds best if concentrated below 600Hz (which my own tests confirm) might partly be a side-effect of the lack of phase compensation – since a bass boost of S relative to M of 8dB produces a phase lag of about 25° in the S signal centred around 600Hz – at a frequency at which the ears are particularly sensitive to such 'phasiness'. To make things even worse, the ears are more sensitive to phase lags in the S channel than corresponding phase leads, as can be demonstrated from BBC psychoacoustic data[5,6].

In the absence of proper phase compensation of the M and S signals, one normally has to tolerate some phasiness and blurring at mid frequencies if using shuffling or spatial equalisation. It may well be (I haven't tried it yet) that spatial equalisation might work to frequencies significantly higher than 600Hz if proper phase compensation is used. However, it is interesting to note that small microphone spacings of the type shown in Fig 5 produce phase *leads* in the S signal relative to M at frequencies around the crucial 600Hz region. As a result, spatial equalisation without phase compensation can actually improve the 'phasiness' for crossed-over cardioid stereo, especially for sounds fairly close to the centre of the stereo stage and for microphone spacings of 5 to 10cm. Simple spatial equalisation has defects that undo the defects of these microphone techniques, giving a happy 'synergy'. The converse is that the defects of the microphone technique of Fig 4 are made even worse by spatial equalisation without phase compensation!

One can, in other situations, reduce the phase errors in S relative to M by arranging that the transition between low and high frequency gains is as slow as possible, rather than changing rather sharply around 600 Hz. I have found that images *do* sound sharper and less phasey if one sets the graphic equalisers so that the transition takes place over a few adjacent bands, and this is something that can be adjusted by ear for best effect.

Spatial equalisation is particularly suitable for reprocessing recordings for professional use made on amateur equipment. This is because, as already noted, many cheap stereo electrets have a spacing matched to the properties of spatial equalisation, and also because tape azimuth errors in cassettes or poorly maintained analogue reel-to-reel machines are not worsened by the processing. Additionally, unlike simple width enhancement, spatial equalisation does not worsen audible noise, since it leaves frequencies above about 1kHz unaltered.

Another situation where spatial equalisation can prove effective is with pseudo stereo derived by the Orban stereo synthesiser device from a mono original. Spatial equalisation can enhance the spaciousness without exaggerating the artificialities of the pseudo-stereo technique. For example, I have found it to work well applied to the Orban pseudo-stereo of Robert Parker's well-known reprocessing of old jazz recordings.

Having indicated many of the possibilities and limitations of shuffling (and there is of course much more that could be said at a technical level), we describe the practical implementation promised earlier using readily available equipment and with easy adjustment. Ideally, for the implementation to be described, one should use a stereo graphic equaliser in which the two channels are ganged. Failing that, two separately adjusted stereo channels can be used, although this means one has to adjust more controls. (I use a 7-band ganged graphic for this application.)

The method to be described has the unusual feature of automatically decreasing the sum channel gain as the difference channel gain is increased, so as to maintain an even overall frequency balance however the controls are set. Its description is so late in the article because it is not as easy to understand how it works as Fig 2, and it would have been confusing to introduce that complication earlier. However, once set up, it is very easy to use.

The basic idea is to use a mixer having a phase-inverting stereo signal path, which we shall term the 'processing signal path'. In many mixers, this can be provided by two of the input channels plus a suitable stereo output bus or headphone outputs. One needs gain controls somewhere in that signal path, preferably ganged, *eg* a headphone output gain control, or a ganged input gain on the two input channels. By way of example, on the cheap Realistic Cat No *32-1200B* mixer, the path between the auxiliary stereo input to the headphone outputs is phase inverting and has in-path gain controls. One then feeds the processing path's outputs back to the processing path's inputs, but connects the leads to interchange channels, so that the left output is fed to the right input and vice-versa (Fig 6). Besides the processing path which is fed back as described, the mixer also needs other stereo inputs mixed into the processing path, and a main stereo output subject to its own gain control (Fig 6 – in the Realistic mixer, any other input can be used, and the main output is used for outputting the shuffled signal).

One then mixes the stereo from other inputs, which is to be processed, into the fed-back signal path (Fig 6). The effect of the external feedback loop is to modify the stereo. For signals identical in both input channels (*ie* central mono or M), due to the inverting property of the processing signal path, the feedback is negative feedback, so that the M gain is reduced. For signals opposite in phase in the two channels (*ie* the S signal) the feedback is positive since the fed back signal adds to the signal in the other channel rather than cancelling it – this is due to the signal in the other channel being in opposite phase, so being in phase with the inverted fed back signal from the other channel. Thus the S gain is increased. As a result, as one turns up the gain in the processing signal path (by stereo-ganged gain controls at its input or output) the width is increased – up to the point where the positive feedback becomes unstable.

When setting up the feedback (in the Realistic by feeding the headphone output into the auxiliary inputs with channels swapped) take care to keep input and output gains down to start with and turn them up slowly to find out the point at which feedback howl occurs. (Warning – keep speaker or headphone levels well down when doing this!) Below this point, the circuit acts as a stereo widening control for stereo signals mixed in with the processing signal path, allowing adjustment all the way up to infinite width at the point of feedback howl. The simultaneous reduction of M gain as S gain increases give subjectively almost constant gain as width is varied in many mixes, although central images become quieter and edge images louder as the width is turned up. Interestingly, the configuration of Fig 6 turns the mixer into one whose panpots cover a wider stereo stage than usual (depending on the setting of the width-control gain) – so there is no reason why most studios cannot start using their mixer as a wide-stage mixer.

If one now inserts a stereo graphic equaliser into the feedback path (assuming the equaliser is not phase inverting – if the equaliser phase inverts, then a non-inverting processing signal path should be used) one has the possibility of varying the amount of feedback with frequency (Fig 7). Unfortunately, the equaliser will have some gain at all frequencies (even at maximum cut), so will tend to widen the image at all frequencies – this is not usually wanted or desirable.

To counteract this, one first sets up the system without the equaliser (or with the equaliser bypassed) at reasonable feedback settings, and feeds the stereo signals to be processed into the mixer via panpots. The idea is to narrow the inputted stereo image with the panpots to counteract the widening at unit gain in the feedback path. One pans the two channels in towards the centre. If one turns down the right channel input level on the main stereo input, while feeding a signal into the left channel, then turn the left channel panpot to that position at which no output emerges from the right channel main output (so that the panpot counteracts the widening of the feedback path). Similarly, one then turns down the left channel input gain, feeding in a right channel signal, and adjusts the right channel input's panpot to the point at which no output signal emerges from the left channel main output. Turning both input gains back up, the narrow image produced by the panpots at the signal inputs is now adjusted to counteract the widening of the feedback path at its unity gain, so as to retain normal stereo at this setting. (On the Realistic mixer, if the headphone and auxiliary input gains are both set halfway to 5, then panpots set to about 2.2 divisions from centre counteract the feedback effect.)

When one now re-inserts the graphic equaliser into the feedback path (Fig 7) its central unity gain settings will again give normal stereo, however, boosting any frequency band on the equaliser (equally in both equaliser channels) widens the stereo image in that band, and cutting it narrows the stereo in that band. Thus one has achieved an effective stereo shuffler by the circuit of Fig 7, and adjustment of the stereo-ganged equaliser bands simultaneously modify S and M gains so as to preserve frequency balance. If the equaliser has a bypass switch, one can use it to directly compare the processed and unprocessed stereo. Because of the ganging of the stereo bands, this system requires fewer control adjustments when being altered than the system of Fig 2, and so is easy to use. Moreover, one also has overall width control available by adjusting the overall gain within the feedback path.

Rather than using an external graphic equaliser, it is also possible to use equalisers built into the mixer channels used in the processing signal path instead. This has the advantage of requiring no external mixer circuitry other than connector leads, however, the two equaliser channels have to be adjusted separately, not (usually) being gangable. The other disadvantage here is that the equalisers built into the mixer are not (usually) graphics, being designed for creative alteration of tonal quality rather than shuffling. This makes it more difficult to visualise instantly the kind of shuffling produced by settings of the equalisers.

For regular use as an overall width processor and shuffler, it may well be worth obtaining a modest mixer such as the Realistic, with the minimum of unnecessary facilities, just for use as a width and shuffler control, in conjunction with a stereo-ganged graphic equaliser of the type encountered in some domestic hi-fi equipment. The alternatives are building equipment specifically designed for this processing, or using more highly specified professional equipment that is less convenient to adjust.

Inevitably, this article has only scratched the surface of stereo image reprocessing. More sophisticated techniques are possible. These include improvements in shuffling equaliser design (*eg* using phase-compensated equalisers[2]) through methods of modifying image sharpness, to dynamic signal dependent modifications of the stereo such as have been used in variable matrix decoders. Beyond that, there are the additional enhancements of Ambisonic reproduction technology[7,8], either applied to decoding stereo signals or to 'transcoding' them into an Ambisonic format[8,9].

Although I don't agree with everything in them, I recommend references 3 and 4 for additional ideas on possible uses of shuffling. I hope this article has provided you with a useful introduction to stereo enhancement techniques and useful tools in day-to-day recording and reprocessing work. You are certainly encouraged to experiment using different shuffler settings with different kinds of recordings and processing techniques. There is no telling what kinds of effects and improvements that you might come up with.

Acknowledgements

1 would like to thank David Griesinger, Tony Faulkner and Ivan Vernon McKinney, Jr, for drawing my attention to various aspects of microphone and shuffling techniques used in this article.

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Width settings and crosstalk

It is often useful to know what the gain of S (relative to that of M) is, so that one knows how much width increase has been applied. This can most easily be done by measuring the crosstalk of a left only signal on to the right channel (or vice-versa) – something that can be done on the mixer's meters. The crosstalk is the same whether the S/M gain is reduced or increased by a given number of dB! If S gain is increased, this crosstalk is out of phase but if the S-gain is reduced, the crosstalk is in phase.

S/M gain dB	**Crosstalk dB**	**S/M gain dB**	**Crosstalk dB**
±0.0	–∞	±7.0	–8.3
±1.0	–24.8	±8.0	–7.3
±2.0	–18.8	±10.0	–5.7
±3.0	–15.3	±12.0	–4.5
±4.0	–12.9	±15.0	–3.1
±5.0	–11.1	±20.0	–1.7
±6.0	–9.6		

The figures in this table are accurate only for the case when S has no phase shift relative to M, and so are most easily applied to frequency-independent width control.

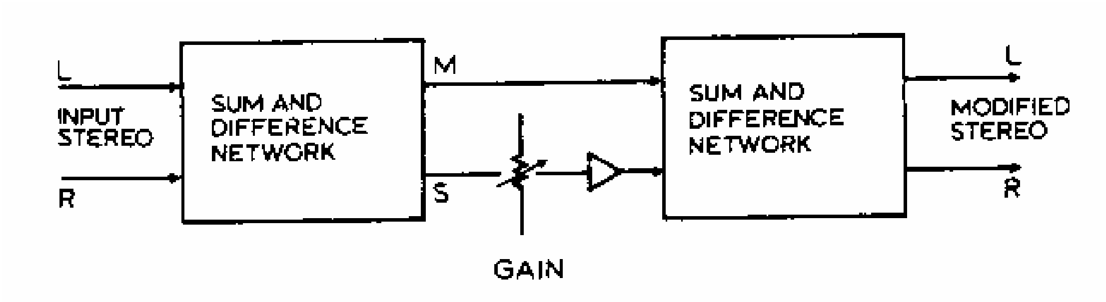

Fig 1: Blumlein's sum-and-difference width control

Blah

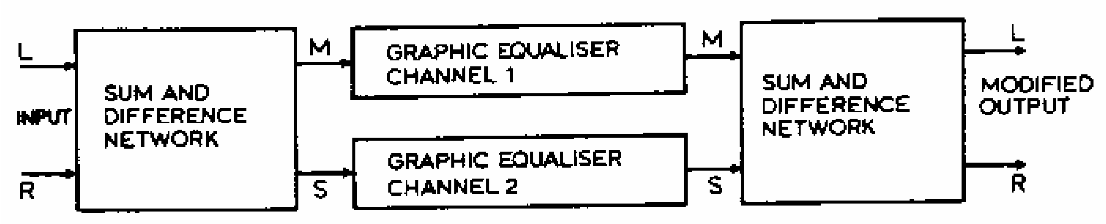

Fig 2: Sum-and-difference shuffling system

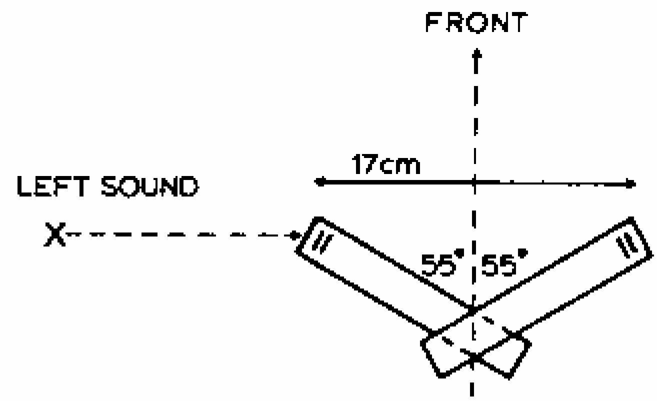

Fig 3: ORTF spaced crossed-over cardioid technique

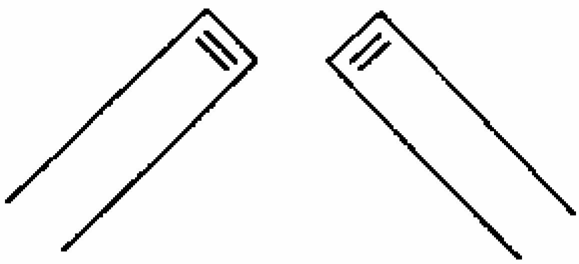

Fig 4: Non-recommended spacing of cardioid microphones for stereo recording

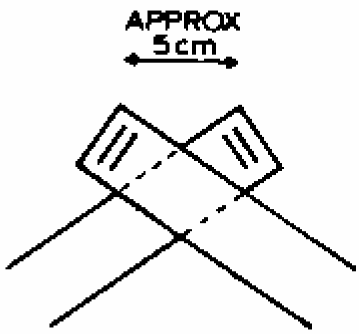

Fig 5: Crossed-over spaced cardioids for improved stereo imaging

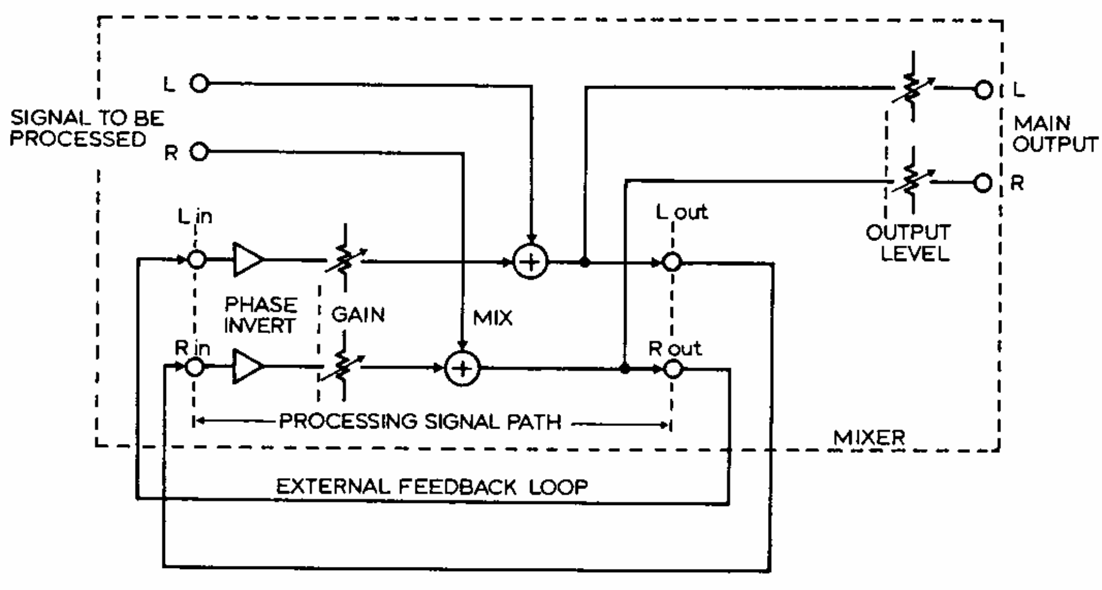

Fig 6: Stereo widening control implemented by feedback around a mixer.
The phase inversion and feedback can be anywhere in the feedback or processing signal path

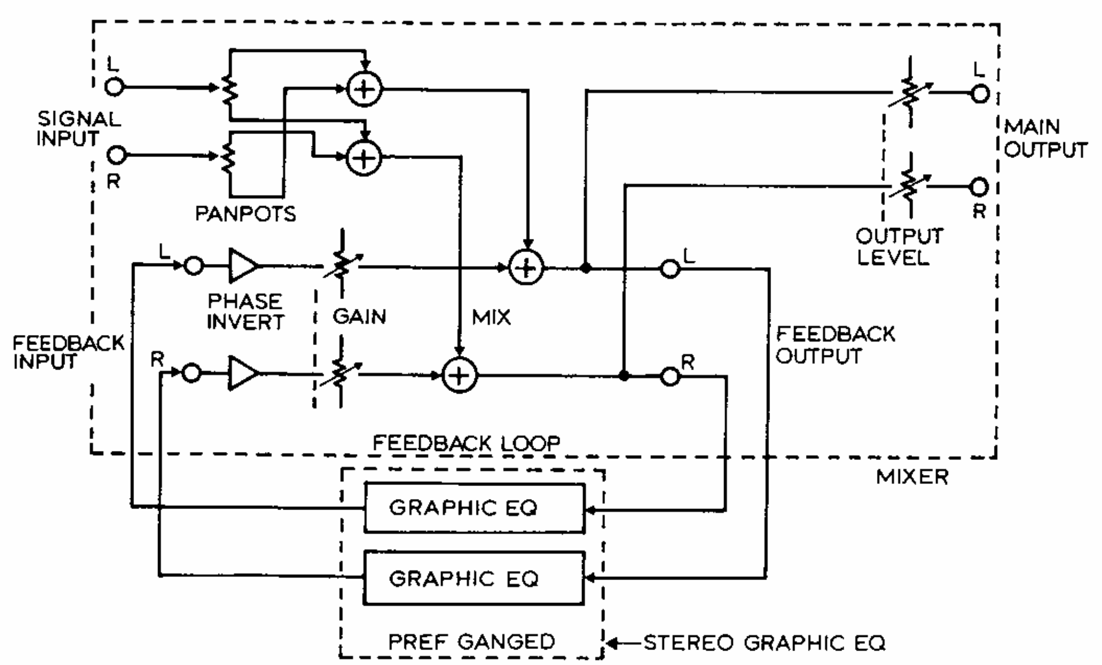

Fig 7: Complete shuffling system based on Fig 6 plus stereo graphic equaliser (preferably with ganged controls). Panpots in the main signal inputs compensate for the basic widening effect of the feedback loop