Systems for Stereophonic Sound Reinforcement: Performance Criteria, Design Techniques, and Practical Examples

Jim Brown
Audio Systems Group, Inc.
Chicago, IL, 60640 USA
jim@audiosystemsgroup.com

This paper was presented at the 113th AES Convention in Los Angeles, October, 2002.. You can search the complete AES Electronic Library at http://www.aes.org/e-lib/ This paper is available as Preprint 5666.

ABSTRACT

Although stereo systems for large rooms were pioneered in well documented work at Bell Labs in the 1930's [1, 2, 3, 4], most modern practitioners appear to be ignorant of the most important aspects of that work as applied to modern sound reinforcement. This paper draws on the author's experience over nearly twenty years with both portable and permanent systems using two and three front referenced channels. Design criteria and examples are presented to illustrate both good and bad design practices, and some important pitfalls are noted.

INTRODUCTION

It is well known that human perception of the directionality of multi-channel sound (and likewise of a sound arriving directly at a listening position in combination with one or more reflections) is determined by temporal differences between the channels (arrival times), amplitude relationships between the channels (arrival levels), and directional cues provided by the human hearing system.

Many researchers, Haas [5] being the best known but 100 years after the first (Gardner cited nine publications on perception of echoes prior to Haas) [6, 7, 8, 9, 10, 11] have shown that temporal differences resulting from a listener being only 15 cm closer to one source than another of the same signal can often convince the listener that the closest loudspeaker is the only source active, even though the more distant source may be several dB greater in level. This mechanism is known as the precedence effect.

It is generally accepted that amplitude is the least powerful of these mechanisms, and that timing is the most powerful. In fact, temporal differences of only 1 ms can offset amplitude differences of nearly 10 dB. It wasn't until 1981 that Rodgers showed that the human pinnae and ear canal were also capable of providing powerful directional cues. [12]

In most spaces large enough to need a reinforcement system, the arrival time differences for sound from stereo loudspeakers will typically vary throughout audience seating over a range of 0 - 30 ms. Thus, precedence can cause a single microphone that is sent equally to two or more loudspeakers to be perceived by most of the audience as originating from the closest loudspeaker unless some other factors intervene. It is important to understand that this effect is quite strong for relatively small time differences (< 3 ms) but weakens with greater time differences. [13]

What is less well known is that the directional response of human hearing above about 1 kHz provides effective directional cues on the basis of amplitude differences alone, and these cues can overcome or reduce the effects of arrival time differences. Human hearing is far from omnidirectional, especially at high frequencies. The difference in interaural amplitude sensitivity for a sound arriving at the human head varies with direction and frequency between roughly 10 dB and 30 dB between 1 kHz and 15 kHz. Additional cues are provided by the pinnae for broadband sounds that include components above 5 kHz. [12] The relative importance of each of these cues is very dependent on the program material and the direction of arrival.

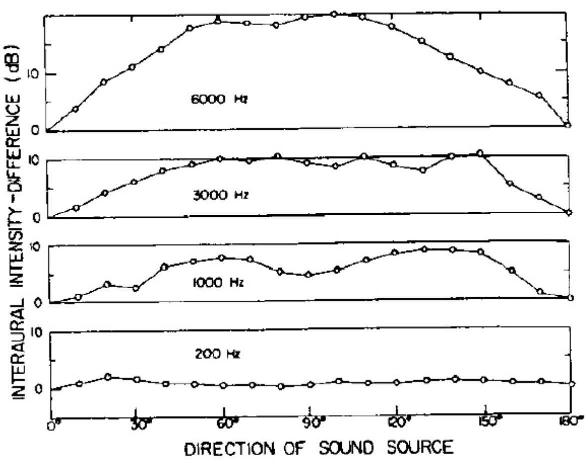

Fig. 1 Interaural intensity difference measured at the two ears as a function of the azimuth of the sound source [14] Reference 15 includes more detailed data.

Once the powerful effects of precedence became widely known, the commonly accepted "wisdom" was that stereo sound reinforcement was neither useful nor practical. But the "wisdom" was wrong. Snow showed how to overcome precedence using a combination of amplitude shading of the loudspeaker coverage of the audience and acoustic delay. [16] Twentieth Century Fox film sound engineer Lorin Grignon, while acknowledging the powerful influence of arrival time cues, noted in 1953 that amplitude differences as small as 2 dB could affect localization [17]. The amplitude shading used alone (i.e., without the delay) is quite effective in countering the effects of precedence. This author, ignorant of Snow's work until doing research for this paper, reinvented it (the amplitude shading) for use in his first stereo reinforcement system (c.a. 1984) and in every subsequent system. In every case, it proved quite effective throughout the audience.

As part of Harvey Fletcher's engineering team at Bell Labs, William. B. Snow filed at least two patent applications in 1936. [16, 18] The first of these, filed in May, noted that some powerful factor other than amplitude differences was causing a shift in perceived direction, but made no mention of time differences. The second application, filed September 30, 1936, showed that he was quite aware that the more powerful factor was arrival time differences, and addressed it with two very innovative solutions to counter the effects of precendence. [16] In a 1954 paper Snow said that perceptual work had been done on precedence as early as 1934 but was published only in the patent [10]. Uzzle speculated that this work was undertaken but not published earlier because Bell Labs eventually intended to develop stereo sound with films, but World War II intervened. [19]

The combination of human hearing directivity, pinnae cues, and the adaptability of human senses to their surroundings is what allows stereo to work in large rooms. For most program material, precedence stops working abruptly (some might say that it "falls off a cliff") when the ratio of delayed to direct sound exceeds about 10 dB. Fig. 1 shows that human head directivity above 3 kHz can provide part or all of that difference; an additional 3-6 dB of amplitude shading can extend the effect below 1 KHz.

Some programs (or parts of programs) will be perceived with greater directional realism than others. For example, time differences are less powerful in localizing sounds that are relatively continuous, like legato components of orchestral or choral music, while they are more powerful for percussive sounds. Directivity of sounds with energy predominant in the octaves between 125 Hz and 1 kHz may be perceived on the basis of interaural time differences, or on the basis of the amplitude cues provided by higher order harmonics, or even on time cues provided by percussive transitions.

All of the human senses, as well as the brain's interpretation of the data they provide, are quite adaptive to their surroundings. We adapt to changes in ambient temperature, the intensity and spectral content of light, the overall loudness and frequency balance of an audio system, and many other environmental conditions. The author submits that we also adjust to the nature and strength of the directional cues provided to us by our acoustical environment and by any audio system operating in and interacting with that environment.

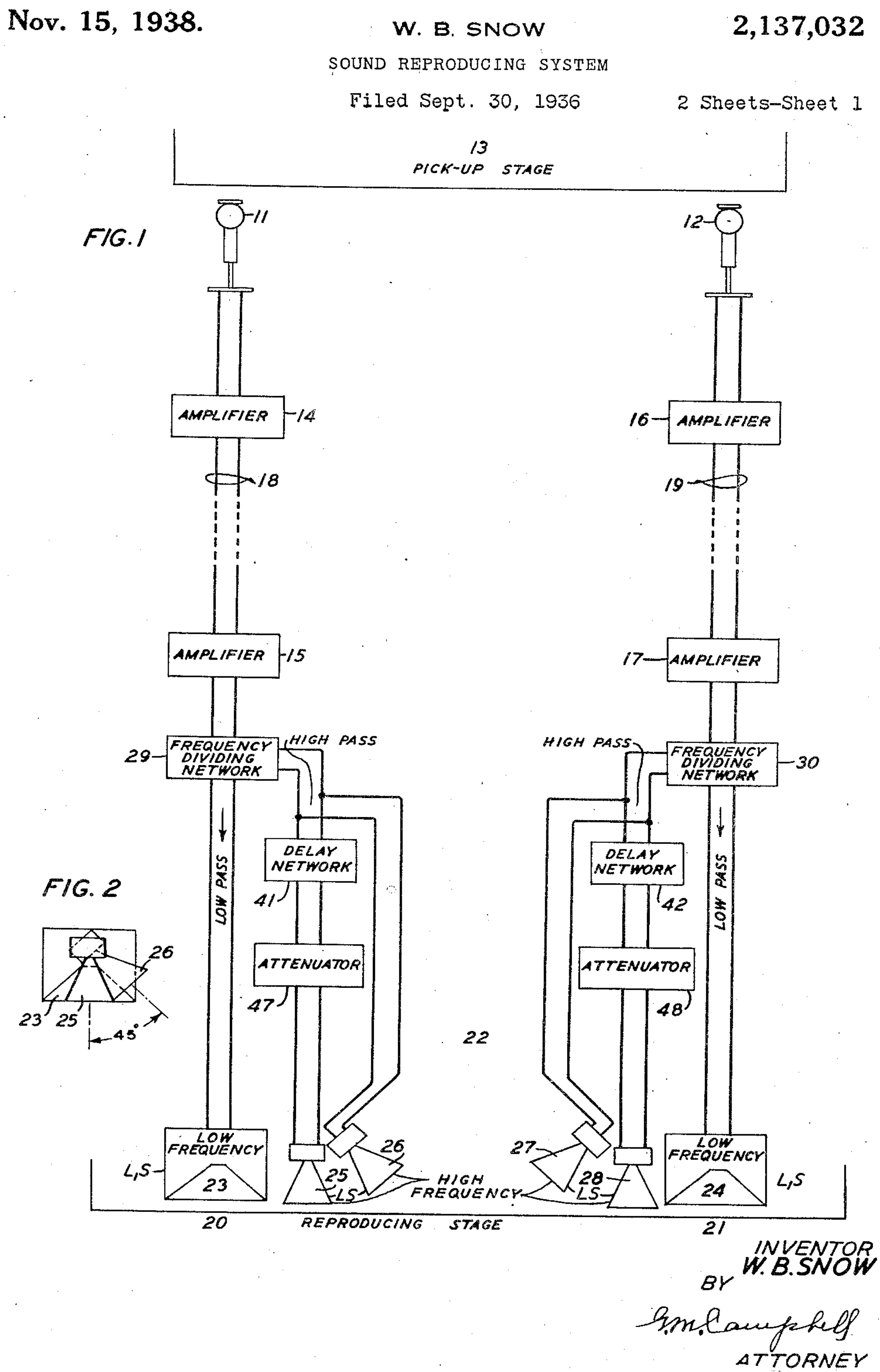

Fig. 2 An Illustration from Snow's Stereo Patent [16]

When a performance is being reinforced at moderate levels, effective precedence cues can also be provided by direct sound from the talker or performer. These cues will generally be effective if the level difference between the reinforced and unreinforced sound does not exceed 10 dB and the spectral balance is well matched.

The practical result is that very effective stereo can be achieved in large rooms but only if the systems are designed according to criteria based on all of these perceptual realities, and program material fed to the system is produced and mixed with those same understandings firmly in hand.

What is Stereo?

Some of its early practitioners thought of stereo as "spatial sound reproduction through multiple channels." Snow defined stereophonic as "a system employing two or more microphones spaced in front of a pickup area, connected by independent amplifying channels to two or more loudspeakers spaced in front of a listening area. This creates the illusion of sounds having direction and depth in the area between the loudspeakers." It "produces an abnormal sound pattern at the listener's ears which his hearing sense interprets as indicating direction in the limited space between the loudspeakers." [13] The key words here are "interpret" and "illusion." The weight of research shows that the interpretation varies from one observer to another, and the illusion need not be a perfect one to be of great value.

An acceptable illusion may be created by as few as two channels; three or more channels can produce a more robust one. There is nothing "magic" about two or three channels. Rather, we came to think of stereo as having two channels because it was practical during the 1950's to manufacture 2-channel phonograph records, and the resulting illusion was acceptable for commercial purposes. On the basis of cost and diminishing returns, Snow came to the conclusion, with which this author concurs, that three front channels is a very practical upper limit for most reinforcement. [13] This does not rule out special cases, and Ahnert has used five and more channels for very large outdoor venues with very wide seating. [20]

Why Stereo?

Strictly accurate directional realism for panned sources is not the only goal of stereo reinforcement, nor is it even the most important one. For example, a choir, horn section, or orchestra performing outdoors or in an inhospitable space like a gymnasium or arena often needs reinforcement. A gospel choir singing with an electronic soul band or rhythm section also requires it. These musical sources can be picked up much more effectively by multiple microphones than by a single microphone, but multiple microphones will combine quite poorly into a single reinforcement channel (what Burroughs [21] called "acoustic phase cancellation" and what is more commonly known today as "comb filtering"). Snow [10] reported results of stereo tests that this author interprets as indicating much less listener fatigue for stereo systems as compared to monophonic ones. So one of the major reasons for using stereo reinforcement is to allow more effective microphone and mixing technique with large sound sources.

Second, stereo systems are able to achieve a higher level of audience satisfaction at lower overall sound pressure levels than a monophonic system of comparable quality. The author estimates this level difference to be on the order of 6 dB. This is also true of design techniques like bass arrays that by improving low frequency directivity [22, 23] provide the greater uniformity of direct sound needed to make stereo work well. With sound levels at musical events commonly exceeding levels at which hearing damage can occur, this is a powerful reason for using stereo!

The third major reason for multi-channel reinforcement is, indeed, directional realism. While it is well known that single-channel panned sources don't image very well because of precedence, sources that are picked up by well placed (and widely spaced) stereo arrays of microphones are much less affected by this problem, because the microphone array provides temporal cues as well as amplitude cues.

Fourth, it has been the author's experience that localization for complex musical and dramatic programs is better than is predicted by studies of single panned sources or echoes. While localization of single panned sources will be pulled toward the closest loudspeaker, a mix of the output of stereo arrays of microphones and multiple panned sources will be perceived as stereo in a very pleasing way if the loudspeaker system is suitably designed for the listening space. Listeners far off centerline -- i.e., much closer to one channel than the other(s) -- will, indeed, perceive a stereo image of panned sources that is narrower than will be perceived by listeners seated near centerline, and that image will be centered somewhere between the center and the closest loudspeaker. But -- and this is critically important -- the perceived quality of a good stereo mix through a good stereo system will be much higher than for a good monophonic mix though an equivalent monophonic system, even for off-axis listeners. [17, 24]

Fifth, lower overall sound levels can significantly reduce problems with acoustical feedback.

Design Criteria

If stereo reinforcement is to work well, several basic requirements must be met.

1. Each listener must hear each channel well, but not at exactly the same level. In fact, the perception of stereo in off-centerline seats is greatly improved if the level from the more distant channel is a bit higher than for the closer one. <u>In other words, "amplitude shading" of loudspeaker coverage from one side of the audience to the other can partially compensate for a listener being closer to one loudspeaker than another, and is critical to good stereo perception in off-center seats</u>.

2. Each listener must hear adjacent channels in a time relationship that satisfies the criteria for good speech intelligibility. With good ratios of direct to reverberant sound, good results have been obtained if cleanly defined direct sound from adjacent channels is within 25 ms and there are no strong echoes. Speech intelligibility will start to degrade when arrivals spread beyond 30 ms. Ahnert says that this interval can be extended to 40 ms, and has done so for very large outdoor systems. [20]

3. Each listener should receive direct sound from each channel with the minimum practical time differences between channels.

4. Each listener should receive direct sound from each channel with essentially the same spectral response. In other words, the frequency response of each channel should be uniform over the audience.

5. The total direct sound pressure level received from all channels combined should be as nearly uniform as practical over the audience. In practice, it is usually possible to achieve equal levels, ±3 dB, throughout the audience, with levels being highest near centerline and lowest in front corners of the audience (i.e., near the stage).

6. The venue should not be excessively reverberant, and the loudspeaker system should minimize the spill of sound to the perimeter walls, ceiling, and the reverberant field.

Some question the use of left to right amplitude shading in a stereo system because they say it causes those seated most distant from centerline to hear a less well balanced sound mix. Those off centerline do receive an unbalanced mix of <u>direct</u> sound from a good stereo system -- but so does an audience in those same seats listening to an unamplified performance! In the case of the unamplified performance, the shading is caused by inverse square law, room geometry, source directivity, and other acousticalal parameters. Moreover, those level differences for the acoustical performance are generally most pronounced at the front of the audience and least so at the rear.

Not all seats for an unamplified performance are equally good. It would be nice if they were, but they aren't. Some have a more optimum blend of direct and reverberant sound, and a better balance of the orchestra (or actors), than others. Some seats in a cinema are far less than ideal for viewing the screen. Certainly it should be a design objective to make all of the seats as good as possible consistent with budgetary and physical realities, but it's silly to say that the balance can't be compromised a bit in a few seats to make the overall sound quality much better in the vast majority of seats.

It is also important to realize that reverberation in most large rooms causes overall sound levels to be much more uniform from seat to seat than one would expect when considering direct sound only. This is a significant factor even in spaces with relatively low levels of reverberation. Thus the reverberant field permits every listener to hear the entire mix, even though amplitude shading of the direct sound is providing directional cues! The author's experience has been that well designed systems using amplitude shading do result in good perceived balance for a well mixed program throughout an audience, even in front corner seats!

Eargle quotes Snow as saying that the delay element of his patent was not very practical. [25] But to put this in context, it should be remembered that during Snow's lifetime (he died in 1968) delay could only be achieved by means of a tape loop or the method described by the patent, a long extension tube between the horn and driver. (When you had Wente and Thuras on your team and Bell Labs to sign the checks, you could have anything you wanted in the way of transducers!) It is well known today, however, that delays on the order of 2-8 ms between adjacent loudspeakers can be quite useful over certain frequency ranges. Coffeen pioneered this technique c.a. 1986 to minimize audible interference between adjacent high frequency elements of a loudspeaker cluster, [26] and the method was subsequently published by Mochimaru. [27] While it is certainly possible to do audible damage to the waveform with misalignment, Snow's delay concept clearly deserves a second look.

None of this is new, and only this analysis originated with this author. Indeed, Snow and Fletcher are the giants upon whose shoulders we stand. Amplitude shading has subsequently been found effective in small rooms [28, 29, 30, 31, 32, 33, 34] and Kates continues to pursue it. [35] Myati and Aoki applied both amplitude shading and delay shading to small rooms in 1984, but seemed unaware of Snow. [36] Holman notes the usefulness of a mild form of amplitude shading in cinema systems. [37]

A Simple System

A simple 2-channel system for a small theater illustrates basic concepts. Audience seating is about 20 m wide by 12 m deep, and there is a small balcony. The loudspeaker system is a single spaced pair, suspended 8 m above the stage floor, just downstage of the proscenium, and at a spacing of 12 m. [Note: In theatrical terms, "downstage" is in the direction of the audience, "upstage" is away from the audience; the terms carry over from the early days of theater when some stage floors sloped downward from upstage to downstage. Throughout this paper, left and right directions will be referenced to the perspective of the audience, not to theatrical stage directions, the "front" of the audience refers to those seats closest to the stage, and the "rear" of the audience refers to those seats more distant from the stage.]

The full range loudspeakers, having nominal coverage of 90 degrees horizontal by 40 degrees vertical, are cross-aimed. In this simple system, the audience left loudspeaker is aimed to the right-most seat in the last row of the audience. In a larger system where delayed loudspeakers cover the rear of the audience, the front loudspeakers will be even more severely cross-aimed to seats much closer to the stage.

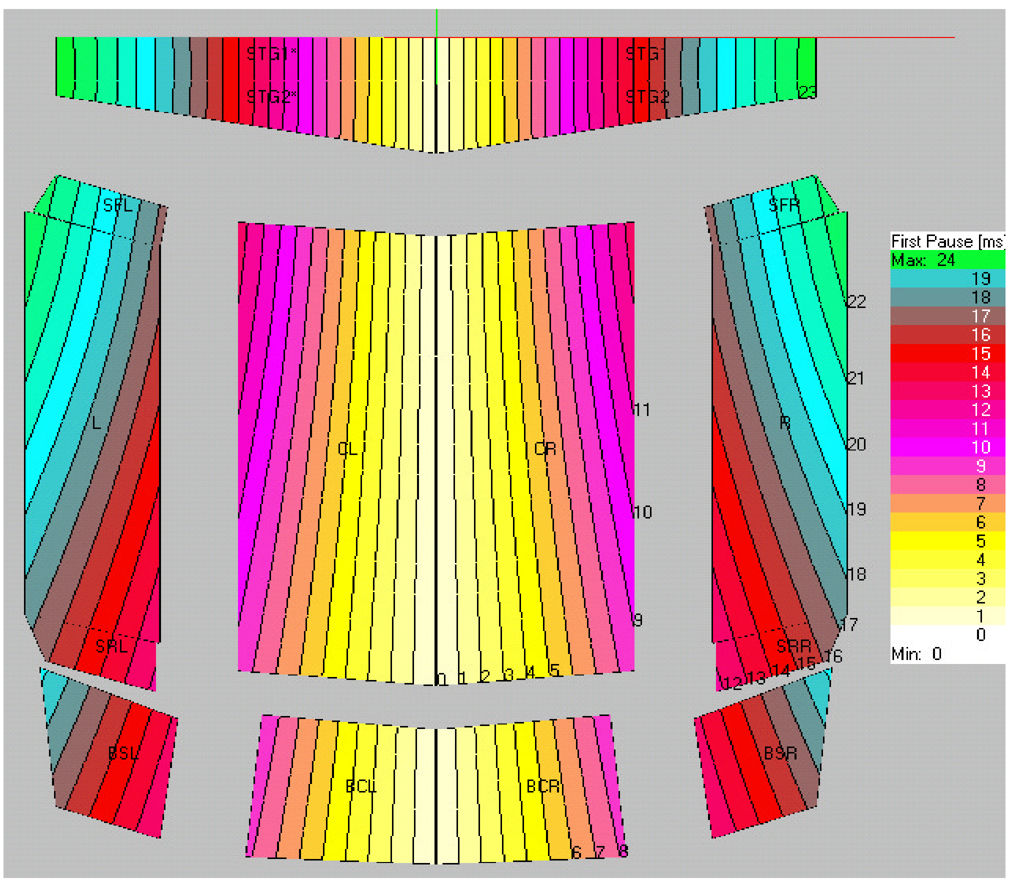

Fig 3. The EASE prediction of the difference in arrival times for left and right channels for a simple stereo system in a small theater. The greatest differences in arrival times occur at the front corners of the audience, the smallest differences are at the rear of the audience. Contours are at 1 ms increments.

Near the rear of the audience, both arrival time differences and level differences are relatively small. Moving closer to the stage, level differences are purposely increased to compensate for greater differences in arrival times. A good system design will maintain these level relationships over the widest practical frequency range by using loudspeakers with well controlled (and more uniform) directivity versus frequency.

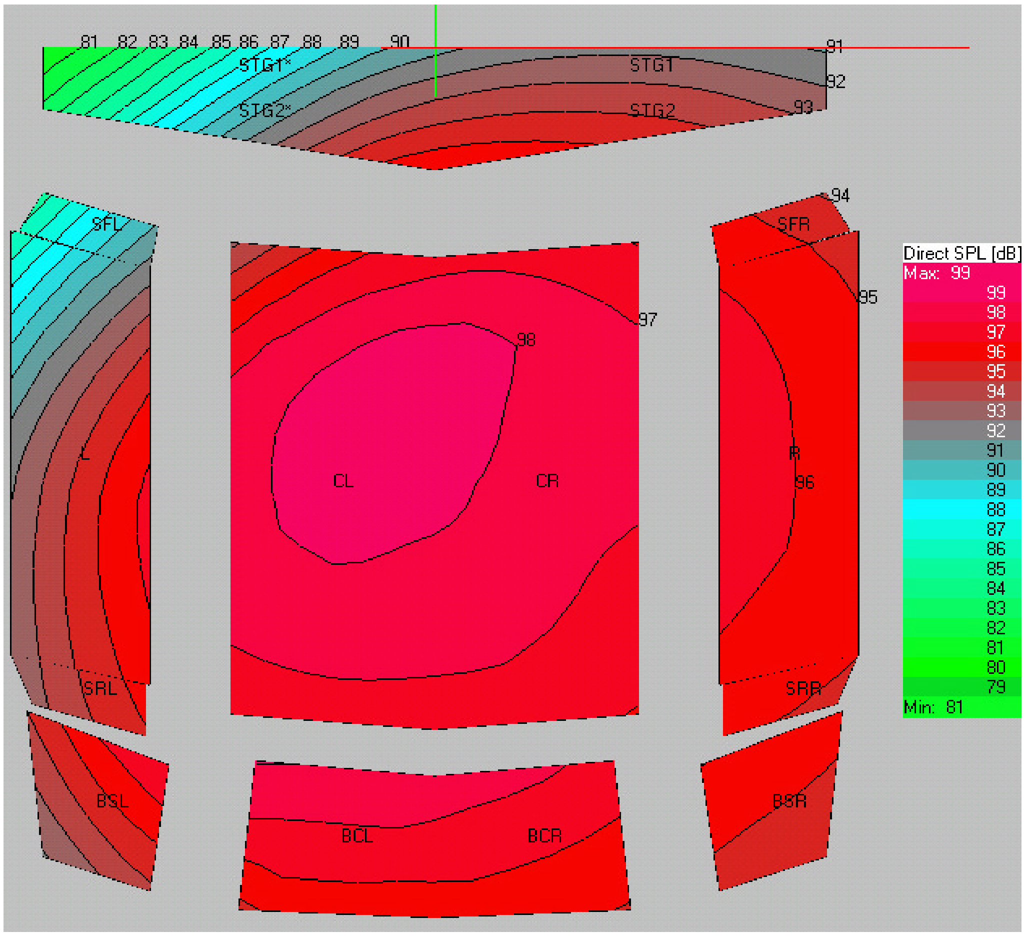

Fig. 4. The EASE prediction of direct sound for the left channel only. Contours are at 1 dB increments. The five sided audience area at the top of the plot is the forestage.

Deep Rooms And Delayed Loudspeakers

In many rooms, it is either undesirable or impractical to cover the entire depth of audience seating from front clusters. Some reasons for not covering all seating from the front are that 1) doing so might require that the loudspeaker system be larger than the available space, 2) the loudspeaker system might be so large that it interferes with lighting or staging, 3) the loudspeakers might spill from to rear and side walls at an angle near perpendicular, with resulting long delayed echoes near the front of the audience, and 4) seating at the rear of an audience is sometimes shadowed by an overhanging balcony. There are several possible approaches to supplementing the front loudspeaker system.

The cleanest and most obvious solution for supplementary loudspeakers in a two-channel system is simply a delayed pair (or an LCR triplet for a three-channel system) further back in the hall. This is usually the best solution when room geometry and architecture allow good coverage from acceptable loudspeaker locations. It is quite important, however, that delays be set to the absolute minimums that provide good precedence. Because there are both left and right main loudspeakers and left and right delayed loudspeakers, it's possible for intelligibility to be degraded if the interval between the first and last arrivals is too long.

When the area needing delayed coverage is reached by both left and right clusters but only needs about 3 dB more level, a monophonic delay fed by the sum of left and right can be effective. The key to success is that the level of the monophonic signal in the coverage area should be no greater than the level of the signal provided by the left and right channels. The monophonic signal should be delayed so that it arrives just slightly later than the discrete left and right channels, so that the latter always provide precedence. Thus, while the monophonic signal will be degraded by comb filtering of spaced pairs of microphones (but not panned mics), the listener will always be hearing discrete left and right at a level which is equal to or greater than the monophonic signal. Perceptually, the resulting sound quality is quite satisfactory, and the additional level provided by the monophonic delay maintains good "presence" in seats that would otherwise be a bit lacking in direct sound. Thus, although a monophonic delayed system is usually not as good as a full stereo delay system, not all room geometries or budgets permit a full stereo delay, and the monophonic delay system is better than no delay at all.

Under-Balcony Systems

The shadow caused by a balcony is not sharply defined, but takes effect gradually and is most pronounced at high frequencies. Low frequencies diffract around the balcony face. On the other hand, seating under a balcony, especially with a low underbalcony ceiling, is deprived of some of the reverberant energy received by the rest of the audience.

Full stereo from the main system can easily be achieved near the front of underbalcony seating by direct coverage from discrete left and right channels. Even in the shadow area, some diffracted direct sound will provide precedence for a row or two. Deeper under the balcony there are several possible approaches. The simplest is a monophonic delay, fed by the sum of left and right. This works pretty well perceptually if there is direct sound from discrete left and right to sum with the monophonic delayed signal. Beyond that range, however, the monophonic delay sounds good for speech and panned mics, but doesn't sound as good for the parts of a music mix generated by spaced stereo pairs.

The second solution is a horizontal line of loudspeakers spaced relatively close together and facing the back rows of the audience. The loudspeakers are alternately fed left or right signals, so that all listeners hear both left and right channels, but half the listeners hear an inverted right left/right perspective. This solves the comb filtering problem with spaced pairs into a monophonic delayed system, and the result is generally pretty acceptable for music.

A third solution may be practical if there is sufficient ceiling height in the area to be covered and the seating area isn't too wide or is split into somewhat isolated segments (for example, two sections on either side of a booth that's on centerline). It can also be the most costly option. For this solution, each seating section is covered by a delayed left/right pair of loudspeakers cross-aimed into each such seating area. With this option, each listener in the delay zone hears stereo with the correct left/right perspective.

Wide Rooms and the Number of Channels

It can be much more difficult to achieve good stereo coverage in a very wide room. The reasons are twofold. First, the greater width causes arrival time differences between channels to be greater in seats at greater angles to centerline. Second, it is much more difficult to achieve acceptable amplitude shading from a single cluster location without hot spots (i.e., areas of the audience that receive too much level), and without loudspeaker clusters getting larger. Again, the front corners of the audience are the most difficult to cover, and even more distant seating at the extremes of a fan shaped room can be problematic.

Snow’s rule of thumb for loudspeaker spacing and the number of channels needed to cover a given stage width was to add a channel when the spacing exceeded about 8 m [13]. It is likely that the basis of his “rule” was an array of loudspeakers that were not much above the level of the audience (that is, behind a projection screen). This compares quite well to this author’s time of arrival criteria (< 25-30 ms between adjacent channels).

Much wider physical spacing can be used if the loudspeakers can be sufficiently elevated above the audience and still remain within the time of arrival criteria. Height cannot, of course, be increased without encountering other acoustical limits. The total delay, including the system’s electronic delay (latency) and the time of flight between performers and the loudspeakers that reach the performer cannot be so great that an echo is perceived. Long delayed echoes are well known to degrade performance by causing fatigue or interfering with tempo.

The strength of the echo in the performance area can be reduced by using loudspeakers with sufficient directivity and locating them well. The echo problem can also be helped by providing undelayed (or slightly delayed) foldback to the performer, thus extending the fusion time. [38, 39, 40] Tappan used delay on a nightclub foldback system in 1968. [41]

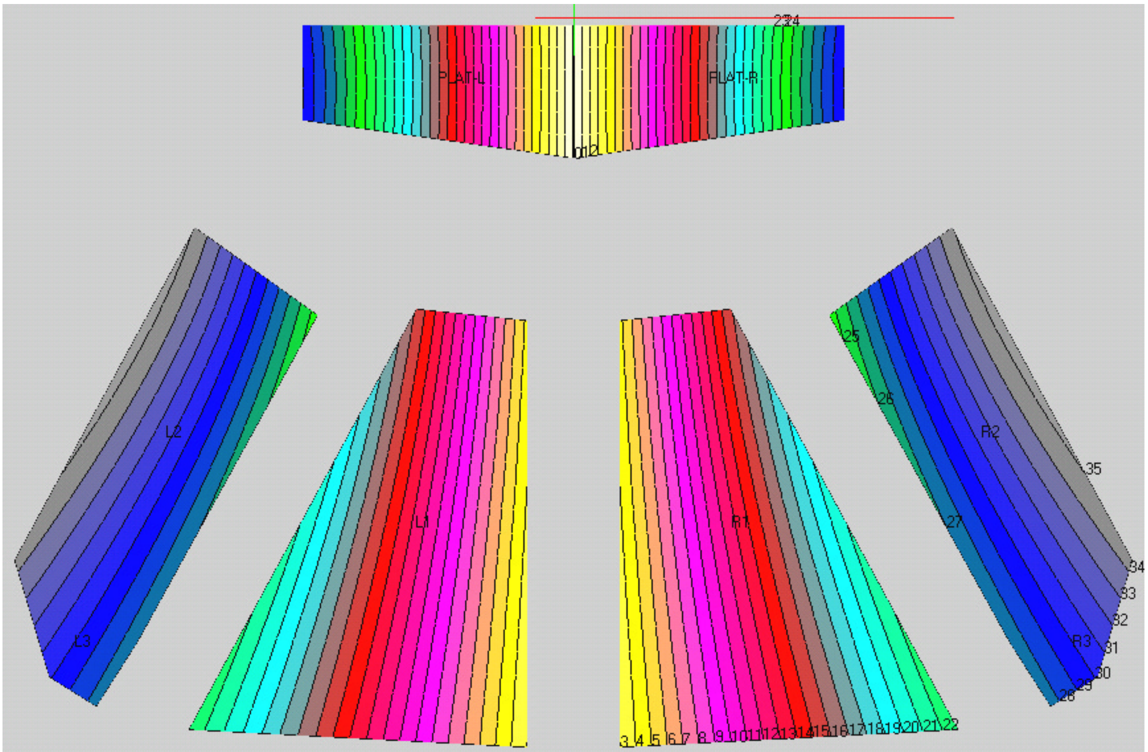

Fig 4. Predicted arrival time differences for left and right front clusters in a wide room with a relatively low ceiling (9 m above the stage). Contours of equal delay are at 1 ms increments, and the criteria for arrival times is not satisfied in much of the side seating areas.

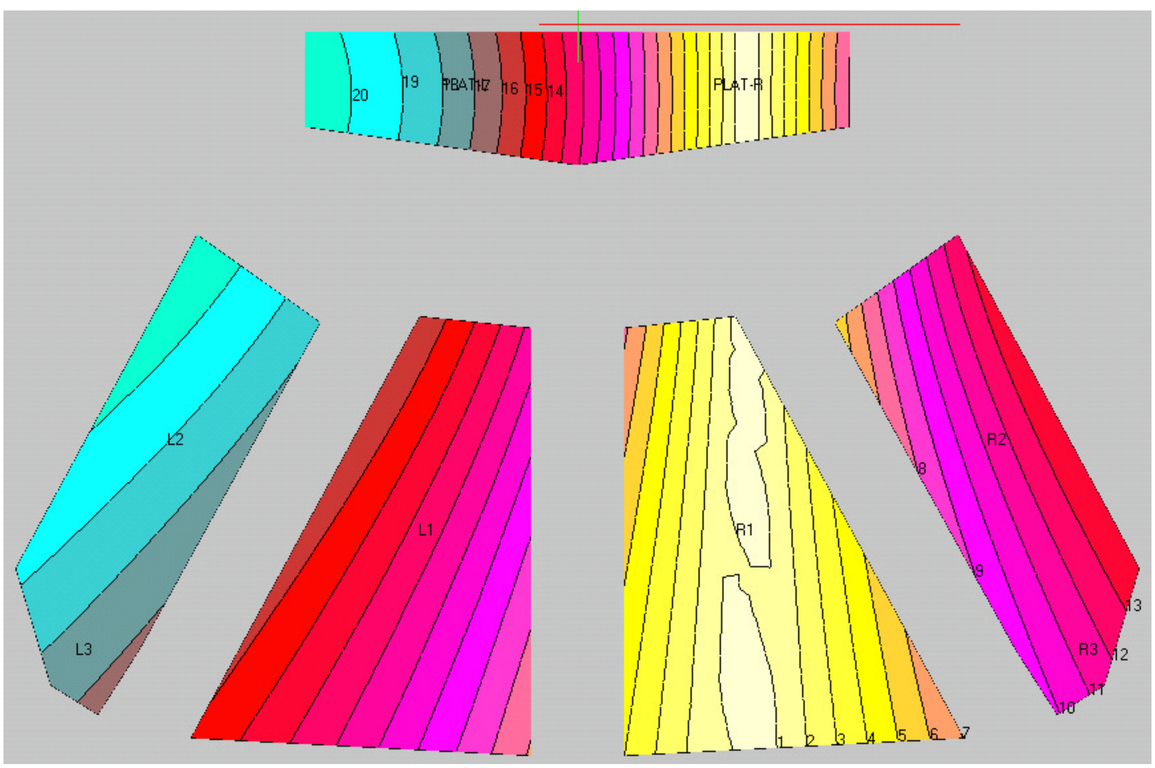

Fig 5. Predicted arrival time differences for right and center front clusters in the same room and at the same height as Fig 4. Contours of equal delay are at 1 ms increments, and the criteria for arrival times is met in all seats.

The loudspeaker system can also be placed too high for the audience. While our ability to localize sound in the vertical plane is much less powerful than the horizontal plane, a cluster that is too high will pull the image up if there is not sufficient precedence from the performance itself. Also, if the arrival time differences are too great, the direct sound from the performance will cause annoyance as a “pre-echo.”

Both of these effects can be countered by adding loudspeakers to locally reinforce the performance at the level of the stage. A common implementation is a horizontal line array of very small loudspeakers built into the lip of a stage or the steps around an altar rail, usually with some delay (and sometimes with delay in segments). This technique is also not without cost, since it can degrade stereo localization in the audience zone that the supplementary loudspeakers serve if not done quite carefully.

A far better solution lies in some variation of that described by Ahnert [42], where multiple loudspeakers are built into architecture or a theatrical set and used with complex panning and routing to increase the level of the precedence signal but with greater geometric accuracy. For example, it is often quite practical to integrate a precedence source into a pulpit or lectern.

Room Geometry Issues - Loudspeaker Height and Seating Width

The room geometry factors that most seriously limit our ability to design stereo systems are wide seating areas and limited height for loudspeakers. If the loudspeakers are too low, it can be difficult, expensive, or totally impractical to meet the design criteria for arrival times, amplitude distribution, and uniformity of frequency response. A rule of thumb that the loudspeaker height should be greater than one half the spacing between adjacent loudspeakers is not a bad starting point, but is far from a definitive answer.

The only practical way to know how well a stereo system will work in any given venue (short of setting up the proposed system itself and doing a lot of listening) is to develop and study an acoustical model of the system using modeling software such as EASE. The author would not consider designing a sound system for any venue without such a model; the model is far more critical with stereo because of the complex amplitude and time criteria that must be met. Only when the proposed loudspeakers are modeled at the proposed locations will the designer know if an acceptable system is practical, how many channels are needed, what delayed loudspeakers are needed, and how well the resulting system will work.

Wide Rooms and Delayed Crossfeeds

One workable solution for a wide room is to add delayed loudspeakers to serve these "wide" seats. For example, a loudspeaker might be added near centerline to provide right channel coverage to the left front corner of the audience, and another for the left channel to cover the right front corner. The added loudspeakers would need to be delayed enough so that they didn't degrade the image in front row center seats. This technique, called a delayed cross-feed, has been widely used in both two and three channel systems, and can work well if properly applied.

One serious misapplication of delayed crossfeeds [43] is to use a loudspeaker that is already part of one cluster to carry both its own channel and delayed signals from another channel. Thus, certain elements of the left cluster carry the left signal undelayed, the center channel delayed by one increment, and the right channel delayed by another increment. The reason this is a bad idea is simple. In producing a mix, microphones often need to be panned between channels, and any panned signal will thus be present in two channels. When that happens, a loudspeaker with a delayed crossfeed will carry both a delayed and undelayed copy of all panned signals. Haas [5] noted that when sound and echo are emitted from the same loudspeaker, “intense distortion of speech results for delay differences up to approximately 20 ms.” We now call this distortion “flanging.” It's useful as a special effect on a guitar, but it isn't something you'd like to listen to on an entire mix of a performance!

Some time differences are more problematic than others because they define the frequency range in which the broadest cancellations occur [5, 21, 44]. Thus, the time offsets and the small differences in level that occur due to path differences reduce the severity of what Burroughs [21] called "acoustic phase cancellation," and what others have variously called "phasing," "flanging," and "comb filtering." Breshears recognized the poor sound quality that resulted from delayed crossfeeds in the same loudspeaker, but concluded that the only solution was never to pan microphones. [45] This seriously limits the usefulness of stereo systems. Designing systems without this limitation is the only acceptable solution.

Comb Filtering Between Horizontally Displaced Loudspeakers

Breshears based his mixing and design criteria on the incorrect assumption that a loudspeaker or horn added to a cluster and fed a delayed signal would introduce just as much comb filtering and flanging as if the signals were summed electrically and fed to one or more loudspeakers. Don Davis has emphasized the importance of avoiding comb filtering between horizontally displaced loudspeakers covering the same seats, especially split clusters. Both of these assumptions are simply incorrect.

There is considerable experience, particularly in live theater, with two closely spaced microphones (usually worn by actors playing a scene quite close to each other such that each microphone picks up both actors) being fed into two closely spaced loudspeakers, one microphone to each loudspeaker, but not to both. It is generally accepted that this technique provides a better result than combining the two microphones into a single reinforcement channel, and the technique is widely used on Broadway where it is commonly known as an A/B system.

Haas noted [5] that "interferences were not perceived in binaural hearing of sound and echo from two loudspeakers, because the distance between the two ears makes us always hear at two different points in the sound field," and also because of "the shielding effect of the human head and body" (that is, the directivity of human hearing). Because two sound arrivals must be nearly equal in amplitude for significant phase cancellation (comb filtering) to occur, that directivity essentially eliminates comb filtering above a few kHz. And below that frequency range, the relatively small phase difference between left and right ears not only causes little cancellation to occur but also causes in-phase summation!

This can be understood intuitively by the following. When two delayed copies of a signal having equal level combine in a single electrical circuit, the time and level relationships between them is constant. If instead they are fed to two different transducers and traverse slightly different acoustical paths to the listener, their magnitudes and phase are modified by virtue of any slight differences in the acoustical path and the directivity of the devices. Maximum degradation from the summing of two copies of the same signal having different delays occurs when components of the two signals cancel each other because they are precisely equal and precisely 180 degrees out of phase. Relatively small differences in level will greatly reduce the depth of cancellations.

A simple series of experiments clearly demonstrates these principles. Experiment 1: Two loudspeakers separated by a distance on the order of 2 meters and at approximately the ear height of a standing observer are fed the same pink noise to generate a moderate sound pressure level. The observer stands about 2 meters in front of the loudspeakers, midway between them, and faces the geometric midpoint between the two loudspeakers. The observer then moves from side to side while continuing to face the midpoint. For this experiment, the left ear is toward the left loudspeaker and right ear toward the right loudspeaker. Mild acoustic phase cancellation will be heard at frequencies below about 1 KHz. Replace the test signal with speech and music and repeat.

Experiment 2: The loudspeakers are set up and driven in the same manner, but the observer turns 90 degrees so that one ear is directed to the geometric center of the two loudspeakers and moves slowly back and forth either side of the midpoint. Strong acoustic phase cancellation will be heard. Replace the test signal with speech and music and repeat.

Experiment 3: The setup and observation are the same as for Experiments #1 and #2, except that the drive to one loudspeaker is delayed by intervals between 1 and 30 ms. The strength and character of the phase cancellation will vary with delay and position. Repeat the test with speech and music.

Experiment 4: The setup and observation are the same as for Experiments #1 and 2, except that both loudspeakers are fed an equal electrical mix of undelayed sound and sound that has been delayed by the same varying intervals as experiment 3. Flanging will be heard, the severity of which depends on the delay between the two signals, added to the acoustic phase cancellation observed for Experiments #1 and #2.

These understandings can be used to great advantage in the design of stereo loudspeaker systems, especially when adding loudspeakers to fill areas to the far left and right of the audience. If there's already a center channel cluster, it makes sense to separate the delayed crossfeed loudspeakers from this cluster so that head directivity can reduce comb filtering between the crossfeed loudspeakers and the center cluster. It also makes it easier to fit these loudspeakers into the area over the stage.

A More Complex Design Example

The room of Figures 4 and 5 is the basis of a second design example. Audience seating is nearly 30 m wide but only 16 m deep, and the ceiling is at 9 m. Rear and side walls are acoustically dead. As shown by Figure 4, the room needs to be served by three channels. From study of the EASE model it was determined that one suitable system could consist of one 90°x 40° loudspeaker and one 40°x 40° delayed loudspeaker for the left and right channels, and two 60°x 40° loudspeakers for the center channel. All loudspeakers are along a line just in front of the proscenium, 8 m above the stage.

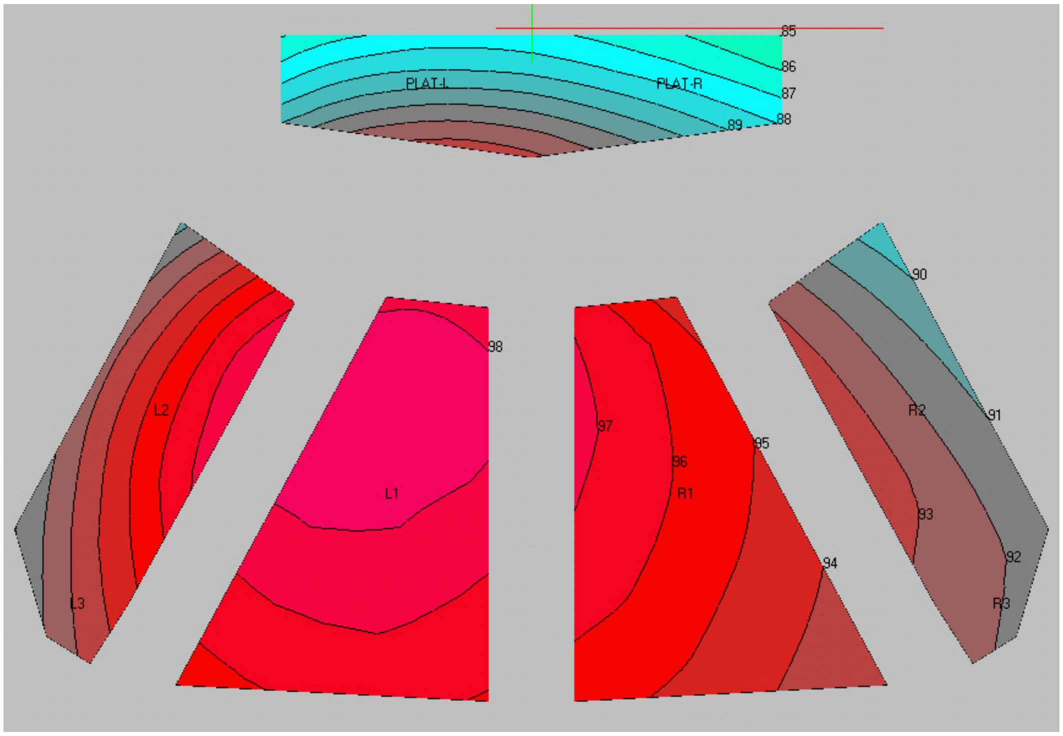

Fig 6. Predicted direct sound from a single 90°x 40° loudspeaker, 9 m left of center, aimed to the center rear of the right center seating in the room of Figs. 4 and 5.

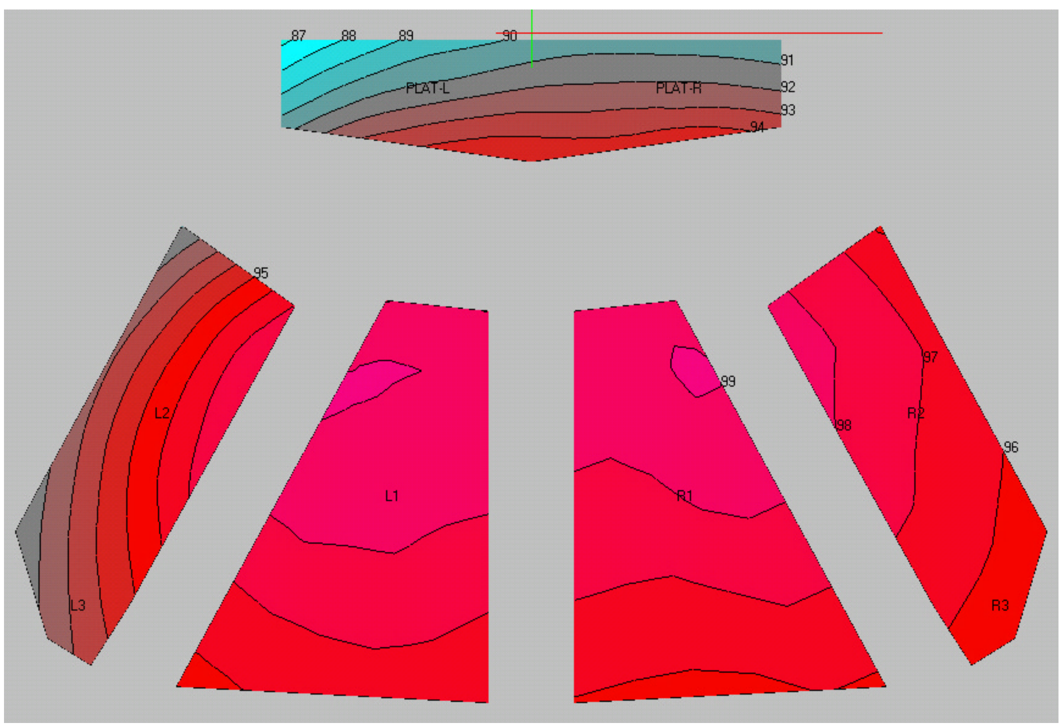

Fig 7. Predicted direct sound for a loudspeaker delayed by 18 ms added to the left channel loudspeaker of Fig. 6. Contours of equal level are at 1 dB increments. The delayed loudspeaker is 1.5 m to the left of centerline, is aimed to the right rear corner of the audience, and is operating at 3 dB below the level of the main left loudspeaker.

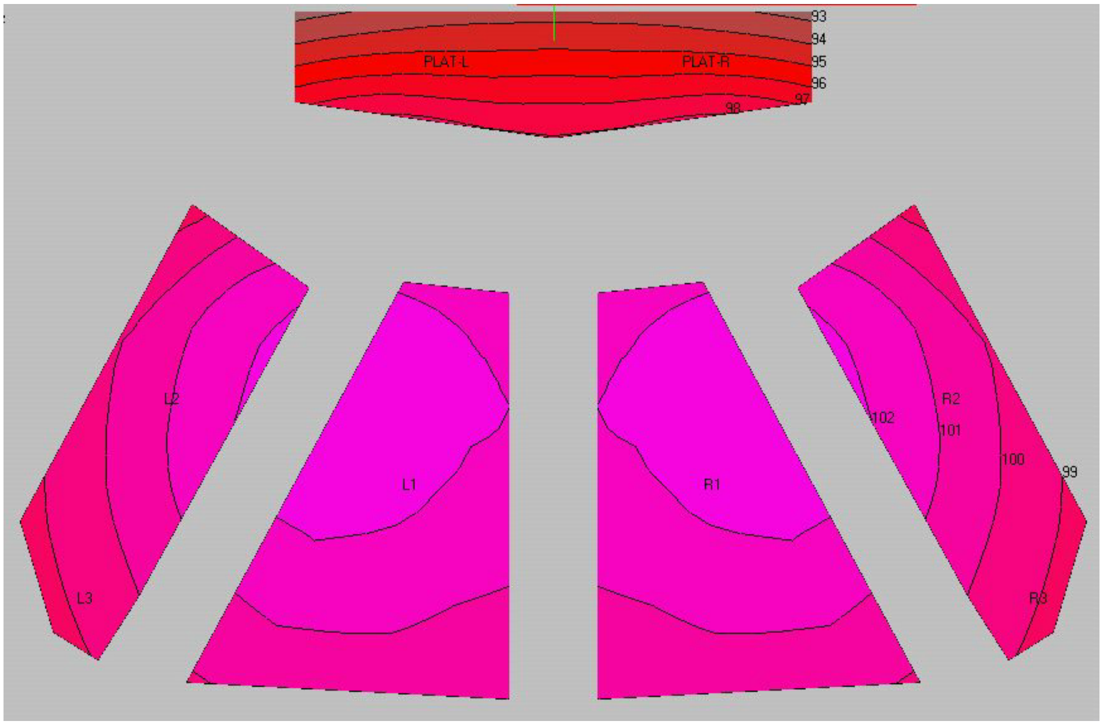

Fig 8. Predicted direct sound for all three channels, including delayed loudspeakers. Contours of equal level are 1 dB increments.

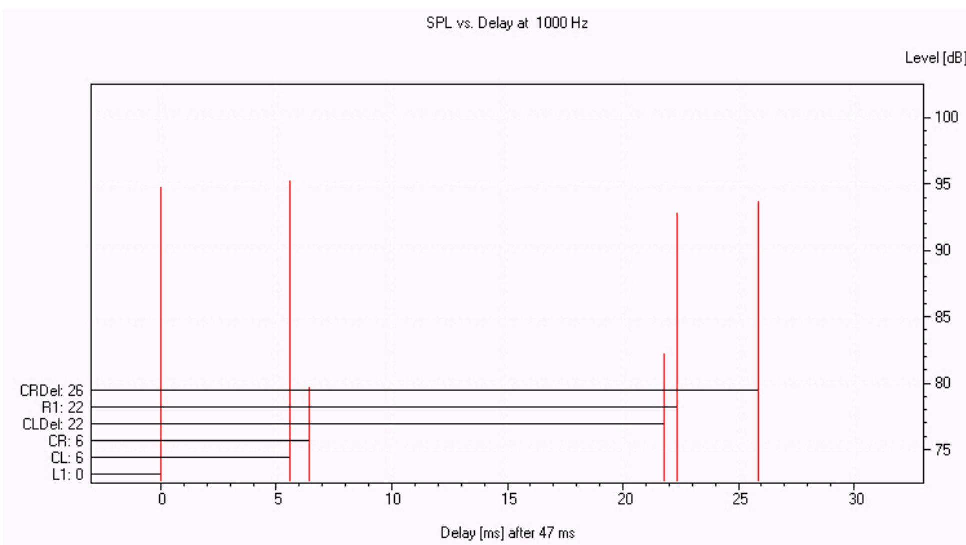

Fig 9 EASE prediction of direct sound arrivals at a seat near the left rear corner of the left center seating bank. A study of the results of calculations like this one is required for seats throughout the audience to verify that the design meets the criteria for good stereo.

This example system takes advantage of head directivity to minimize comb filtering between the main and delayed side channel loudspeakers. Figure 7 shows some hot spots near the front center of the audience, and an analysis of displays like Figure 9 for those locations indicates that criteria for good stereo are not being met. The next design iteration would likely move the delayed loudspeaker downstage a bit so that it is over the audience (assuming that such a location was practical in this venue).

The Center Channel

A center channel has three primary purposes. The first, and most obvious is to provide a hard center image for a "star" or for dialog. Program material panned to the center feeds only the center cluster, as opposed to a left/right signal panned equally to left and right channels of a two channel system. There's no component of this signal in the side clusters to pull the image to the closest loudspeaker.

The second primary purpose is to optimize intelligibility in acoustically difficult spaces by providing a dedicated channel for speech, and by reducing time smear as opposed to a two-channel system. This is important in relatively reverberant spaces, but is usually not significant in the relatively dead spaces being designed for theater and contemporary worship. And if it is a factor, careful design of stereo systems can minimize any loss of intelligibility.

The third primary purpose of a center channel is to reduce the time difference between adjacent panned signals, especially in very wide rooms. Some years ago, this author proposed a protocol for left/center/right panning for reinforcement whereby a signal would be panned from left to center with nothing in the right channel, from center to right with nothing in the left channel, and to the center with nothing in left or right channels [46]. The protocol was subsequently described by Gerzon, [47] who went on to develop an innovative mathematical description of the “law” of the pan pots used to produce the pan. The use of this protocol is critical to the success of 3-channel stereo systems on large rooms.

From the point of view of perception, it is always desirable to have a center channel in a stereo system, but how important it is depends on a number of complex factors, and not all venues can afford a three channel system. So the questions are, when faced with a budget constraint, which gets eliminated from the project first – the left/right channels or the center channel, and when do we need three channels? The answers lies in the project budget, the uses of the audio system, room geometry, and architecture.

A lecture hall or traditional church that only reinforces speech and never needs music playback or reinforcement certainly doesn’t need any form of stereo, and neither does a high school gym that’s used only for sports. On the other hand, virtually any theatrical or music performance space can benefit greatly from stereo, and so can a church that uses contemporary music in their worship.

Cost Issues

One of the major fallacies about stereo reinforcement systems is that they are always much more expensive than monophonic systems. In this author’s experience, this is rarely true except in very wide rooms or those where loudspeakers must be at relatively low elevations over the audience, and even then the differential is not that great [22]. There are several reasons why this is so.

First, two-channel systems rarely require many more loudspeakers than monophonic systems. The number of loudspeakers required is usually driven by two factors -- the sound levels required and the coverage angles needed to cover the audience. A left or right channel cluster is farther from the audience, so fewer loudspeakers are usually required, per channel, than for a central cluster to achieve the desired coverage.

Second, when loudspeakers are added to increase the overall sound pressure level, it makes little difference whether they are added to a monophonic or stereo system.

Third, since a stereo system doesn’t have to run as loud as a monophonic system to achieve the excitement and overall level of satisfaction, [22] the loudspeaker systems and their support electronics can be reduced in size.

Fourth, the loudspeaker system and its support electronics are generally a relatively small part of most permanent installations. While 3-channel mix consoles are an extra cost item and there are few good choices in the marketplace, even the lowest cost mix consoles provide 2-channel stereo main outputs and 2-channel stereo mix groups.

In the author’s experience, properly configuring a reinforcement system for 2-channel stereo typically adds between 0% and 25% to the cost of the main reinforcement system alone, but has no impact on other system costs. That is, there is no increase in the cost of mixing, mics, tie lines, intercom systems, surround effects systems, stage monitor systems, dressing room systems, power, conduit, system grounding, the mix position, etc. In fact the author has encountered facilities where a 2-channel reinforcement system is actually less costly than an equivalent monophonic system.

Three-channel systems are another story. In effect, both a 2-channel and monophonic system must be installed, and the mix console will be more expensive. In the author’s experience, the cost of a 3-channel system is typically 50% greater than the cost of an equivalent monophonic system for the reinforcement component and the mixing.

For a typical project, the main reinforcement system and the mix console account for 25-35% of the project budget. Thus a 2-channel system typically increases total project cost by 0-10%, and a 3-channel system increases it by 20-35%. These cost differentials could be reduced significantly if there were more good choices in lower cost Left-Center-Right (LCR) mix consoles. In these examples, it is assumed that the stereo system can operate at 6 dB lower sound pressure levels than the monophonic system.

Loudspeaker Directivity for Stereo Systems

In general, "pile it up (or fly enough) 'til it's loud enough" arrays of loudspeakers are unlikely to work well for stereo. To be effective in a stereo system, a loudspeaker must have very good and relatively narrow pattern control in the vertical plane, and it should maintain that pattern control over the widest practical frequency range. A loudspeaker that has vertical coverage that is too broad will generate "hot spots" in the center of the audience and be unable to establish the careful amplitude shading needed for stereo. This means that loudspeakers should generally have relatively constant directivity versus frequency.

The author finds that 40 degrees of symmetrical vertical coverage is usually a practical maximum for a loudspeaker when used as an element of a stereo cluster. On the other hand, loudspeakers and cluster elements with asymmetrical coverage patterns can be very useful building blocks. When the loudspeaker's directivity is asymmetrical and carefully controlled so that radiation at lower angles is lower in level, effective vertical coverage angles of 50 degrees or more can be practical.

Microphone Technique for Large Room Stereo

Much work has been done in the last half of the 20^{th} century to develop and understand techniques for 2-channel stereo recording. Until the advent of surround sound for home listening, almost all of this work has been based on the assumption that the listener will be in a small room and centered between two relatively small loudspeakers, and much of it has focused on symphonic music. Almost none of it has addressed listeners in large rooms or contemporary music.

In the discussion of perceptual issues, it was noted that very different production criteria and techniques must apply to program material intended for presentation in large rooms as compared to home listening rooms. Holman recognized this by using mixing theaters to replace control rooms for the production mixing of cinema sound tracks. [48] Grignon took it into account in the 1948 work he was doing to develop stereo microphone techniques for cinema [17]. Some of those techniques have survived to this day. Gerzon acknowledged this difference as well. [34]

In Lipshitz's comprehensive study of microphone techniques "Are the Purists Wrong?" (purists usually are!) [49], strong arguments were advanced in favor of coincident and near-coincident microphone techniques for recordings destined for small room playback, noting that these configurations provide the coherent interaural phase cues needed for good spatial perception below 1 KHz. A basic assumption of Lipshitz's work was that the listener remains centered between two loudspeakers in a small room. Unfortunately, low frequency phase cues are of little use when listeners are displaced from a center listening position by 1-20 ms! In small rooms, interaural phase differences become ambiguous at wavelengths shorter than the interaural spacing. In large rooms, they are ambiguous at almost all wavelengths for almost all listener seats, no matter what microphone technique is used.

Snow appears to have understood stereo microphone technique for large room systems, and rejected the Blumlein approach. He showed graphically [13] how an array of microphones across the front of a stage, one per channel at approximately the spacing of the loudspeaker clusters could broaden the image and make it stronger. As a sound source moves to the left, the travel time to the left microphone is reduced and to the right microphone it is increased. These times are additive to the time of flight between the respective arrays and the listener. A principal benefit is to strengthen the contribution of the side components of the mix. This broadening can be controlled by the spacing of the microphone array and its proximity to the sound source (stage, orchestra, choir, etc.)

By contrast, closely spaced microphone arrays (including M-S arrays) do not strengthen the stereo image because they can produce only very small time shifts as the source moves. As a result, they can provide a useful stereo image in only a very few seats near centerline, and the image off center is no better than for a single panned microphone. As a source moves to the left from center, the delay increases equally in both channels (coincident) or increases slightly more in the right channel than the left (near-coincident).

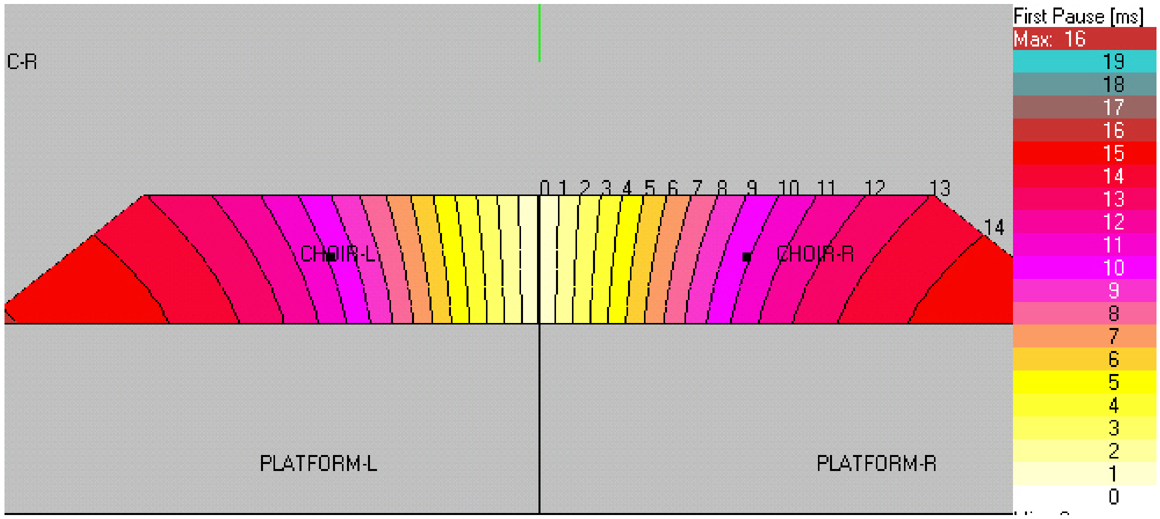

Fig. 10 Predicted arrival time differences (ms) for singers at various locations throughout a choir for a pair of microphones spaced 5.5 m apart and suspended 2.5 m above and 2 m downstage of the choir. As compared to a singer on centerline, a singer 3 m to the left of center is about 6 ms closer to the left microphone and 4 ms farther from the right microphone. These differences add (algebraically) to the travel time differences between the loudspeakers and the listener, and strengthen the stereo image for off-center sources and listeners.

Live sound production almost always involves compromises, and often requires a combination of approaches to achieve a successful mix. For example, a spaced microphone array, one microphone per channel, is usually the best way to reinforce a choir, classical orchestra, or the horn section of a big band. [21] Within the same performance, it is usually best to pick up quieter instruments and soloists within an ensemble using individual closely placed microphones that are then panned within the mix. Similar techniques are also quite effective for live theater. For a production of Sweet Charity at Caesar's Palace in Las Vegas in the mid-1970's, three variable-D cardioid microphones on microphone mice were placed at the lip of the stage. Chips Davis, who was working as a live sound mixer during the run of that show, got the idea from Burroughs. [21] Davis reported that the system (i.e., microphones and loudspeakers) had excellent upstage reach and localization on what he described as a very deep stage [50].

Console Panning

Gerzon's proposed law for LCR panning was specifically intended for use in small room stereo. It includes polarity reversal of the outer channels when the signal is panned more than halfway to either side. This author questions the viability of such a law for use in large rooms for a variety of reasons, not the least of which are its complexity and the high level of crosstalk between center and side channels. The overall panning method, however, from left to center with no signal fed to the right channel, and from center to right with no signal fed to the left channel, and with only the center channel fed when the pot is centered is without question the only viable protocol for LCR panning. This panning method, but with a constant power or sine/cosine law is referred to as the Gerzon/Brown protocol.

The most critical reason for use of the Gerzon/Brown protocol for LCR panning is that a single monophonic input panned using this protocol can never be present in both left and right clusters, thus the greatest arrival time difference that can occur between clusters for any locally generated signal (i.e., not a playback from recorded media) is that determined by the spacing between a side cluster and the center cluster. That spacing is, by definition, one half the spacing between left and right clusters! So inherently, the Gerzon/Brown protocol cuts in half the worst case time differences between arrivals at any seat as compared to panning only between left and right clusters. Or, looking at it another way, it allows a much wider physical spread between left and right clusters than could be used if no center channel was present. This is quite important when in very wide rooms.

Another virtue of the Gerzon/Brown protocol is that it is simple and intuitive for the operator. When panned to the left, the channel is sent only to the left cluster, when panned between left and center it feeds only the left and center clusters, when panned to the center it feeds only the center cluster, and likewise for the right cluster. The protocol is well suited to panning wireless microphones worn by actors as they move about a stage, and to panning instruments or microphone arrays between channels.

In the last few years, a few badly misguided manufacturers have built consoles with a strange form of panning that utilizes a conventional left/right pan pot, and adds a "center to left/right" pan pot that at one extreme maximizes the send to the center cluster while minimizing the send to the "standard" pan pot and at the other extreme maximizes the send to left/right and minimizes the send to the center. When both pan pots are in the center, the channel is sent at equal level to all three clusters!

Consider what happens when both pan pots in these mis-designed consoles are in the center. The left, center, and right loudspeakers are all reproducing this input channel at equal level, but those seated well off centerline are hearing three sound arrivals widely displaced in time. Not only will this degrade intelligibility in a wide room, but it destroys the stereo image by causing that channel to be perceived as coming only from the nearest loudspeaker! Unfortunately, when the mix console is set up on centerline the operator never knows there is a problem because he's hearing left and right equally and because of his location, they're not badly delayed relative to the center. Those sitting far to either side are not so lucky.

Panning by Means of Delay

From the earliest days of stereo, panning by means of differential delay between channels has always been seen as a very powerful way to achieve a stereo image, but until the implementation of digital mix consoles it was expensive to implement. Some early stereo films were produced using delay for almost all panning of dialog, music, and effects, [13] and Snow patented at least one method of achieving it using adjustable heads on a tape recorder. [51]

There is a critical difference between delay added to a recording and delay added to reinforcement. By virtue of loudspeaker placement (mostly the height) and latency in digital signal processing systems, most modern reinforcement systems already have more delay than is desired. This causes actors and singers to hear themselves reinforced 35-50 ms after they speak or sing, with resulting fatigue and tempo problems. While this can be partially compensated using foldback (stage monitor) loudspeakers, adding more delay for panning may cause more problems than it solves. The extent to which this is practical in any given application will depend on the design of the system, its relationship to the stage and the performers, and room geometry.

The author encourages mix console manufacturers to include this capability in their mix consoles. It is critical that the delay parameters and the pan laws be quite flexible, both for the console globally and for individual inputs, and it must be easy to vary them when a mix console is moved from one venue to another or used for a different type of production. One thing should remain invariant – the pan should default to the Gerzon/Brown protocol for LCR outputs when delay panning is not in use and for most delay configurations. While in theory the use of delay panning could allow a signal to be fed successfully to all three channels with varying amounts of delay, this flexibility offers many more ways to get in trouble, and it should <u>not</u> be a default.

The author also realizes that any commercial product, especially a relatively specialized or costly one, must find a wide variety of uses if it is to be commercially viable. Including Gerzon's polarity reversing 3-channel pan law as an option for DSP-based mix consoles could help achieve that goal if that pan law were to find acceptance.

Delay panning introduces another set of complexities. While a microphone may be panned using delay to LR or LCR front clusters, that microphone must be sent with a single delay to loudspeakers that are fed a monophonic mix of left and right (or L, C, and R) so that flanging distortion is not heard in the delayed loudspeakers. It is certainly practical for the more powerful DSP-based mix consoles to be designed to do this, but so far none are. The same limitations apply to L/R delays that supplement an LCR system. This is a non-trivial problem for the console designer/programmer. It is not uncommon for a permanent system to include LCR front clusters, monophonic delays for side boxes, monophonic or stereo delay under a balcony, and stereo delay for the balcony itself. And in all cases, the amount of delay needed for panning will be strongly dependent on the geometry of the venue and the sound system.

Consider the example of a 3-channel system in a concert hall with left, center, and right front clusters, 2-channel delayed loudspeakers alternating left/right under the edge of a balcony, monophonic delayed loudspeakers over the back rows of underbalcony seats, and 3-channel delayed loudspeakers over the balcony. Assuming that DSP was used external to the mix console to provide the delays for loudspeaker system precedence, such a system would require that each input module simultaneously provide delay-based panning to left/center/right outputs, non-delay-based to left/right outputs, and a simple monophonic output. And it is entirely possible the parameters for panning delay for the balcony might optimize differently from those for the front loudspeakers, in which case it could be useful to have two independent sets of delay-panned left/center/right outputs!

Delta Stereophony

Ahnert developed a very useful system for stereophonic reinforcement that has found use in large scale theatrical and musical productions. [42] The system utilizes a complex network of precedence and reinforcement loudspeakers supported by an equally complex system for mixing and signal processing. When first developed, the mixing and signal processing required stretched the limits both of available hardware and the people to operate it during a performance. Modern DSP-based mixing consoles and other automated mixing have the potential to make delta stereophony systems much more useful and practical. These mixing tools should be designed to supporting this very powerful technique.

CONCLUSIONS

A complex set of psychoacoustic principles allow human perception of three dimensional sound. While directional realism can be difficult to achieve, it is not the most important objective. Criteria for successful systems have been established based on these principles and the author’s experience. Understanding Snow’s patent [16], his 1953 SMPTE paper [13], and these perceptual issues is the key to understanding stereo in large rooms (i.e., performance spaces, cinemas, churches, etc.). Stereophonic sound reinforcement can be much more effective than equivalent monophonic systems, and can be operated at lower sound pressure levels while achieving equal levels of listener satisfaction as compared to monophonic systems. Very detailed design using modern acoustical modeling techniques is required to verify that the criteria are met for all listeners. Different loudspeaker types and arraying techniques are often required as compared to monophonic systems. Production techniques tailored to the needs of large rooms, including mixing techniques and the choice and placement of microphones are critical to the final result. Mix consoles to support 3-channel stereo are still few in number, and too many of those have fatal design faults. Systems using more than three channels are practical where geometry and production needs justify it, however the cost and complexity of loudspeaker systems, signal processing, mixing, and production for these systems limits the number of occasions where they can be used.

ACKNOWLEDGEMENTS

The author would like to thank the authors of the references cited for sharing their work with us, especially Harvey Fletcher and William B. Snow; Bob Lee, David Prince, and the Northwestern University Library for their help in obtaining historical papers; Peter Tappan, Chips Davis, Dr. Wolfgang Ahnert, Robert Coffeen, and Bruce Olson for their thoughtful discussions while this paper was in preparation and review of the manuscript; David Klepper for sharing his work with the Society in papers, workshops, and technical tours; and my wife, Dr. Rachelle Weiss for proofing the manuscript.

REFERENCES

1 Harvey Fletcher et al., *Electrical Engineering*, vol 53, pp 9-32, 214-219 (1934 Jan)

2 Harvey Fletcher, "Basic Requirements," *Soc. Motion Picture Telev. Eng. J.* 61:415-419 Sept '53

3 Harvey Fletcher, "Stereophonic Recording and Reproducing System," *Soc. Motion Picture Telev. Eng. J.*, 61: 355-363 Sept '53

4 John Hilliard, "Loudspeakers and Amplifiers for Use with Stereophonic Reproduction in the Theater," *Soc. Motion Picture Telev. Eng. J.*, 61: 395-407 Sept '53

5 Helmut Haas, "The Influence of a Single Echo on the Audibility of Speech," *J. Audio Eng. Soc.*, vol 20, pp 145-159, 1972 (Originally published as a doctoral dissertation in October 1949)

6 S. S. Stevens and E. B. Newman, “*The Localization of Actual Sources of Sound*,” *Am. J. Psychology*, vol 48, 1936, pp 297-306 (submitted for publication October, 1934)

7 Hans Wallach, Edwin B. Newman, and Mark R. Rosenzweig, “The Precedence Effect in Sound Localization,” *Am. J. Psychology*, Vol 62, no 3, July 1949 pp 315-336

8 Joseph Henry, “On the Limit of Perceptibility of a Direct and Reflected Sound,” *Proceedings American Association Advancement of Science*, vol v, pp 42, 48, May 6, 1851, reprinted in “*The Scientific Writings of Joseph Henry Vol I,*” Smithsonian Institution, Washington, DC, 1886, pp 295-296

9 Joseph Henry, “On Acoustics Applied to Public Buildings,” *Proceedings American Association Advancement of Science*, vol x, pp 119-135, August, 1856, reprinted in “*The Scientific Writings of Joseph Henry Vol II*,” Smithsonian Institution, Washington, DC, 1886, pp 403-420

10 William B. Snow, “Effect of Arrival Time on Stereophonic Localization,” *J. Acoust. Soc. Am.*, vol 26, pp 1071-1074, 1954

11 Mark Gardner, “Some Single and Multiple-Source Localization Effects,” *J. Audio Eng. Soc.* vol 21, nr 6, pp 430-437 (1973 July/Aug)

12 C.A. (Puddie) Rodgers, Pinna Transformations and Sound Reproduction, *J. Audio Eng. Soc.,* Vol 29, pp. 226-234. (1981 April)

13 William B. Snow, “Basic Principles of Stereophonic Sound,” *Soc. Motion Picture Telev. Eng. J.*, vol 61, pp. 567-589 (1953 November), reprinted in *AES Anthology of Stereophonic Techniques,* Audio Engineering Society, 1986, New York

14 Fedderson et. al., “Localization of High Frequency Tones,” *J. Acoust. Soc. Am.*, vol. 29, pp 988-991 (1957 Sept); reprinted by James M. Kates, “Optimum Loudspeaker Directional Patterns,” *J. Audio Eng. Soc.* vol 28, nr 11, pp 787-794 (1980 November)

15 J. C. Steinberg and William B. Snow, “Auditory Perspective – Physical Factors,” *Electrical Engineering,* Vol 53, no. 1, pp 12-15 (1934 January); reprinted in *AES Anthology of Stereophonic Techniques,* Audio Engineering Society, 1986, New York

16 William B. Snow, *US Patent 2,137,032*, filed 30 Sept 1936

17 Lorin D. Grignon, "Experiment in Stereophonic Sound," *Soc. Motion Picture Telev. Eng. J.* 61: 364-379 Sept '53 (originally presented May 18, 1948, revised and presented again April 27, 1953)

18 William B. Snow, *US Patent 2,126,929*, filed 15 May 1936.

19 Ted Uzzle, “*The History of Multi-Channel Sound with Picture*,” read before the Audio Engineering Society, 10 November 1994

20 Wolfgang Ahnert, personal communication with author, June 2002

21 Lou Burroughs, “*Microphones: Design and Application*,” Sagamore Publishing Co, Plainview, NY. 1974 pp 109-130, 184-186

22 Jim Brown, “Using DSP in Small Systems,” *Sound and Communications*, March 2002, pp 38-53

23 Jim Brown, “New Audio Rules at the Palace,” *Sound and Communications*, June 22, 1998

24 Gilbert F. Dutton, “The Assessment of Two-Channel Stereophonic Reproduction Performance in Studio Monitor Rooms, Living Rooms, and Small Theaters,” *J. Audio Eng. Soc.,* Vol 10, no 2, pp 98-105 (April 1962)

25 John Eargle, “Stereophonic Localization: An Analysis of Listener Reactions to Current Techniques,” *IRE Trans. Audio*, Vol AU-8, nr 5, Sept-Oct 1960, pp 174-178

26 Robert Coffeen, personal conversation with author, June 2002

27 Akira Mochimaru, “An Advanced Concept for Loudspeaker Array Design in Sound Reinforcement Systems, AES Preprint 3139, 91st Convention, October 1991

28 B. B. Bauer, "Broadening the Area of Stereophonic Perception," *J. Audio Eng. Soc.*, vol 8, pp 91-94 (1960 Apr)

29 D. M. Leakey, "Some Measurements on the Effects of Interchannel Intensity and Time Differences in Two Channel Sound Systems," *J. Acoust. Soc. Am.*, vol 31, pp 977-986 (1959 July)

30 D. M. Leakey, "Further Thoughts on Stereophonic Sound Systems," *Wireless World*, vol 66, pp 154-160, 238-240 (1960 Apr, May)

31 F. H. Brittain and D. M. Leakey, Two-Channel Stereophonic Sound Systems," *Wireless World,* vol 62, pp 206-210, 331-334 (1956 May, July)

32 James M. Kates, "Optimum Loudspeaker Directional Patterns," *J. Audio Eng. Soc.* vol 28, nr 11, pp 787-794 (1980 November)

33 Ronald Aarts, "Enlarging the Sweet Spot for Stereophony by Time/Intensity Trading," AES Preprint 3473 94th Convention, 1993

34 Michael Gerzon, “Microphone Techniques for 3-Channel Stereo,” AES Preprint 3450, 93rd Convention, San Francisco, 1992

35 James M. Kates, "Dipole Loudspeaker Response in Listening Rooms," *J. Audio Eng. Soc.* vol 50, nr 5, pp 363-374 (2002 May)

36 Hiroyuki Miyati and Shigeaki Aoki, "Stereo Reproduction with Good Localization in a Wide Listening Area," *J. Audio Eng. Soc.,* vol 32 #8, pp 433 (1984)

37 Tomlinson Holman, "New Factors in Sound for Cinema and Television," *J. Audio Eng. Soc.,* Vol 39:7/8 pp 529-539 (Jul/Aug 1991)

38 H. Kutruff, “*Room Acoustics*,” Applied Science Publ, London, 1973

39 Y. Ando and D. Gottlieb, “Effects of Early Multiple Reflections on Subjective Performance Judgements of Music Sound Fields,” *J. Acoust. Soc. Am.,* vol 65, pp 524-527

40 Don Davis and Chips Davis, “The LEDE Concept for the Control of Acoustic and Psychoacoustic Parameters in Recording Control Rooms,” *J. Audio Eng. Soc.,* Vol 28, no 9, pp 585-595, September 1980

41 Peter Tappan, “An Unusual Nightclub Sound Reinforcement System,” AES Preprint 626, 35th convention, October 1968

42 Wolfgang Ahnert, "The Complex Simulation of Acoustical Sound Fields by the Delta Stereophony System DSS," AES Preprint 2418 81st Convention, 1986

43 Vance Breshears “Design Approach for Multi-Channel Sound Reinforcement Systems,” AES Preprint 4834, 105th Convention, San Francisco, Sept 1998

44 Richard C. Heyser, “Concepts in the frequency and time domain measurement of loudspeakers,” Proc IREE, vol 37, no. 3, pp 67-76, 1976 March, reprinted in “*Time Delay Spectrometry, An Anthology,*” Audio Engineering Society, New York, 1988

45 Vance Breshears, “Mixing Techniques for Multi-Channel (Left/Center/Right) Sound Reinforcement Systems,” AES Preprint 5486, 111th Convention, NY, Dec 2001

46 Jim Brown, “Design Concepts for Mixing Consoles for Multichannel Reinforcement Systems for Music and Theater,” AES Preprint 2849 94th Convention, 1989

47 Michael Gerzon, "Panpot Laws for Multispeaker Stereo," AES Preprint #3309, 92nd Convention, 1992

48 Tomlinson Holman, Oral presentation at AES Workshop on Cinema Sound, c.a. 1985

49 Stanley P. Lipshitz, "Stereo Microphone Techniques: Are the Purists Wrong?," *J. Audio Eng. Soc.,* vol 34:9 p 716 (1986)

50 Chips Davis, personal communication with author, June 2002

51 William B. Snow, *US Patent 2,831,069*, filed July 1, 1954