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# Audio Amateur

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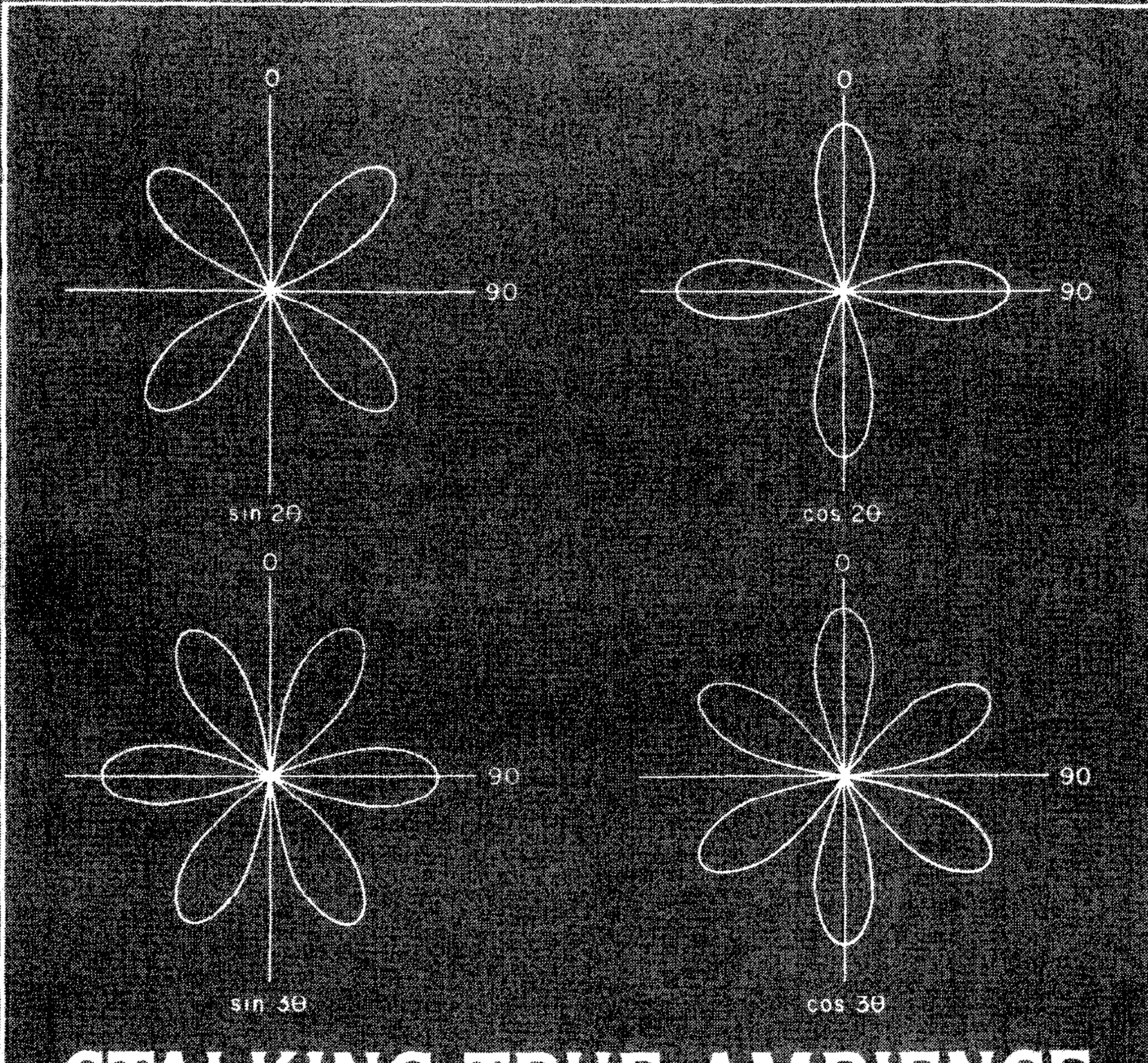


STALKING TRUE AMBIENCE

*Part I*

# AMBISONICS COMES OF AGE

BY WILLIAM SOMMERWERCK

*An Ingenious Man who had built a flying-machine invited a great concourse of people to see it go up. At the appointed moment, everything being ready, he boarded the car and turned on the power. The machine immediately broke through the massive substructure upon which it was builded, and sank out of sight into the earth, the aeronaut springing out barely in time to save himself.*
*"Well," said he, "I have done enough to demonstrate the correctness of my details. The defects," he added, with a look at the ruined brickwork, "are merely basic and fundamental."*
*On this assurance the people came forward with subscriptions to build a second machine.*

*—(the great American satirist and misanthrope) Ambrose Bierce*

MR. BIERCE'S observation sums up the history of quadraphonic sound. Manufacturers dumped millions of dollars into its development without any serious examination of its fundamental assumptions. Had anyone "in authority" thought twice about what was going on, *none* of the proposed systems would have been used. All the discrete formats (Q-4 and Q-8 tapes, CD-4 records) and the two most popular matrix systems (SQ and QS) are based on incorrect assumptions about the way we hear directional effects.

Many people knew these things, but none of them worked for the major record companies, and those who did speak up (e.g., John Eargle and Duane Cooper) were ignored. Surround reproduction ought to have become part of any good playback system and *especially* of perfectionist systems. The shortsightedness of en-

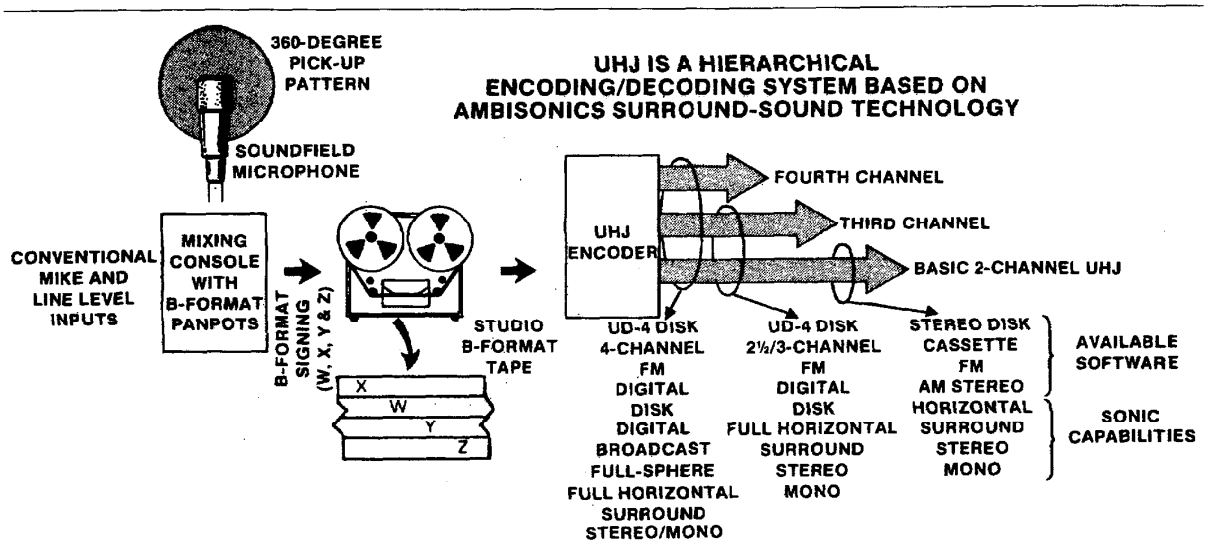




*FIGURE 1: The UHJ hierarchical system of encoding and decoding directional sound information within the Ambisonics technology.*

gineers and the greediness of recording companies guaranteed the slow, agonizing demise of quadraphonics.

If quad is dead, why am I writing an article about Ambisonics? Ambisonics is *not* a quadraphonic system. The only common characteristic is the use of at least four speakers. Ambisonics is a universal recording technology (*Fig. 1*). I can't emphasize this too strongly: Ambisonics can create almost any sound effect imaginable with amazing directness and simplicity. These effects range from a highly accurate reproduction of a concert hall's ambience, including the vertical reverberation, to placing individual sound sources anywhere around the listener—even above and below—with any desired size, motion or apparent distance.

**In the Beginning**

To see why Ambisonics is right and the other systems are wrong requires more than a simple explanation of ambisonic technology. A brief discussion of the history of surround-sound technology will make it easier for you to understand where most of the designers went wrong.

The first surround-sound recording was the sound track of *Fantasia*, made in the early 1930s. This recording included seven audio tracks on a separate film reel, with additional control tracks to expand the dynamic range and to determine which speaker received a particular channel.

Where did the "Fantasound" system come from? Although there were some experiments with stereo and binaural techniques in the late 19th century, I have found no references to surround experiments. Even the Bell and Decca stereo patents of the early '30s make no mention of surround effects. (Alan Dower Blumlein, the scientist who filed Decca's patent, is generally given credit for "inventing" stereo—i.e., reducing it to a workable system. His patent, which is cited by almost every surround-sound patent, anticipates matrixing, with-height effects and even the 45-45 disk recording system. But nowhere does it say anything about horizontal surround.) It appears, then, that Disney Studios—or, more likely, RCA, who built the system—should receive the credit for surround sound.

After *Fantasia*, surround sound disappeared for another quarter of a century, only to resurface in the early 1960s with the Revere-Wollensak tape-changer system. This system included a third track along the center of the tape to carry ambience. No one knows who developed the system, and there is no evidence of experimental three-channel recordings.

**Let There Be Light**

Not until 1969 did surround sound make its first "modern" appearance. At that time, Acoustic Research (AR) and Vanguard announced their experiments using two extra channels to convey the acoustic character of the performing space. The idea was to use four-channel, open-reel tape, with tracks 2 and 4 carrying the ambience. Two mikes at the back of the hall would pick up reverberation.

Sonic nirvana! Nonetheless, many cynics felt that quad had been developed just to sell more equipment. To use the system, a listener had to buy a new tape deck *and* another amplifier *and* the special tapes from Vanguard.

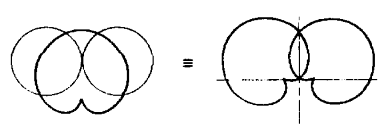

*FIGURE 2: You can use the Middle-Side (M-S) technique to obtain conventional left and right channels.*

Because nothing important was happening in audio at this time, audiophiles greeted this announcement from two of the industry's leaders with great interest. Thus the industry stumbled into the modern age of surround sound. "Stumbled" is the right word, for with one stroke, Vanguard and AR managed to create two myths about surround reproduction.

**Surround-Sound Error #1**

*Ambience is to be recorded by placing the ambience microphones to the rear of the hall.*

No matter how directional the rear mikes are, there is no way to prevent them from picking up some of the direct sound. The difference in acoustic path length between the front and rear mikes delays this sound, so you hear the direct sound twice. This never occurs in real life because your head can be in only one place at a time (physically speaking). Such recordings have a fairly obvious slap echo.

It is surprising how many otherwise knowledgeable recording engineers made—and still make—recordings this way. The Ambiphon (Sonar) tapes engineered by Mitch Cotter use widely spaced omnis and have a slight, but noticeable, slap. Marc Aubort and Joanna Nickrenz, the producers of most of the Vox/Turnabout quad recordings, also use this technique. Again, you often hear a distinct front/back effect as the direct sound passes each set of mikes.

However pleasant the overall effect (it *does* enlarge the acoustic space), it is objectively incorrect because the listener is hearing the direct sound from one position and the reverberation from another. If the producer wants to use spaced omnis, he or she should place the ambience mikes below the front mikes (to prevent acoustical interference), facing into the hall. These mikes must be directional, and acoustical baffling should be used to block any direct sound that might reach them.

In one respect, it is easy to understand this kind of mistake. After all, conventional stereo consists of sampling the sound field at two points. What could be more natural than sampling it at four points for surround sound? This leads to the second surround-sound error.

**Surround-Sound Error #2**

*If there are x loudspeakers, then there must be exactly x distinct channels of information feeding them.*

This is a more subtle error. We are so brainwashed by 30 years of two-channel reproduction that we quite naturally assume that each speaker requires a discrete signal. A mathematical view of the subject suggests that we might be overlooking something. Two points define a line, and in regular stereo, the image is strung out in a line between the speakers. Only three points are needed to define a plane. Why, then, should we need four signals to position the sound around the listener? Wouldn't three be enough?

At first blush, you wouldn't think so. How do you get four speaker feeds out of three signals? You can understand the technique for doing this by examining how you get two speaker feeds out of two signals. Take stereo FM. Speaker-feed signals are not directly transmitted. Instead, a sum (L+R) signal in the baseband provides compatible reception for mono receivers, and a difference (L−R) signal in the subcarrier allows you to regenerate L and R by taking the sum and difference of these two signals.

A similar technique, called M-S (for Middle-Side), was common in the early days of stereo recording. One microphone, usually a cardioid, faces the orchestra to provide a mono pickup. A second mike, always a figure-8, is arranged sideways to specify the direction of the arriving sounds. By adding and subtracting these two mike outputs, you can obtain conventional left and right channels (*Fig. 2*). By simple extension, you could set a third figure-8 mike facing forward to obtain front-back directionality. Its output would be added and subtracted to each of the two signals created from the first pair of mike outputs.

This would create a total of four signals. Using this approach, each added transmission channel enables you to double the number of speaker feeds. This does not necessarily mean that these speaker signals will produce a correct image. It does show, however, that there might be better ways to transmit surround sound than by direct speaker feed.

### To Market

There was just one catch to this "practical" system from AR and Vanguard: it worked only with open-reel tape, which has never been a mass-market item. If there were any profit (aha! the magic word) to be made in quad, it had to come from phonograph records. But no one had found a reliable way to record four channels on a disk.

JVC was experimenting with one method that added an ultrasonic carrier to provide an ambience channel. The AR-Vanguard demos encouraged JVC to speed up its research, and about a year later, they announced CD-4 (Compatible Discrete Four-Channel) records. The technology involved in producing these carrier disks—improved vinyl formulations, half-speed mastering, superior cutting and playback styli, wideband pickups—made a major contribution to disk recording and reproduction. It is a shame the same cannot be said for CD-4.

With most of the pickups designed for this system, CD-4 stood for "Continuous Distortion to the Fourth Power." Many recordings showed a kind of gurgling midrange mush, combined with severe high-frequency splatter and breakup. As one reviewer put it, you were constantly on the edge of your seat "waiting for something to happen." The disks varied widely in quality, with the earliest and latest giving the best performances. Even when they played properly, however, a high level of coloration falsified instrumental sound. In short, the sound quality of CD-4 was far below that of the best stereo records. Ironically, CD-4 paved the way for today's high-technology disks.

The biggest breakthrough in the development of quadraphonic sound, however, came from Peter Scheiber, the musician-engineer who invented matrixing. Matrixing made it possible to put the four channels of the master tape on a conventional disk.

From an over-simplified mathematical point of view, a matrix transforms one set of discrete quantities into another set of quantities. In this case, the four original channels are transformed into two, which can be readily transmitted via any two-channel medium. The transformation is made in a controlled fashion, so it should be possible to reverse it.

### Loss of Information

The catch is that we started with four pieces of information, but ended up with only two. It is a law of nature that we cannot unambiguously solve for four unknowns with only two equations. Why do I refer to audio signals as "unknowns"? Simply because they are just that. If a listener knew exactly what they were, he or she would not have to buy the recording. This is a fundamental principle of information theory. Because the intended receiver does not know the exact nature of the information being transmitted, such information is subject to noise and distortion. If the receiver knew exactly what would be sent, it could be received with perfect fidelity. But if he or she knew what would be sent, there would be no need to send it.

To see this, consider the SQ encoding equations. Lt and Rt signify the two transmitted signals, which are the left and right track, respectively. Lf, Rf, Lb and Rb should be self-explanatory.

$$Lt = Lf - (0.707 \times j \times Lb) + (0.707 \times Rb)$$

$$Rt = Rf + (0.707 \times j \times Rb) - (0.707 \times Lb)$$

where j is a 90-degree phase shift, which most matrix systems use to reduce encoding ambiguity.

You can play with these equations for the rest of your life, but you cannot manipulate them to get each channel back by itself. For example, if you try to solve for Lf, there will always be some Lb and Rb mixed in. This represents undesired crosstalk among the original four channels. A general analysis shows that, for any given channel, you can design a matrix to provide complete separation from one other channel, with the remaining two channels appearing as crosstalk, 3dB down. You can manipulate the matrix coefficients to reduce this crosstalk, at the expense of greater crosstalk in the direction of "full" separation. Unfortunately, the best you can do is about −7dB crosstalk, all around. This is not enough to give subjectively perfect separation. Because of this, virtually every technical advance in quadraphonics in the past decade has been directed at enhancing matrix separation, first with Vario-Matrix, then with Audionics' Shadow Vector and the Tate Directional Enhancement System for SQ.

### Garbage In, Garbage Out

And *that* was the mistake. Channel separation has *nothing whatsoever* to do with good surround sound. The point of surround recording is accurate reproduction of recording site acoustics.

To put it a bit differently, there is only one criterion for a "good" surround-sound system: does the system allow the acoustical character of the performing space to be accurately reproduced? Because this requires highly accurate localization in all directions, a system that is good at recreating ambience will also excel at creative or arbitrary effects. The discrete master tape, which CD-4 records and the output of advanced matrix decoders are supposed to mimic, is *not* a valid reference. Only live sound and the producer's intentions can be considered as such.

Many of you must be thinking that this is a lot of semantic hair-splitting. If separation of the original four channels on the master tape is maintained, won't the directionality be correct? The answer is no. That reasoning assumes that what is on the tape correctly reproduces the directional character of the sound source, real or imagined. But the tape does not provide accurate reproduction because "quad" recordings were always made with psychoacoustically incorrect techniques. This will make more sense when you examine the next surround-sound fallacy.

**Surround-Sound Error #3**

*The proper way to encode directionality is to pan the signal between adjacent pairs of speakers.*

For example, if a sound is to come to the immediate right of the listener, it should be fed at equal levels to the right front and right rear speakers. This technique is known as Pair-Wise Mixing (PWM). This mistake, more than any other, has blocked the proper development of surround sound. The PWM assumption is so dreadfully wrong that it is amazing so few people have blown the whistle on it. John Eargle's *Sound Recording* points out this error in his chapter on quadraphonics, and Katsumi Nakabayashi of the NHK mentions it in the 4/75 *JAES*.

Scientific American has recently published *The Science of Musical Sound*, by John R. Pierce. (Pierce spent many years at Bell Labs and is one of the great scientist/electrical engineers of this century.) In his chapter on sound reproduction, Pierce suggests that conventional quad systems are wrong and that the only "proper" approach is to extract the pressure and velocity components of the sound field and to adjust their levels at the listener's ear to match those in the live field. His suggested speaker layout places the speakers at the points of a tetrahedron, with one speaker directly above the listener and the other three below. As I will show later, this is the ambisonic approach, although its speaker layouts are more suited to domestic conditions. *Figure 3* shows a more practical arrangement.

We are accustomed to hearing PWM in almost all stereo recordings. A given sound is positioned between the speakers by adjusting its relative amplitude in each channel. Paradoxically, this technique is used both in gimmicky multitracked studio recordings and in single-point "purist" recordings. In studio recordings, each instrument or performer is assigned its own track, which results in dozens of little mono recordings. When the two-track cutting master is made, a pan pot adjusts the relative channel levels to position the sound where the producer wants it.

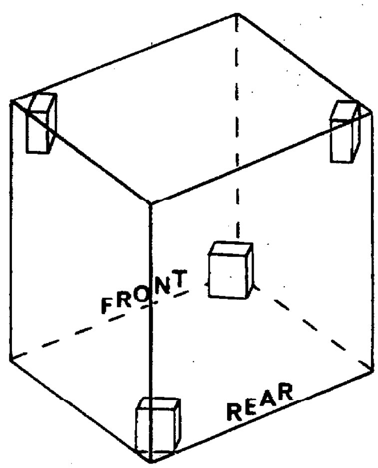


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*FIGURE 3: Tetrahedral loudspeaker layout, embedded in a cube.*

Oddly, the same effect results when making a single-point recording. With two closely spaced cardioid mikes, there is almost no separation to introduce phase or timing differences. Because the mikes are directional and are aimed in different directions, anything other than a central source will produce a different output level from each mike. This creates the amplitude differences needed for a directional effect.

In both cases, you use amplitude differences to position sound sources. This is called amplitude panning. Of course, the ear and brain also use phase and arrival time differences for localization. Another form of purist recording technique, spaced omnis, uses these additional cues. As I have shown, however, this technique produces fundamental errors when applied to surround-sound recordings, so I will ignore it.

**Amplitude Panning**

Amplitude panning is used almost universally in conventional stereo. If someone asked you how to pan the sound around the listener, what would you suggest? The most obvious technique is to treat adjacent speakers as pairs and pan the sound between them. Guess what? It doesn't work.

To see this for yourself, try the following experiment. Set your system on mono or play a mono recording or broadcast. Face the speakers. If your system is set up correctly, the sound should appear to be coming from dead center. Turn 90 degrees to the left or right. Now where does the sound come from? It is *not* from dead right or left. There is no clear-cut source—just an amorphous blob of sound—or it might appear to jump from one speaker to the other, particularly as you rotate the balance control to "pan" the sound. The distinction between face-on and sideways listening should be obvious, but if it is not, try adding a third speaker in the middle and use it as the reference. If you still cannot hear the difference, I would suggest selling one of your speakers and going back to mono.

If you own the Audionics Space & Image Composer, the Fosgate Tate II or a Sansui Vario-Matrix decoder, you might have noticed similar effects. Sitting sideways, you might hear instruments or performers clearly positioned between the (true) front and rear speakers. When you face forward again, the preciseness of their positioning disappears. Sometimes it becomes more difficult even to *hear* the instruments. Likewise, when expanding stereo recordings into a "horseshoe," you might have noticed that the sides are noticeably bereft of sound sources, unless you turn to face the sides. All this is due to the fact that SQ and QS recordings are pair-wise mixed, and the Tate and Vario-Matrix systems blindly follow this paradigm in their decoding action.

What about live recordings? What happens if you place four cardioid mikes in a square, at 90-degree angles to each other? Won't that work properly? No. You are still using pair-wise mixing. Reverberant sounds arriving from the sides are amplitude panned between the front and rear mikes on that side. Thus, their directionality is not correctly reproduced on playback. The side reverberation is critical to a proper appreciation of the exact sonic character of the performing space. I will discuss exactly how this psychoacoustic failure affects the reproduction in the section on miking technique.

It should now be apparent that every quadraphonic recording ever made has its intended directionality incorrectly encoded. *Every one*. Regardless of whether manufacturers used coincident mikes, spaced omnis or pan potting, they failed to recognize the fundamental engineering approach needed for good surround sound. That approach is not a question of how to transmit four sound channels, but rather of how to create an accurate, stable sound field in the listening room that closely mimics what the listener would have heard live or what the producer imagined. In other words, what technology gives accurate imaging? Instead of focusing on this problem, engineers focused on poor separation, an inherent weakness of the matrix system, which was the most practical commercialization of quad.

The problem of obtaining subjectively correct imaging is not new. It has been around for more than 50 years, since the advent of stereo. Few (if any) recording engineers can tell you what kind of mikes you should use and where you should place them so that the listener hears the performers at the correct angular position, at the right apparent distance and with the ambience in its correct relationship. The reason for their inability to do so is that no systematic study of stereo imaging or the technology required to produce accurate or arbitrary effects has ever been conducted.

### Getting Their Act Together

The *raison d'etre* of surround sound is the ability to position a sound image at any point around the listener, including above and below. Any surround-sound system must include a technology for controlled imaging. Otherwise, it is a waste of time, money and engineering effort. Unfortunately, all the proposed discrete and matrix systems *were* a waste because they were based on the wrong assumptions about the way the ear and brain determine directionality.

In approaching the design of a psychoacoustically correct surround-sound system, forget, for a moment, mono and stereo compatibility. Also ignore the idea that four discrete signals should feed four speakers. Instead, consider the recording and playback chain as a whole to see what technique of encoding and decoding directionality gives the best results.

First, let's define encoding and decoding directionality. The former refers to the way we specify the amplitude and phase of the signals on our transmission channels for any particular source direction. The latter refers to the signal processing necessary to produce speaker-feed signals that accurately reproduce that directionality.

Ideally, the encoding and decoding should be independent of each other. This means that in addition to providing a precise specification of directionality and using the least transmission space possible, the encoding should not limit the decoding process. That is, it should not restrict you to a fixed number of speakers or special speaker locations. The listener should have some freedom in speaker placement and the ability to add speakers for greater positional accuracy and a broadened listening area.

The encoding must not carry implicit assumptions about how it is supposed to be decoded. As we gain increased understanding of the hearing process, it might be possible to create improved decoders for superior imaging accuracy from all existing material. This approach is in sharp contrast to discrete quad, where exactly four speakers are arranged roughly in a square, and the encoding technique (PWM) cannot provide correct imaging.

It is important to point out that none of the quadraphonic "systems" are really systems at all, as they do not specify directional encoding in a way that can be correctly decoded. In essence, whether discrete or matrix, they are simply ways of transmitting four uncorrelated channels of information, without any regard for positional accuracy.

Compare this with color TV, which *is* a true system. Broadcast standards focus on producing a signal that accurately represents the colors of the original scene, within the limits of existing technology. Because nothing is said about how this information is to be displayed on the home receiver, there is room for improvement at both ends. A modern Philips camera produces a noticeably more transparent and vivid picture than an early RCA, and this improvement is visible on any receiver. Likewise, the development of new phosphors has broadened the range of achievable colors, enhancing the reproduction of existing material.

The same general principles apply to surround sound. Certain classes of encoding schemes are fundamentally correct and will allow accurate directionality in playback. Similarly, the best playback techniques require encoding that is not limited by PWM or other erroneous techniques. Any optimized playback system should also allow use of as many speakers as you would like to improve the accuracy of the directional effect.

### The Right Stuff

Not long after Scheiber's initial work, one of the seminal papers on surround sound appeared in the *Journal of the Audio Engineering Society* (Volume 20, Number 5, 1972, p. 346). The paper presented the Universal Matrix (UMX) system and was written by the widely known acoustical engineer Duane Cooper (then of the University of Illinois and best known for the Cooper Time Cube, an early digital delay system) and Takeo Shiga, of Nippon Columbia (Denon to us). It is important for two reasons: it introduced the first correct way to encode directionality, and it addressed the question of whether a given encoding/decoding produced the desired directional effects.

The encoding technique they used is known as azimuthal harmonic synthesis, which is a variation of Fourier analysis. Remember that Fourier showed how any repetitive wave motion could be analyzed into a fundamental frequency, plus harmonics. Each waveform has its own characteristic pattern of harmonics, and the harmonics have distinctive phases. Each waveform also has an average, or DC, component.

If you surround a listener with performers or have the orchestra in front and the reverb all around, you will be able to recognize some pattern in the distribution of sound sources. Each time you trace a circle around the listener, the pattern will repeat. If you could analyze this repetitive waveform, you would have a useful representation of the position of all the sound sources, including ambience. How can you do this?

The easiest way to see this is in terms of microphone patterns. Think of an omnidirectional microphone placed at the listener's position. An omni mike has no directional preference: it responds identically to sounds arriving from any direction. Its output is independent of the direction of the arriving sound: it conveys no directional information. It simply presents the sum of direct and ambient sounds—a mono pickup. Walking around the mike's pattern, you can see that the output is constant, in the same way a DC signal is constant. To put it another way, the monophonic component of any sound field is equivalent to the DC component of a Fourier analysis of the field. But what about the AC components?

Obviously, you need a pattern that varies with direction—but what kind of variation? We saw that an omni mike has no variation and that when a DC signal was bent into a circle, it matched the pickup pattern of an omni. Let's look at this idea more closely. The fundamental frequency of a Fourier analysis is a sine or cosine wave. What do you get when you bend a cosine wave into a circle? More accurately, what is the pickup pattern of a microphone whose response varies with the cosine of the angle of incidence? What you get is a figure-8 pattern, with the point of maximum sensitivity facing front (0 degrees).

If you look at a table of cosines, you will see that the cosine of any angle between 180 and 360 degrees is negative. Does that surprise you? It should not. After all, the output of a figure-8 mike is inverted in phase for rear sources. Remember that most figure-8s are ribbon mikes, in which a thin foil ribbon is suspended in a powerful magnetic field. A membrane cannot respond to signals coming from the sides, but will give full output for sounds that are directly to the front. According to simple trigonometry, the effective pressure gradient at the diaphragm varies with the cosine of the angle of incidence—from 1.00 at 0 degrees to 0.707 at 45, 0.5 at 45 and 0.0 at 90 degrees. Because sound sources at the rear move the ribbon in an opposite direction from those at the front, their outputs are inverted in polarity.

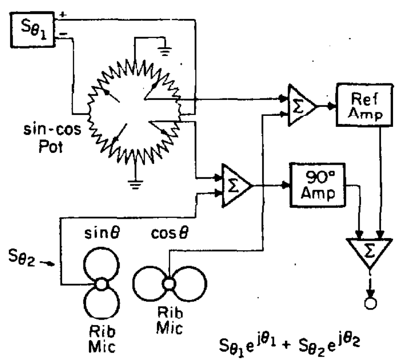


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*FIGURE 4: Azimuthal harmonic synthesis showing electrical and acoustical mixing.*

Remember from Fourier analysis that the fundamental and each harmonic have a specific phase angle. This angle is the arctan of the ratio of that harmonic's sine and cosine component amplitudes. To specify the phase angle of the azimuthal harmonics, therefore, you need a sine component, in addition to the cosine component from the figure-8 pickup. Sine(theta) equals cosine(theta – 90). In other words, there is a 90-degree phase angle between sine and cosine. Simply turning the figure-8 mike to the left or right by 90 degrees produces a sine-weighted output (*Fig. 4*).

### Simplicity Is the Key

You can take a similar approach with higher azimuthal harmonics by weighting the sound sources according to the sine and cosine of two theta, three theta, and so on (*Fig. 5*). These are easily produced from mono sources by adjusting amplitude and polarity. They pose serious problems to live recording, however,

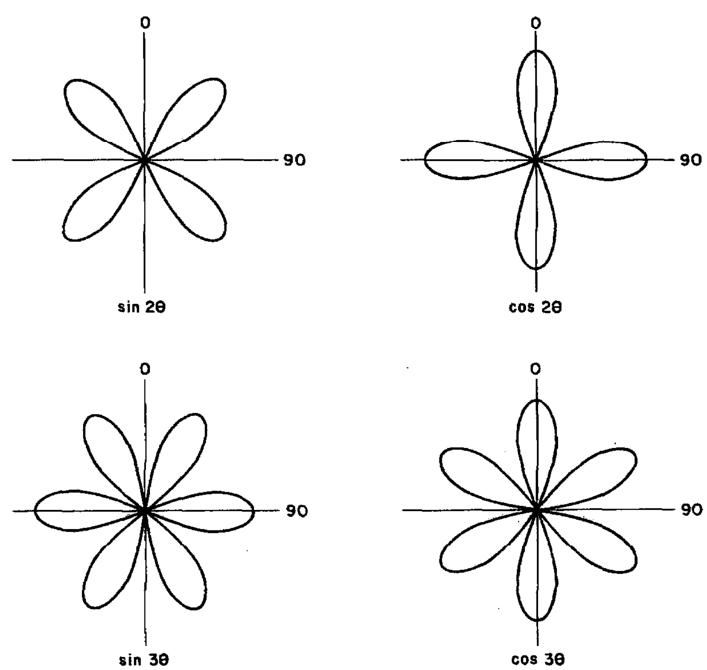


*FIGURE 5: Higher-order azimuthal harmonics.*

where avoiding multitrack-mono limitations would require a mike with these complex patterns. Consequently, you should not use anything higher than first-order harmonics. Because there are a finite number of channels, let's look more closely at what you can do with a mono component, plus the sine and cosine of the fundamental.

Clearly, while the mono component provides the sound, the figure-8 sine and cosine components indicate directionality. (If this is confusing, refer to the previous discussion of M-S miking. The principle is identical.) Are these three signals enough to specify directionality unambiguously? Let's see. Think of the four cardinal points of the compass. As a sound source moves from one point to the next, the relative outputs of the sine and cosine-weighted channels move in opposite directions, one rising while the other falls. The ratio of their amplitudes determines the relative angular position of the sound source within the quadrant.

Which quadrant is that? Here is where you use the relative polarity. In the left front quadrant, sine and cosine are positive. In the right front sector, cosine is positive and sine is negative. Each quadrant has a distinct pair of polarities. At least from an electrical point of view, it is possible to define the position of a sound source anywhere in a plane with only three channels of information. Any more information is redundant. Anyone who insists that you need four channels to preserve compatibility or to provide full artistic freedom is wrong.

We have now precisely encoded the directionality of any horizontally located sound source with only three signals and no reference to speaker-feed signals. This leaves the door open to using any number of speakers. This is in stark contrast to QUADraphonics, which assumes that acoustic space will be sampled at *four* points and that these *four* samples will be transmitted to the listener's *four* speakers via some *four*-channel medium. Since when is four a magic number?

Of course, the full ambisonic system *is* four channel, but the fourth signal carries up/down information. None of the channels is a speaker-feed signal. Furthermore, you can show that there is a hierarchy of possible surround sound systems, where the sound field is sampled with n^2 (1, 4, 9, 16, 25, and so on) patterns. It is a coincidence that the simplest practical ambisonic system needs four channels and that quad reproduction also has four. There is no fundamental similarity between quad and Ambisonics.

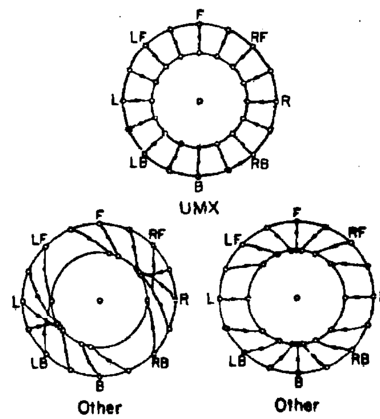


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*FIGURE 6: The Cooper/Shiga Universal Matrix (UMX) system compares favorably with the "other two" systems. It satisfies the Makita localization for every position around the listener.*

### Delivering the Goods

How do you process these signals so that the listener hears correct directionality? Cooper and Shiga made their major contribution in this area. They suggested that these signals be decoded by placing at least four speakers in a circle around the listener. The speakers need not be "toed-in," just equidistant from some center point. The omni signal is fed to all the speakers with identical amplitude and polarity. The front/back figure-8 signal is also fed to all the speakers, except that it is weighted by the inverse of the cosine of the angle the speaker receiving it makes with center front. The same thing happens to the front/back signal, except it is weighted by the inverse of the sine of the angle. Notice that sine and cosine can be positive or negative, and this must be carried through in the weighting. Therefore, the left/right signals fed to the right speakers will have a negative polarity with respect to the mono signals, since the sine of the angles between 180 and 360 degrees is negative. Similar reasoning shows that the front/back signals fed to the rear speakers will also have a negative polarity.

Basically, they applied the encoding process to decoding, but in an inverse fashion—i.e., division replaces multiplication, and the positions of the speakers replace the positions of the sources. In showing that this worked, Cooper applied a theory of directional hearing called Makita localization.

Developed by a Japanese acoustician of the same name, this theory allows for any number of sound sources, at any azimuthal positions, equidistant from the listener. They all carry the same signal, but with arbitrary phases and amplitudes. When you plug the amplitudes, phases and positions into a formula, an angle pops out. The sound appears to come from this direction when you turn to face it. (This means that the apparent angular position of the source might be slightly different from the actual angular position when you turn toward it. The same effect occurs with live sources.)

Cooper showed that his proposed encoding/decoding system, dubbed UMX, satisfied the Makita localization formula for every position around the listener. He also ran listening tests, which verified this. He displayed the results in three plots (*Fig. 6*), contrasting UMX with SQ and QS. Cooper refused to identify the other two systems, considering it a breach of professional ethics to do so. Frankly, the paper would have had more impact if he had. After all, he *was* telling the truth.

The importance of Cooper's work —a system that properly encoded directionality and the practical proof that such encoding could be used to create precise directional effects on playback—was ignored. Part of the problem was the opaqueness of the writing. Most technical papers are badly written, and Dr. Cooper's was no exception. The other part of the problem was that many AES members are just not interested in what their co-workers are doing.

*Continued on page 58*

# LETTERS

"Showcase" piece *(TAA 4/83, p. 53, Photo 8)*, the word is used—improperly, I think—to indicate idling collector current.

In the dear old days of vacuum tubes, we used the potential drop across the cathode resistor to bias the tube into the proper operating condition, and the grid was tied to ground through a rather high-value resistor. Frequently, we increased the gain with a "bypass" capacitor. Sometimes we used a fixed negative voltage on the grid as a bias. We did all this to bias the characteristic of the tube into a more linear region of operation.

With transistors coming on the scene, the "Shea bias" was common. This is a resistive divider on the base that induces an idling current, putting the device into more linear operation. In addition, solid-state power amplifiers invariably have some circuit arrangement to "bias" the output devices into class B, AB or even A operation. Without such bias, the output would be near class C, and the crossover notch would be horrible. Now the "bias" can be adjusted to control the collector current of the output—either single ended, complementary push-pull or quasi-complementary.

What all this means is that Mr. Vikan has "biased" his output to run at an idling current of 400mA. In the future, I think we should be careful to use the term properly.

L.B. Dalzell
El Cajon, CA 92020

## CONNECTION CORRECTION

I am writing in response to Darin Ernst's letter in *TAA* 5/83 (p. 55). Mr. Ernst and I have had a very fruitful exchange of letters and telephone calls in an effort to solve his problem. He did not use the standard layout, but mounted Old Colony's boards in an elegant enclosure of his own design. We both considered exotic and obscure causes for his problem, but Mr. Ernst finally discovered the real cause himself—bad solder joints and bridges.

Let this be a lesson to other readers. If something does not work, check and recheck the connections and boards, then have someone else check them again. In my experience, if a transistor or IC is bad, it usually got that way as a result of one of my wiring errors.

Bernhard F. Muller
Milan, MI 48160

## AMBISONICS

*Continued from page 13*

Badly written or not, the Cooper/Shiga paper is very important in the history of audio engineering because it was the first step in creating a rationalized, practical system of directional effects. Cooper's association with Shiga was also significant commercially, as Denon later released the system under the name UD-4. This was carrier-disk technology, with a significant difference. Unlike CD-4, the basebands were a matrixing of the four UMX signals, which could be decoded to give credible results by themselves or combined with signals from the carriers for uncompromised performance.

Cooper showed that the carrier signals did not need a bandwidth greater than 2 or 3kHz to provide good directional effects. (CD-4 disks modulated the carriers up to 15kHz.) Since the carrier's sidebands did not have to extend to as high a frequency, a quieter, lower-distortion, longer-wearing record was possible. It is difficult to understand why UD-4 failed and JVC's CD-4 succeeded, when UD-4 had less distortion and more accurate directional effects.

### The Kernel

The choice of names also was detrimental to the Denon system. UMX stands for Universal Matrix, while UD-4 stands for Universal Discrete Four-Channel. UMX is neither a matrix nor a discrete system: it is a kernel system. Matrices are mathematical transformations, or mappings, of one set of variables into another. Usually, the transformed set has a smaller dimension than the original set. SQ and QS are examples of this, where four channels are transformed into two. The significant thing about matrix transformations is that they operate on discrete values.

A kernel also transforms variables from one set of dimensions to another, but the transformation is continuous—that is, it is defined as a smoothly varying function, not as a fixed set of discrete coefficients. This distinction is critical. A kernel allows you to discard the idea that each transmission channel must correspond to a speaker-feed signal. Instead, you can implicitly specify directionality for all possible positions, without exactly referring to how the directionality is to be presented.

On the other hand, in both discrete and matrix quadraphonics, the four speakers are treated as four separate sources. Sounds are panned between speaker pairs, and each speaker-feed signal is transmitted in such a way as to preserve its individuality. There is a one-to-one matrix relationship between the original four channels and the final four speaker feeds. In this sense, all discrete systems are actually matrix systems. This is unconsciously acknowledged in the 4-4-4 terminology used to distinguish discrete systems from 4-2-4 matrix systems.

For whatever reasons (probably the lack of commercial response to the UD-4), UMX never developed past this point. The British then took up where Cooper and Shiga left off.

*Next time, Mr. Sommerwerck will continue his discussion of the development of Ambisonics.*

## POWER SUPPLIES

*Continued from page 22*

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