

Using A Stereo Microphone – A Primer

Stereo Recording

Practically all of our recordings today involve stereo. While surround releases are becoming more common, even these are largely based on a primary stereo image enhanced by effects and ambient sound to the rear and sides of the listener.

The standard studio technique for popular music production is to create a stereo sound field from multitracked recordings of monophonic sources panned left-right, with delays and reverberation added to create the sense of space. True stereo recording - capturing the full acoustic environment from a single vantage point using a process similar to that of the human ear provides a distinctly different kind of sound.

Single-point stereo miking has long been used for classical music recording. Here, the goal of the recordist is to capture and convey to the listener the experience of actually being in a concert hall with the orchestra. This is probably still the most common application of the single-point stereo microphone due to its convenience and its ability to accurately capture the sound of the acoustic space. Stereo microphone techniques need not be limited to classical recording, however, and are often appropriate for recording in an intimate jazz club or a church. In addition, recording such physically large instruments as pianos, drum kits, or large vocal groups with a

stereo microphone can prove useful in the multitrack mixing environment.

Hearing In Stereo

The two primary cues that our brain uses to help us locate a sound source in space are differences in arrival time (phase) and amplitude between our two ears. The process works pretty well in real acoustic spaces, but it's confounded somewhat by loudspeaker reproduction, where we hear sound coming from two discrete sources (the loudspeakers) as well as reflections from the surfaces of the room.

We can move the perceived location of a sound between the speakers simply by changing the amplitude of the signal fed to each speaker - this is what a pan pot or stereo balance control does. Another way of changing the apparent location of the sound between speakers is by sending the identical signal to both speakers at the same amplitude but delaying the sound to one speaker relative to the other. The sound arrives first in one ear, then the other. Our brain tells us that the source of the sound is coming from the direction of the ear which hears it first.

In general we need an amplitude difference of about 15 dB in order to make a sound to appear to be coming from one speaker or the other, but this is dependent on the program material. At low frequencies, a 15 dB amplitude

difference between speakers makes practically no difference in apparent location. The bass would have to be 25 to 30 dB louder on one channel than the other in order for it to be perceived as being off-center, and many playback systems do not have sufficient dynamic range to allow this.

Mixing engineers almost always pan bass to the center for this reason. In addition, engineers who were raised on vinyl learned that moving a large amount of low frequency energy off center causes the needle to jump out of the groove or damage the disk cutter. Back when “mastering” was the first step in the record manufacturing process, it was fairly common for the mastering engineer to separate the lows from the highs, mix the two low frequency channels to mono, and put the tracks back together, assuring that the bass was centered.

With CDs, those mechanical problems went away, but the problem of dealing with a large low frequency amplitude difference between channels in order to create the illusion off-center bass still remains. Phase difference due to the different path lengths from a source to each ear still works at low frequencies however. This is the mechanism by which we are able to locate low frequency sounds in the stereo field.

Alan Blumlein, an engineer and acoustician at EMI in England is generally considered to be the father of stereo recording and reproduction as we know it today. He

demonstrated mathematically that for lower frequencies, level differences between speakers are translated into phase differences, with the ear on the side of the louder speaker leading the other ear in phase.

This phase shift varies approximately linearly with frequency, so it represents a difference in time-of-arrival fairly accurately. At high frequencies, the phase differences get swamped out by amplitude differences, which is what our brain uses as a cue to their position.

The real world throws in some wrinkles like the shape of the listener’s head, but the bottom line is that in spite of a lot of theoretical and practical obstacles, we can actually hear stereo when listening to loudspeaker playback. As recording engineers, it’s our job to capture the sound field so that it can be realistically reproduced through loudspeakers, and there are several ways to do so.

Stereo Microphone Applications

During the 1930’s, both Blumlein in the UK and Harvey Fletcher (of the Fletcher-Munson loudness curves) at Bell Laboratories in the US developed techniques for using microphones in pairs for recording in stereo. The results of their experiments and demonstrations are all still relevant and applicable today.

Typically a stereo mic setup is appropriate in situations where acoustics are good (or at least not detrimental to the sound) and when the group being recorded is playing

all together and is well balanced. As you might surmise, this best describes orchestras, jazz bands, and acoustic folk artists. Since classical music is traditionally performed in good sounding halls, a well-placed stereo microphone setup is frequently all that's required to capture the performance accurately. And there's nothing like a stereo recording from a good seat in the house to hear what your band actually sounds like.

Typically classical stereo recordings are plug-and-play with no overdubs or punch-ins, no mixing or ducking, and perhaps only some editing between alternate takes. Stereo miking isn't limited to recording of a fully integrated performance, however. In the modern music studio, recording single instruments with a wide frequency range and large radiating area such as acoustic guitars, pianos, and percussion instruments can add a useful dimension to a multitrack mix. In addition, a stereo recording of a well balanced instrumental or vocal section as part of a multitrack project can add realism as well as save studio and mixdown time.

Stereo Microphone Arrays

Stereo microphone arrays fall into two general categories: coincident and spaced (sometimes called "near coincident"), which describe how the two microphones are positioned relative to each other.

A spaced stereo pair has the two microphones separated by a fairly small distance, usually less than a

foot, and sometimes with a baffle placed between them.

An ideal coincident pair has the diaphragms of the two microphones occupying the same physical space. Since that's physically impossible, we make do with "pretty close." This can be accomplished by positioning two individual microphones with their heads just about touching, and with an imaginary line passing through both diaphragms.

The important concept here is that the two microphones work as a pair to give us both the amplitude and phase differences we need in order to locate a sound in space.

Coincident Arrays

Coincident arrays require that the microphone diaphragms be placed as close together as possible. The LSD-2 incorporates two multi-pattern microphone elements in a single case with the diaphragm of one positioned directly above the other, making it very near to an ideal coincident stereo mic.

The mics in a stereo array can have a cardioid, bi-directional, or in certain cases omnidirectional directivity pattern. Since the mics are either pointed in different directions or are some distance apart, each mic gets a different perspective on the sound field. Since a directional mic's sensitivity varies with angular position, any sound originating off center will appear at a different level in each mic of the pair, giving us the amplitude cue to locate the sound in space.

Because the microphones are located in essentially the same space, a coincident pair has the advantage of mono compatibility – that there’s little or no loss of content when a coincident stereo recording are summed to mono. This is more important than you might realize. Although you may not consciously listen in mono very often, consider all the people who wake up to clock radios or still hear television audio through a single loudspeaker. If your stereo recording contains too much out-of-phase information between the channels, certain frequency ranges or overall ambience can drop out when mixed to mono, leaving a thin sound.

Spaced Arrays

Although the LSD-2 by nature of its construction doesn’t lend itself to use in a spaced array (unless you buy two of them), we would be amiss in a general discussion not to mention such an arrangement. In a spaced stereo array, the diaphragms of the two microphones are positioned several inches apart and are typically splayed outward from their center line. The spacing causes the arrival time from a source to be slightly different between the two mics, giving a phase cue for directional location. Spaced mics often give pleasing spaciousness to the recording, but their mono compatibility is poor.

Choices, Choices

Both spaced and coincident arrays have their strengths and

weaknesses and it’s up to the engineer which arrangement to use.

As a broad generalization, the coincident approach produces a stereo image that’s more focused, allowing the listener to more precisely pinpoint a sound source. By contrast, the spaced array yields a less precise stereo image, tends to produce recordings that are more open-sounding with a wider stereo spread.

Either technique can work well in a good recording environment, but in a room with a lot of nasty reflections that you can’t do anything about, spaced omnidirectional mics or coincident bi-directional mics are more likely to save the day than the more common coincident cardioid arrangement.

Some engineers have used a combination of spaced and coincident setups, combining them to take advantage of the best characteristics of each technique.

The LSD-2’s design allows you to experiment with different coincident arrangements easily, and the only way to tell which works best is by listening.

X-Y Coincident

The X-Y configuration is what most people think of as a “stereo pair”. The common starting position when using cardioid mics is to splay them 90 degrees apart with the line bisecting that angle pointing “straight ahead”. Hypercardioid mics usually

want to be splayed a bit further, around 110 to 120 degrees.

The mics are positioned at a distance from the source that gives a good balance between direct and reverberant sound, and the angle between the mics is adjusted to adequately cover the width of the source. This is counter-intuitive, but to get a wider coverage, the angle between the mics is decreased. The important thing to keep in mind is that 90 or 110 degrees is just a starting point, with the final decision resting with your ears and good judgment.

X-Y is simple to use since you get left and right channels directly from the mics. The mic that points toward the left side of the source becomes the left channel, ditto for the right.

When using a stereo pair in a mix with other mics, you don't always have to pan or assign the pair full left and right, you can put them anywhere you want. You can even reverse them. Whatever sounds right is right.

On the LSD-2, you'll notice a series of engraved dots around the ring at the top of the grill of the lower microphone capsule. The dot above the Studio Projects name is "front-and-center" with the other dots marking 45 degree increments. The front of the upper, rotatable capsule is marked with a single dot.

Don't forget to note the pattern switches. For X-Y operation, be sure to switch both microphone elements to the cardioid position.

Experience with the LSD-2 shows that 110 degrees is a good starting point.

There's no dot to mark this position, but it's conveniently located in line with the breakpoint of the low cut filter indication on the switch for the upper microphone.



With the upper capsule rotated to this position, the center line for the pair is located at 55 degrees. Again, there's no dot to mark this, but it's in line with the cardioid marker for the lower microphone switch. Aim that



cardioid marker to the center of your sound source (the Studio Projects name will be to the left as

you face the microphone) and you'll have a good starting place.

Remember that changing the angle between the capsules is done by rotating the top capsule relative to the bottom one. Rotating the top capsule will change the center line for the pair, so if you change the angle between mics, you'll need to

re-center by turning the housing slightly in its shock mount.

The distance from the source where the direct and reverberant sounds are equal in amplitude is known as the “critical distance.” Normally the mic position is always forward of the critical distance, but how far depends on how much reverberation you want in your recording.

Since the lower capsule is pointed toward the right side of the source when the mic is in its normal orientation on a stand, the XLR connector at the split end of the LSD-2's cable will be the right channel. If the mic is hanging with the cable going upward toward the ceiling, left and right will be reversed.

The Classic Blumlein Array

While Blumlein studied and documented many different microphone arrangements for stereo recording, the one that most often bears his name is a variation of the X-Y arrangement, employing two bi-directional (figure-8) mics rather than cardioids. In the classic Blumlein configuration, the microphones are always crossed at right angles (90 degrees) and positioned so that a line bisecting the angle between the capsules is aimed toward the center of the source.

Many people consider the classic Blumlein to be the perfect stereo microphone. Blumlein worked with ribbon microphones which, by nature of their design, have perfectly symmetrical directivity patterns and a perfect null at 90 degrees off axis.

Because a condenser microphone creates its bi-directional pattern by mixing of the outputs of two separate diaphragms, its pattern isn't perfectly symmetrical, but the LSD-2 in the figure-8 setting does an excellent job in a Blumlein setup.

While the bi-directional mic has equal sensitivity in both directions, the polarity of the signal produced by sounds arriving from opposite directions is opposite. Therefore it's important to be aware of which side is “front” so you can position both capsules so that the polarity between the stereo channels isn't inverted.

The front of the fixed capsule is the side of the body with the Studio Projects name. The front of the rotatable capsule is marked with the single dot. Viewed from above, if you rotate the top capsule counterclockwise so its dot lines up with the second dot around the lower capsule ring, you'll have the two capsules set correctly for the classic Blumlein configuration.

The 45 degree dot (above the figure-8 of the pattern selector for the lower capsule) is your center mark. Aim that toward the center of the sound source.

Again, like the X-Y pair, the lower capsule points toward the right in the normal orientation, so connect the red-jacketed XLR to the right channel of the recorder.

Since the bi-directional mics have considerably more sensitivity toward the rear than cardioids, the Blumlein array will pick up more reverberation

and ambience than an X-Y pair in the same location. You'll probably want to position the Blumlein pair somewhat closer to the sound source than you would an X-Y pair in order to avoid an overly reverberant recording.

If the acoustics of the room are poor, the classic Blumlein is probably not the best choice, but in a room with great acoustics, it's hard to beat.

M-S Coincident

Another popular coincident microphone setup is known as M-S, for Mid and Side. An M-S stereo setup tends to be more fussy about the acoustical environment than X-Y (more accurately, it's easier to find a workable position an X-Y pair than an M-S when the acoustics aren't so great) but M-S offers a greater degree of control than X-Y.

In M-S, a forward-facing mic (the Mid mic) is positioned for a well balanced mono pickup of the sound source, and a sideways-facing bi-directional mic (the Side mic) contributes width information.

Since the figure-8 directivity pattern has a null in what's the forward direction of an M-S pair, it (in theory, and pretty much in practice) does not change the overall balance established by the Mid mic.

The Mid mic is conventionally a cardioid pattern, but it can also be omnidirectional or bi-directional. The selectable patterns of the LSD-2 makes experimentation with different

Mid patterns in an M-S setup painless.

To use the LSD-2 as a traditional M-S array, select the cardioid pattern on the lower capsule and the bi-directional pattern on the upper capsule. Rotate the upper capsule 90 degrees counterclockwise (looking from the top) so that the dot on the rotating ring is lined up with the second dot on the fixed ring. Position the microphone so that the lower capsule faces the sound source. Note that you can use either capsule as the Mid or Side mic, but since we put our name on the front, you might as well point it toward the performer

M-S Arithmetic

M-S is more complex than the other configurations we've described because the output signals of the two microphones don't translate directly to left and right channels. Some electrical manipulation is required to derive meaningful left-right stereo from the Mid and Side outputs.

We usually use the terms "encoding" and "decoding" or "matrixing" to describe the process used to translate M and S signals to left and right. To understand how this works, we need to use some simple algebra.

By convention, positive pressure on one side of the diaphragm of a bi-directional mic produces a positive output voltage, while positive pressure on the opposite side produces a negative voltage. A cardioid mic produces a positive

voltage for a positive sound pressure toward its most sensitive side.

The outputs of the two mics oriented as described combine in phase when a sound is arriving from the left side of center and out of phase for a sound arriving from the right. Because of the bi-directional microphone's sensitivity null on-center, an on-axis source is heard by the Mid microphone only.

The basic M-S relationships are as follows:

Mid + Side = Left
Mid - Side = Right

A sound arriving from the left produces signals from both the Mid and Side mics which are in phase. By simply adding them, we get the left channel signal. A sound arriving from the right will be out of phase between the mid and side mics. To get the right channel, we must subtract the Side signal from the Mid signal. We do this by inverting the polarity of the Side mic before it to the Mid mic signal.

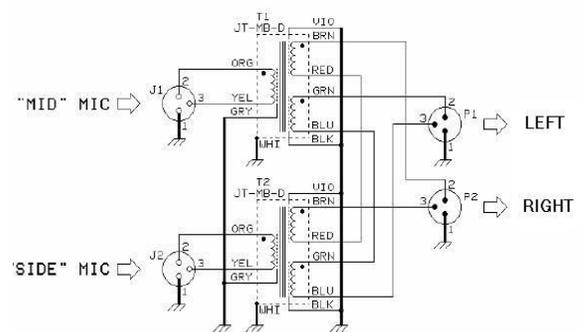
Taking this one step further, we see that:

$$\begin{aligned} L + R &= (M + S) + (M - S) \\ &= M + S + M - S \\ &= 2M \end{aligned}$$

Notice that when the left and right channels derived from an M-S pair are summed to mono, the side mic cancels out of the equation leaving only the forward-pointing mid mic for perfect mono.

Matrixing

An M-S matrix can be constructed from a pair of transformers with split secondaries wired to produce the sum and difference of the voltages at the two primaries. Follow the schematic below (courtesy of Jensen Transformers) and you'll see what happens. The dots on the transformer windings indicate polarity.



Let's work backwards from the outputs. Pin 2 of the Left output is connected to the dotted end of one winding of the Mid mic transformer. The other end of this winding is connected to the dotted end of one winding of the Side mic transformer, and the other end of the Side mic winding is connected to Pin 3 of the Left output connector. This puts the two windings in series and in phase.

Pin 2 of the Right output is connected to the dotted end of the other secondary winding of the Mid mic transformer. The opposite end of this winding is connected to the undotted end of the Side mic transformer, and the dotted end of that winding is connected to Pin 3.

It's this backwards wiring of the lower secondary winding on the Side

mic transformer that does the subtraction to give us the Right output.

There's a hitch with using transformers at the mic end of the recording chain. Since you can't pass DC through a transformer, phantom power supplied by the console or outboard preamp won't make it to the microphones. You could add a phantom power supply to a transformer matrix but a more satisfactory arrangement is to connect the mics directly to the preamp and decode the line level preamp output on its way to the recorder.

This can be done with transformers, however the Jensen JT-MB series shown in the above diagram are not designed to operate at the high output level typical of a modern mic preamp. Audio Engineering Associates is one source for an M-S decoder designed for these levels.

Decoding at the Console Inserts

When using a console's mic preamps, a line level decoder can be conveniently patched into the channel insert points. In a typical console, the insert send directly follow the mic preamp while the insert returns come before the channel faders. With a decoder patched between the M-S mic channel insert sends and returns, the faders act as if they're directly controlling left and right stereo mics.

The mic preamp gain trim and anything else ahead of the insert point (EQ on some consoles) affects

the relationship between mid and side mics, so you have to be aware of what's in the signal path.

One more tip: Since the channel inserts on most modern consoles are unbalanced, a simple single ended sum and difference circuit based on op-amps can be constructed without much difficulty.

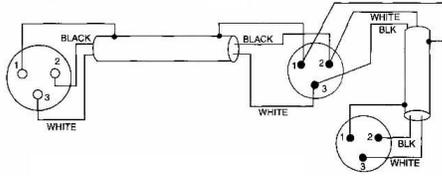
Keeping it M and S

One approach favored by some field recordists is to record the mid and side signals directly, connecting an M-S matrix only in the monitor (or headphone) path when recording. This allows you to hear a normal stereo image while deciding where to place the microphone, and for monitor in stereo while tracking. Afterward, you can "remix" the Mid and Side tracks through an M-S decoder in your studio and adjust the width while listening in a familiar monitoring environment.

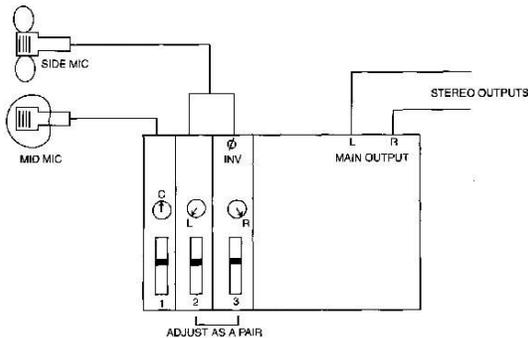
A few mic preamps have M-S decoding built in. One of these preamps, together with the LSD-2 and a stereo recorder makes a very convenient and compact location recording package. M-S decoding is also found in many DAW programs. This is a good solution for the home recording engineer with a studio based around a computer, or for post-production of a raw M-S recording.

The Console as a Matrix

You can also decode the M-S signals using a mixing console and a simple "Y" cable.



Connect the Mid mic to one channel and center its pan pot. Split the Side mic's output with the Y cable and connect it to two other (usually adjacent) channels.



Pan one of those channels full left, pan the other full right, and invert the polarity of the right-panned channel.

If your console has a polarity (phase) reverse switch, use it as shown in the diagram above. If not, swap Pins 2 and 3 in one branch of the Y cable to give the required polarity inversion. (Mark that reversed polarity connector!)

By the way, it's OK to "Y" together two phantom powered inputs if your console doesn't have phantom power switches for each channel.

Once you have things connected as show, bring the faders of the two side mic channels up to their working position. It's handy to tie them together since they'll always be moved as a pair. A broken piece of

pencil and a rubber band trick will do the trick.

Start with the Mid mic fader all the way off or mute the channel. Set the input trims on the Side channels in the ballpark for what you'll be recording, solo both side mic channels together (or temporarily pan both of them to the same side). Get some sound going into the mic, and adjust either of the trims to null the output. When you get pretty close to the null point, raise the monitor volume so you can really nail it. This makes the level of the two out-of-phase channels equal.

Un-solo (or reset the pans), start the music, and check your levels. As a starting point, when the music is as loud as it's going to get, analog VU meters should reach around -3 VU. A good starting point with a digital meter is slightly below -20 dBFS. If you're very far off, reset the trims on the Side mic channels, then re-null to get the gains exactly the same.

Now listen to the left and right outputs. It should sound positively awful because you're listening to the two channels out of phase and everything is off-axis to the Side microphone to boot. However, when you bring up the Mid mic fader, you'll start hearing real stereo. Leave the Side mic faders in their starting position and adjust the Mid fader for the desired stereo spread.

Be sure to check your recording level and preamp gain (be sure you're not clipping up front) before starting the session. Small level adjustments can be made with the mixer's master

fader, but if you're way off base when the crashing crescendo comes, you may need to re-trim to avoid clipping the mic preamps. It's a bit awkward to juggle all those faders if you need to change overall gain once you've set things up, but once you get the hang of it, it's not too difficult.

Double M-S

One of the conundrums with M-S stereo, actually with any single-point stereo recording setup, is that when the microphone is positioned to get good detail, it may not pick up sufficient room ambiance, yielding a rather dry sounding recording. Moving the mics further back into the reverberant field will capture more ambiance, but go too far back and you'll lose the up-front clarity.

Curt Wittig and Neil Muncy, two engineers who specialized in minimalist recording, when searching for a solution to this problem, came up with a setup which they dubbed Double M-S.

In Double M-S, one M-S pair is positioned as close to the source as necessarily to achieve optimum detail, for the moment ignoring the balance between direct and ambient sound. A second M-S pair is employed for the sole purpose of picking up ambient sound.

The "ambience" pair is placed beyond the critical distance, sometimes even facing the rear of the hall. Its output is mixed with the main M-S pair to provide the desired amount of ambience. Think of it as

the output from a stereo reverb unit which can be added to taste. As a rule of thumb, mic pairs should be less than about 75 feet apart to avoid echoes.

Fans of this technique believe it may be the best of all possible worlds, providing the spaciousness of the near-coincident array, the location accuracy of a coincident array, and the mono compatibility of an M-S recording. And we get to sell you another LSD-2.

Doin' The Blumlein Shuffle

Alan Blumlein conceived stereo not only as left and right signals, but also in terms of a sum ($L+R=M$) and difference ($L-R=S$) that we know from the basic M-S equations. He proved mathematically that these relationships could be used to modify the "stereoness" of recordings. What's particularly cool is that this actually works in practice. By creating a sum and difference from a stereo signal, you can manipulate them like the outputs of an M-S microphone, putting them back together with a different perspective.

Decreasing the level of the L-R (difference) signal with respect to the L+R (sum) signal narrows the stereo width. This is no big deal since you can accomplish the same thing simply by panning the left and right channels toward the center. However, by increasing the level of the difference signal relative to the sum, you can increase the apparent stereo width, effectively turning the pan pots beyond their stops.

You can take this too far, ultimately punching a hole in the middle of the stereo image, but sometimes unconventional procedures yield creative results.

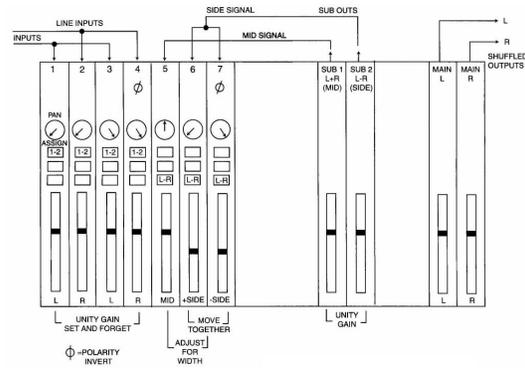
Blumlein also discovered that the image accuracy of a stereo pair could be improved by applying an equalizer to the difference signal. The process of equalizing the sum and difference independently before combining them into left and right is called “shuffling,” effectively creating a frequency dependent width control.

Since most microphones are less directional at low frequencies, shuffling can help to improve the ability to localize bass instruments in the stereo field. Inserting equalizers in line with the sum and difference derived from a stereo signal is a useful way to experiment with this process.

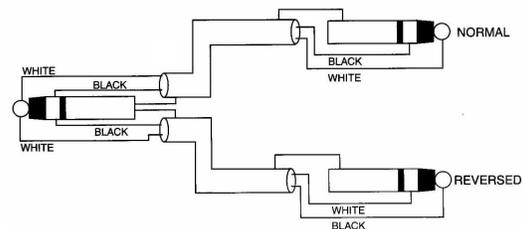
If you have a 4-bus or even a semi-4-bus console (such as the Mackie 1202-VLZ Pro) and some Y-cables (a couple wired for polarity reversal if your console lacks polarity/phase reverse switches), try shuffling a stereo recording. Create the sum and difference signals on one pair of busses. Then return the sum (Mid) and difference (Side) signals to another pair of line inputs just as if they were the two outputs of an M-S mic pair. Assign the outputs of your M-S mix to the mixer’s main left and right outputs.

Now you’re back to stereo, but you can experiment with the balance between sum and difference signals and add some equalization to spread

things out. A good place to start is boosting the difference signal by three to six dB in the range of 400 to 600 Hz. This sounds complicated, but it’s easy to understand what’s happening if you follow the block diagram of the mixer setup below:



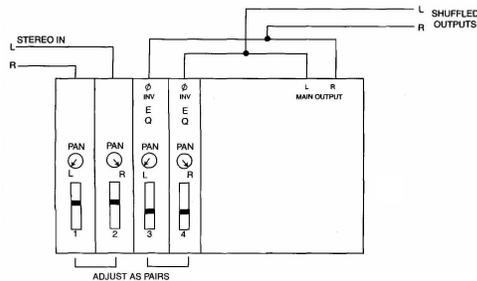
Here’s how to make a TRS “Y” cable for splitting a balanced bus output into normal and reversed polarity signals:



But if you have only a simple stereo mixer, thanks to a clever method conceived by the late stereo guru Michael Gerson, you can still shuffle.

Gerson observed that inverting the polarity of the left and right outputs and cross-coupling them back to another pair of inputs was electrically equivalent to Blumlein’s classic sum and difference circuit. Here’s how to set up this shuffler. Note that left output is connected to the inverted

channel that's panned to the right and the right output is connected to the inverted channel panned to the left. This is the trick that makes it work.



The thing that you need to watch with this setup is that there's really a connection from the output to the input. This is the standard recipe for feedback. Start with the gain of the fed back (inverted) channels at minimum and bring it up slowly. Raise their gain too far and the howling will start, but you'll have plenty of working range before that happens.

Mastering the Shuffle

Equalizing the low end of the difference signal can sometimes save the day even when working with a multitrack mix. If the tracks have a lot of low frequency leakage, for example if the band was recorded live and the bass is on everyone's track, you can sometimes improve the apparent definition of the instruments in the mix using this technique. It also can help to spread out a mix in which things have been too crowded in the center.

Many mastering engineers are aware of this trick and it's employed by a few "mastering" software plug-ins, but you can

A Non-Stereo LSD-2 Application

While the purpose of this paper is to describe how to use the LSD-2 as a stereo microphone, there's a handy studio application worth a mention.

When working with a singer who plays an acoustic instrument, as engineers, we frequently prefer to record the instrument and vocal in separate passes and on their own tracks to allow us to apply different EQ to the voice and instrument, and for when the inevitable "Can I just fix that one word?" question arises. Many singers without a lot of studio experience, however, aren't comfortable recording in this manner. By taking advantage of the almost perfect null of a bi-directional microphone in the plane of the diaphragm, it's possible to position an LSD-2 so that the voice is in the null of the instrument microphone and vice versa, allowing them to be recorded together without a lot of leakage.

Mount the LSD-2 so that it's pointed horizontally and twist and turn the capsules aiming for the best compromise between sound quality and separation. You won't get perfect isolation, but because the two microphones are very closely spaced, any leakage that you have will be in phase, so bringing up the vocal won't thin out the instrument or vice versa, and with a combination of luck and careful placement, you can successfully do punch-ins on one of the tracks without enough of the original part from the other track to be audible in the mix.

Experiment with your microphone. It's fun to hear all the different variations you can get with just a little tweaking here and there. Mic placement and technique is something that you really only learn by doing, but hopefully this guide will turn you on to some techniques that will prove useful in your work.