Mixing For Three-Channel Reinforcement

by Jim Brown

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Jim Brown is an acoustic consultant based in Chicago, where he specializes in the design of large sound reinforcement systems for theaters, churches, stadiums, arenas, and broadcast facilities. He has also done extensive production work and mixing for broadcast, recording, and reinforcement. He received his BSEE from the University of Cincinnati in 1964, has worked in professional audio since 1970, and has been a full time consultant since 1984. His client list includes Wrigley Field, United Airlines O'Hare Terminal, Northwestern University's Football Stadium, Liberty University, Grace Cathedral (San Francisco), NBC, ABC, CBS, NPR, WTTW-TV, WGN-TV, and numerous churches and performance facilities. He earned FCC First Class Radiotelephone (broadcast) and Amateur Extra class licenses in 1959, and is occasionally active as W9NEC. Mr. Brown is a member of SMPTE, SBE, and the Technical Committee of the AES for Acoustics and Sound Reinforcement, and is active in AES Standards Committee work.

Two and three channel sound reinforcement is a concept whose time has finally arrived. My firm has been designing two and three channel systems for nearly ten years now, and can report that the final result is well worth the extra effort. Tom Young (Accentech, NYC) Craig Janssen (Acoustic Dimensions, Dallas), Jim Carey, and Bill Thrasher are all successfully building two and three channel systems for performance and worship spaces, and an increasing number of multichannel systems are being used on Broadway. Michael Rives and Jim Van Bergen are two designers of both systems and creative content who work extensively with multi-channel systems. A well designed two or three channel system can blow the pants off of a monophonic system which uses significantly more powerful equipment.

Choirs, orchestras, and praise bands have much more life, with a far more open and natural sound. A minister, rabbi, or priest can be panned to the channel corresponding to the location where he or she is preaching, celebrating the mass, or receiving testimonials. For dramatic presentations, actors wearing wireless mics can be panned around a stage, and floor mounted mics can each be fed to their own cluster, creating a very natural sound field.

Throughout this discussion, we'll be talking about both two and three channel systems, and we'll use the word "stereo" to describe them. Sometimes we'll be talking about two channel left/right stereo, and sometimes to left/center/right systems with an actual center channel. At the early stages of a project, we'll try to make some initial decisions as to whether stereo should be used at all, and, subsequently, whether to go with a two or three channel configuration. Budgets are always an issue, and even though a three channel system may have been designed, we often must scale the actual installation back to only two. But our success with stereo systems has caused us to consider two channels the minimum for any space which allow it -- when the budget gets thin, we'll nearly always lose the center cluster, not the left and right!

Not all rooms lend themselves to multichannel reinforcement. Room geometry, performance and/or worship locations, and seating layouts are the major determining factors. Each listener must hear each channel at more or less the same level, and within about 20 ms of the same time. Low ceiling spaces, spaces with very wide seating layouts, and spaces with a thrust stage or worship platform can be difficult (costly) or even make these systems inappropriate.

The easiest spaces to cover are those which meets all of these conditions: (1) a more or less rectangular footprint that's deeper than it is wide, (2) a stage/worship focus at the center of a short wall, (3) no deep under-balcony seating, and (4) cluster locations which are at least 25 feet above the front rows and more or less even with the front of the stage/worship focus area. Loudspeaker system coverage modeling is almost a necessity here. We use EASE to model every system, and carefully study coverage and arrival times before making any recommendations.

Cluster height is important because it makes it easier to get good even distribution of levels, and because it helps to equalize arrival times from widely spaced clusters. Greater heights allow better loudness equality by reducing the vertical angle which must be covered by the cluster. And timing equality results from the height component of the triangulated distance sound travels to any given seat. To understand the this, think about a right triangle with one side being the cluster height above seating, a second side being the distance to the listener in the horizontal plane, and the travel time (converted to distance) being the hypotenuse. The greater the ceiling height, the less the travel time changes as you move left to right around the listening area.

SIDEBAR TEXT An example of how loudspeaker height reduces the travel time differences between left and right loudspeakers. The loudspeakers are 30 feet above the listener's ears, and 34 feet apart. A listener is 20 feet directly in front of the left loudspeaker. The distance to the nearest loudspeaker is 20 feet in plan, but 36 feet when height is taken into account. The distance to the right loudspeaker is 40 feet in plan, but 50 feet considering height. The difference is only 14 feet. With sound travelling at 1.15 ft/ms, the difference is 16 ms.

Seating width is important for the same reasons as height. As rooms become wider, it's harder to equalize levels because a cluster must throw to the far side, creating hot spots down the center when the room gets too wide. Greater arrival time differences between left and right clusters begin stretching the limits of the stereo sound field itself, causing the image to break down into three separate sources. And in a worst case scenario, sounds which are in all three clusters (because of normal mic and mixing techniques) will be heard as echoes at seats far off centerline.

Balcony seating is usually easy to cover with delayed left, center, and right satellite clusters, often making these some of the best seats in the house! Seating in the first few rows or deep under a balcony is a challenge, because it must often (but not always) be covered with monophonic delayed loudspeakers.

Thrust stages and worship platforms create a condition where the focus is well out into the listener area, making the definition of left and right vary depending on where each listener is, and "in-the-round" configurations carry this to the extreme. This doesn't mean that some form of stereo or left/center/right system shouldn't be attempted, but it changes the rules significantly for both system layout and mixing.

Making stereo reinforcement work nearly always involves compromises, more in some seats than in others. In most spaces, however, careful design can nearly always yield a final result which is much more satisfying than an equivalent monophonic system, *EVEN IN THOSE SEATS WHICH ARE MOST COMPROMISED!* One of our standard techniques is to counter the effects of precedence for listeners far off centerline by allowing the coverage of the nearest side cluster to fall off as much as 6 dB at the edge of the seating area, while maintaining the sound from the far cluster at full level. We will often provide full stereo to those listeners who can benefit from it and a mix of monophonic and stereo coverage to those who can't (under a balcony, for example). And, realizing that a principle benefit of stereo is to allow more open mic technique, we've been known to use a row of alternating A/B stereo loudspeakers under a balcony. These techniques lose directional realism but keep the more open sound quality and "airiness" of stereo.

Major Configuration Types Most systems fall into one of several basic configurations. The simplest is a true left/center/right or left/right system, with all delayed loudspeakers set up in matching left/center/right or left/right pairs or triplets. Each seat hears all three channels at equal level, and no one hears a mono sum.

The second type is a mostly LCR or LR system, but with some seats covered by a loudspeaker sub-system carrying the monophonic sum. When this type is used, we try to design so that even mono seats get some stereo energy, and set delay times so that the monophonic arrivals are later than all three front clusters in all seats. We also try to limit the ratio of monophonic energy to being no greater than the level of the stereo precedence in as many seats as possible. This is important, because mics arranged in spaced left/center/right triplets or left/right pairs generally don't combine well into a monophonic sum.

Why use monophonic delayed loudspeakers at all in this configuration? Simply (or not so simply) this. First, it's mostly the high end that is falling off at the back of the room, or obstructed by a balcony overhang, so we need a little extra energy to maintain good presence. If we set the delayed loudspeakers to provide approximately the same level as the main clusters, the combined level will be 3 dB higher than if no delayed ring were used, and there will be an acceptable combination of comb filtered and un-comb filtered sound. Ideal? Certainly not. Better than if no delays were used? Yep!

A third type is a hybrid front system dictated by a thrust stage or worship platform. We've got a couple of systems like this in the works right now. One is a theater, and one a Pentecostal church with a very ambitious music program. In both spaces, we're designing a monophonic center cluster, a left/right pair of stereo clusters, and monophonic delayed clusters.

In the theater, a pit band may be either upstage or downstage, but actors will be primarily on the stage. Also, seating surrounds the thrust stage for nearly 300 degrees, so the center cluster is approximately halfway between the upstage and downstage limits of the stage, but the stereo clusters are at the upstage wall. The center cluster here will carry only a mono mix of live actors wearing wireless mics, while the stereo clusters will carry the pit band and ambient sound effects.

In the church, the center cluster will be used for the spoken word and for lead vocals by the praise team, while the stereo clusters will carry only the praise band. But there will certainly be situations where three channel panning will be desired for both singers and the band. Systems for the two spaces will thus be mixed quite differently. More about this later.

As an example of the fourth type of system, consider an "in-the-round" theater, covered by a four clusters encircling the stage at 90 degree intervals. Each cluster has left and right facing elements, and each is fed by its own mix channel. Chances are this system will also need some delayed elements to cover the back of the audience. When you're playing in the round, actors need the most amplification to seats which are behind their backs at any given time. We'll want a combination of mono and multiple stereo mixes for this system, and mixing for it is going to need some serious thought. Again, we'll discuss this a bit later.

Mixing for a "Type One" system is certainly the simplest to describe. For basic speech, we like to pan the talker to the cluster or channel which most closely fits the image. In a typical cruciform church with one of the more traditional forms of worship, this means that mics at a pulpit or lectern left or right of center will get assigned to those channels, often with a hard pan. With this kind of a setup, an altar mic would go straight down the center.

A Simple Traditional Church Stereo mixing doesn't have to be complicated to be effective. One Episcopal congregation for whom we worked does no amplified music at all and uses only three mics, one at a time. The lectern mic goes straight to the left cluster, the pulpit mic straight to the right cluster, and the wireless to both. There's no stereo mixer at all -- logic outputs of the automatic mixer drive VCA's to control how it's output gets routed, and only one mic can be active at once.

This system very effectively supports their liturgy, because it pulls the listener's attention directly towards the talker, rather than from some undefined "up there" location. The result is greatly improved attention span during a sermon, as well as the sense that sound is natural rather than "amplified."

Multi-Channel Theme Parks Michael Rives has designed LCR systems with multiple delayed and surround elements for applications best described as theme parks. Live actors and singers perform using wireless mics and are mixed with recorded music provided by multiple laserdisc players. Mediamatrix provides signal processing, and is controlled by a combination of a live operator controlling levels and initiating cues, and an LCS system to actively configure the system during the progress of the show. Here, a very complex program is mixed to eight or more directional program channels, with additional channels supplementing main LCR channels on delay.

For these systems, there is no traditional console at all. Instead, a programmable show control matrix system provides all of the functions of a console. The show is essentially mixed once by the creative team, with a minimum of dynamic fine-tuning of the mix to compensate for the dynamics of each live performance and audience.

Complex Musical Content Multichannel systems really shine when used to support contemporary music. Best results are always obtained by considering the sound source, the mics, the mixing, and the loudspeakers as a complete **system**, rather than as individual elements. At the same time, miking and mixing a complete production must be thought of as combining many separate, but closely interrelated elements. For these reasons, we'll approach this with a real world example.

Praise Band, Soloists, and Gospel Choir This is one of the toughest problems in modern sound reinforcement. Choir, musicians, and individual soloists must all hear each other clearly if they are to create a coherent musical result. But it's also critical that sound from louder instruments be prevented from leaking into choir mics and mics for quieter instruments. If the praise band plays too loud or is too close to the choir, choir mics will pick up lots of wild sound from bass, drums, and guitars. This wild sound will be muddy and undefined, both because of reverberation, and because it is getting picked up in multiple mics at different distances (that nasty devil, acoustic phase cancellation rearing its ugly head again). To overcome it, the direct feeds and mics for these instruments have to be turned up louder to give them some definition. When that happens, the choir fades further into the background. The result is greater loudness than most worshippers really want, and you still can't hear the choir.

Careful attention must thus be given to where the praise band sets up, where the choir will be, and how the choir will be miked. Certainly, the louder instruments should be baffled as much as possible, then miked and mixed. While we love unamplified sound when it can be effectively used, this isn't usually one of those times. And the choir is going to need to hear soloists and the spoken word, so monitors are needed, but they can't be allowed to get back into the choir mics!

The key to miking and mixing a choir is thus based on several factors, all of which must be well executed. The leakage and acoustic problems are generally so severe that it takes at least 10-15 dB of improvement to reach an acceptable solution.

- Mics must be located close enough to the choir so they don't have to be turned up too loud, and their directivity must be carefully used to minimize leakage.
- All loud sound sources and monitors must be located to the rear of the choir mics.
- Sound from loud instruments or monitors cannot be allowed to bounce off of hard surfaces behind the choir and get into the choir mics.
- The natural directivity of the human voice must be utilized.
- Monitor systems must provide uniform distribution throughout the choir and well equalized.
- To prevent acoustic phase cancellation, mics which hear the same sound at equal level cannot be mixed into the same channel. (Here's where multichannel systems really help.)

The Pentecostal church described earlier illustrates both the problem and possible solutions. The choir is elevated several feet, and at the very back of the worship platform (i.e., upstage). Directly in front of it, but off centerline and in a moderately depressed quasi-pit area, is the praise band. At the front of the praise band is a concert grand piano. And in front of the platform are the solo singers. The church is pretty reverberant for this kind of worship -- about 1.6 seconds throughout the mid range. And there's a marble wall behind the choir, which reflects the praise band straight into the choir mics, which are at head level and facing the choir. The reinforcement system is suspended above the front of the platform, and feedback is not a problem. But leakage is a "train wreck!"

We're recommending a multipronged approach. Each measure reduces leakage, some only a few dB, some a lot more. No one of them will do it alone, but the combined result will be a major improvement.

- A new reinforcement system is needed, to provide much better control of directivity to minimize the spill of sound into the reverberant field. We're recommending an LCR system.
- The drummer needs to be baffled carefully.
- The praise band needs in the ear monitoring. We'll recommend a wired system, designed around Shure FP-22 beltpack units.
- Instrument amplifiers need to be turned down (a lot). This lets our sound system, which has good directivity, keep the sound out of the reverberant field and the choir mics. By contrast, the instrument amps have little directivity at all, generating lots of reverberant mud.
- The choir needs monitors coming in from above and in front of them. Reflections from the marble wall will be grounded into the carpeted floor and the bodies of the choir themselves, so they won't leak into the mics.
- Choir mics need to be suspended above and in front of the choir, carefully angled so that the monitors are at 180 degrees off axis. This location also places reflections of the praise band from the marble wall off axis of the mics. It also takes maximum advantage of the directivity of the human voice.

- Six mics are needed, alternating them between left, center, and right clusters as them move left to right across the platform. This virtually eliminates acoustic phase cancellation. We're recommending that optimum mic heights be first determined "by ear" using portable stands, then flying them in from above once the sweet spots are found.
- The piano needs to be moved forward, away from the praise band. This further reduces leakage into piano mics.
- The soloists need in-ear monitoring. This will be a wireless system.
- Careful equalization is needed. We strongly recommend a very effective high pass filter (at least two poles, preferably three) on **all** mics in a live system.

The Mix Console This system needs a console which can spread singers and the praise band across all three channels. At least four LCR sub-groups are needed. One handles the choir, one the singers, one the horns, guitars, and keyboards, and one the rhythm section.

Most live mixers, myself included, like to use two primary mixing strategies for LCR. The first is to put soloists hard down the center and wrap the band and choir around them in L and R only. The second is utilize all three channels for soloists, band, and choir. Any given program may work better with one or the other, or with a combination of both. For this reason, an LCR mix console should be capable of both LCR and LR panning.

In a paper delivered to the AES in 1989, (NY, October 18-21) this author outlined a configuration for LCR panning which is currently used in all mix consoles designed for sound reinforcement. In this configuration, a centered pan pot sends signal only to the center channel. As the mic is panned toward the left, its output is panned between left and center outputs, becoming all left output with a hard left pan. Panning to the right of center operates symmetrically.

In that same paper, I also recommended that each channel be switchable between two and three channel panning. When in the two channel mode, the pan pot operates between left and right outputs only. This has several advantages. First, it allows any mic to be mixed LR or LCR for any program. Second, it allows the same console to tour into one venue with an LR system and in a different one with an LCR system. Third, it allows a single console to be manufactured and sold to users of both LR and LCR systems, reducing the overall cost of both with economies of scale.

The most important equalization any console can have is the high pass filter. High pass filtering is the key to cleaning the mud out of the mix, by reducing leakage of bass and drums into vocal mics, by minimizing the energy in the reverberant field, by reducing breath pops and wind noise, and by controlling proximity effect.

We recommend a sweepable high pass filter that can be set anywhere between 20 Hz and 500 Hz. Vocal and horn mics get this filter set to about 250 Hz, the left hand piano mic may go down to 50 Hz, the right hand may be set as high as 500 Hz! The bass and kick go down to 40 Hz (or the lower limit of the main system), other drum mics will end up in the 100 - 500 Hz range.

The second most important equalization is high frequency tailoring to fix the excessive high frequency peaking built into so many popular vocal mics. While these "presence peaks" can make an inferior system sound better, they can slice your head off in a good system if not carefully controlled. Presence peaking is generally centered in the 5 KHz to 10 KHz region, just where de-essers operate to remove the sibilance created in large part by the excessive peaking! A sweepable parametric that can get a little over an octave wide when it's 5 dB down is very helpful here.

Live Actors on a Stage Multichannel systems are a very powerful tool when miking a stage for drama. Three mics, one per channel (two mics for an LR system), can provide a very effective sound field. Here, the key is to design the loudspeaker system so that it keeps sound off the stage allowing the mics to be given enough gain in the mix. Most of our designs rely on horns for highs and mids, and the very effective bass arrays pioneered by Craig Janssen (Acoustic Dimensions, Dallas). The 250 Hz high pass filter is very useful here. Mics can be plate mounted cardioids on the floor, or cardioids flown in from above, depending on spatial relationships between actors and the audio system.

Wireless mics are another story. For some programs, they need to be panned around between the channels as the actors or singers move. For others, locking them into the center channel is a better choice. When two actors play a scene very close together, it is often better to use a wireless one of them wears to pick up both voices. (This doesn't preclude having both of them wear their own for scenes they play separately, simply leaving one out of the mix when they are close to each other). The high pass filter is still very important here -- lavs need a fair amount of gain, making leakage a problem, and handhelds will have a lot of proximity effect.

The Multichannel Sound Field Multichannel systems need to spread be out over a relatively wide angle in most spaces. It's important for the mixer to keep the size of the sound field from any source in scale with the real size of the source. For example, a choir which spreads over the entire width of a worship platform SHOULD have its sound spread across all three channels. But piano or a drum kit, each of which occupies a relatively small space on the platform should generally be localized to a relatively smaller space between two clusters.

I like two condenser mics on most pianos (SM-81's and C414's are a couple of favorites), one above the right hand hammers, the other at the "sweet spot" for the greatest left hand loudness (to minimize leakage). If the piano is to the audience's left, for example, I'll put one mic in the left channel and the other either in the center or somewhere between left and center. And great care is needed in positioning and aiming the mics away from loud instruments and monitors to minimize leakage.

Horn sections (as opposed to individual soloists) can be effectively spread between channels or even all the way across the sound field. Again, the key is maintaining a realistic perspective. Three to five trumpets in a section will work nicely with two or three mics, each mic to its own channel, or partially panned to keep the image in scale. M160's are a wonderful choice for trumpets. And especially with trombones, be careful to use a variable-D mic (RE-11, 16, 20) to keep proximity effect under control during a solo. Sure, you can work with a console's eq to pull lows and low mids out, but you'll go crazy following a soloist as he moves in and out.

Sax sections almost need a mic per horn to minimize leakage from the brass. The best spot for the mic is generally about 6-12 inches above the bell and more or less horizontal, facing the neck, never the bell. If you have five horns and three mics, put one on the lead and one between each pair of players on each side of the lead chair. Sound quality is good there, the only problem is leakage. And don't forget the high pass filter at about 250 Hz!

Console Signal Flow and Other Features It's very important for any console to have an effective solo (cue) buss. We like the solo to be post-fade, post-eq to facilitate troubleshooting leakage problems in a mix during a performance. But it's also important for a solo buss to be able to preview an input without putting it in the mix (cueing a tape or checking a mic are two examples), so switching must be present to let the solo function be either pre-fade or pre-mute. A good solo buss is a great learning tool. I wore out the solo buss switches learning to mix on an old PM-1000!

Likewise, we insist on all aux sends being post eq, regardless of whether they are pre or post fader. Our logic is this. First, modern DSP products make it easy and cheap to provide well equalized main systems and monitor systems downstream of the console. Channel eq can therefore be dedicated to solving problems associated with that input -- i.e., mic placement, mic type, the sound of the electronic instrument being brought in on that channel, etc. rather than being used to tune the system. More important, consider the situation where, for budget reasons, you must mix monitors from the house console. When the star asks you for eq on his or her mic but not on the guitar that's also in their monitor mix, which knob are you going to turn if the monitor send is pre-eq? And if his ears are toasted above 4 KHz, a boost at the equalizer for the monitors will fix his whole mix, but won't screw up the house!

We insist on being able to solo sub-groups in LR pairs and LCR triplets. In other words, if I have the choir in sub-groups 1, 2, and 3, I want to listen to 1, 2, and 3 only, with 1 in my left ear, 3 in my right ear, and 2 in both ears, without hearing groups 4-12. And ideally, I'd like to be able to solo 1, 2, and 3 along with 4, 5, and 6, so I can hear how the choir and the soloists are doing together. And so on.

Noisy Power Supplies Lately, we've been running into console power supplies with noisy fans. We're regularly designing systems for RC 20-25 halls, then seeing folks drag in a mix console with a 40 dBA fan in the power supply, and hearing complaints from patrons seated near the console. If ever there was an application for convection cooling, this is it!

Summing it All Up The most important features in an LCR mix console are:

At least 4 LCR triplet sub-groups
True and LCR and LR panning for each input
A sweepable 2 or 3-pole low cut (high pass filter)
Post fade solo, with the ability to listen pre-fade or pre-mute
Post eq sends, switchable pre and post fade
Solo of each triplet sub-groups
A quiet power supply