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4 CHANNEL STEREO
From Source to Sound

By Ken Sessions

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Introduction

Superior sound—quality reproduction of music—has had a rebirth. Widespread interest in home music systems was generated when high fidelity became "the thing" in the fifties. The pace barely slowed when stereophonic sound arrive to push fidelity even further ahead. While many of us were still saying, "Well, all right, now where do we go from here?", a couple of people were just itching to show us and the rest of the world some of the things they had been toying with—things like passive rear channels to add a new dimension to two-channel listening pleasure, and like reproduction of four semidiscrete channels of information from a conventional stereo record.

It took a while for the eyes of an almost disenchanted public to come to rest on what the pioneers were doing; and it took even longer for the ears of the world to listen. But listen we did. And we liked what we heard. But by now there were too many companies attacking a single problem from too many directions. The result, while not quite chaos, was the next thing from it—utter confusion.

Today, there aren't many people who haven't heard of four-channel sound. There aren't many more who haven't actually heard a demonstration of quad in a local hi-fi store. But few indeed is the number of people who can tell you what basic types of four-channel sound exist, which of the systems are destined to failure and which are zooming for the successful position at the top. There aren't many people who know the difference between discrete and matrix systems, for example, or between an active and a passive quadraphonic decoding network, or between synthesis and reconstruction. Most audiophiles who would like to get a four-channel system going feel a bit helpless about it, for they know they are at the mercy of the fellow behind the counter of the store where they buy their components. And audiophiles like to be in the know;

they like to judge one system against another and make a buying decision based on their own knowledge—not someone else's.

Well, that's what this book is all about. It is an in-depth look at four-channel "surround sound," from every viewpoint, technical and nontechnical. With the information in these pages, you will be able to compare the various approaches and select the one that meets your own particular needs. Best of all, you'll not be lulled into a position where you are saddled with a roomful of obsolete stereo equipment—at least not without your eyes wide open.

Ken Sessions

Chapter 1
It Started with Stereo

By comparison with such brain children as the electric light and the telephone, high fidelity might be called an infant of breech birth. Maybe that's as it should be, for surely the world avidly sought the illumination of cities past sundown and the means for instant communication between any two people almost anyplace, any time. High fidelity sound, however, filled no such gap; it is, was, and will be first and foremost a means of entertainment—period.

Had talking movies required families to invest hundreds, perhaps thousands, of dollars, as hi-fi has done, it is doubtful the world would ever have laughed with Laurel and Hardy or smiled and cried with Shirley Temple. For then the era of the talkies would have been delayed until the time when people were willing and able to dole out large chunks of cash to be entertained, which didn't happen until after World War II.

Hi-fi was a plodding development chiefly because of semantic questions. To some, hi-fi was born in 1877, when Edison heard a nursery rhyme played back with his own voice. To others, it was eleven years later, when Berliner invented the phonograph record. The chief difference between Edison's Graphophone and Berliner's Gramophone was fidelity. Berliner's disc and handcranked "turntable" reproduced sounds louder and with less noise; i.e., its fidelity was better.

EARLY DEVELOPMENTS

Perhaps "high fidelity" was born when the first voice was transmitted by wireless and detected with a coil, a hunk of quartz, and a primitive earphone. But I like to give Edwin Armstrong the title of "Father of High Fidelity," because he was among the first to be concerned with faithfulness of reproduction, and he was indisputedly the first to achieve a measure of success in upgrading the quality of sound reproduction.

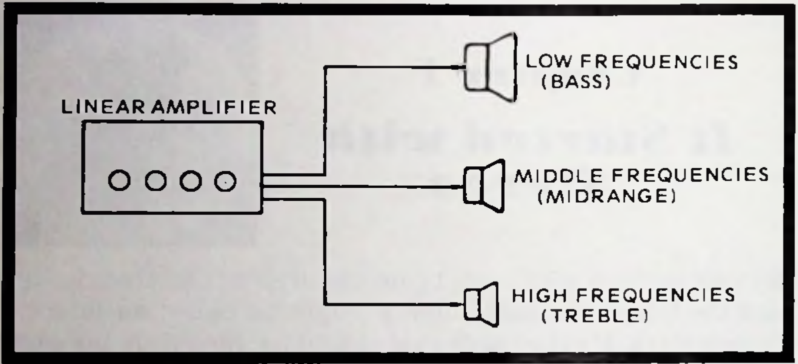


Fig. 1-1. A high fidelity system usually reproduces a wide range of audio signals by linear amplification of wideband signals, and by feeding selected portions of the audio signal into speakers particularly suited to those frequencies.

Armstrong "discovered" the art of frequency modulation (1933) in his search for a better way to transmit voices by wireless means, and even built a complete broadcast station to prove its worth. With FM, the inherent static and noise problems of AM were eliminated, and the foundation for high-fidelity transmission was laid. Coupling the capabilities of FM with enough frequency spectrum to "stretch out" a bit resulted in virtually flawless airing of live performances, at least in the range of audible frequencies.

Armstrong's contributions were just beginning, though. Even though a millionaire, and in his declining years, at 63 he was awarded the first patent for FM multiplexing! And with that offering, he gave the home music entertainment industry the shot in the arm that was to give it enough strength to grow into a full-fledged giant.

It is unfortunate that the term "high fidelity" was adopted in the early fifties—because it attributes the term with a meaning it really should not have: a high-performance single-channel audio system. Actually, hi-fi is a relative term, and a music reproduction system can be said to have high fidelity only in comparison with another system. What we knew as high fidelity in the late forties would be unacceptable to us today. Grunow's floor-standing AM radio was a masterpiece of audio engineering in its day, but it offered no challenge to the ear's upper-frequency-limit capability.

Hi-fi eventually came to mean a reproduction system with a capability of linearly amplifying a wide range of audible frequencies, and reproduction of those frequencies with speakers especially designed to accept signals in limited-range subbands, as shown in Fig. 1-1.

TWO CHANNELS

Regardless of how well a sound system works, it has built-in limitations if it is restricted to a single amplifier-speaker chain. Good-quality, single-channel sound reproduction can be likened to listening to a live concert performance through a hole in the auditorium wall. Delicate, subtle sounds are lost when they occur simultaneously with big, full-bodied sounds. A phenomenon of human hearing known as "masking," coupled with the swamping effect whereby a single speaker tries vainly to reproduce two (or three or a hundred) instrument sounds simultaneously, causes the ear to be inundated with a melange of melted melody. The listener hears the tune, but not the orchestral instruments that produce it.

This "drawback" to early hi-fi did not go unnoticed. A number of experimenters made similar observations and scurried about looking for a practical solution. The practical solution, of course, was the obvious one: expose the listeners to more than one sound source.

Two cameras spaced as eyes are can each take a picture of a nearby object, and when one camera image is fed to one eye and the second image if fed to the other eye, the viewer sees a real, honest-to-goodness three-dimensional image. The two images fool the brain by simulating the viewer's own parallax. Why then, the experimenters theorized, couldn't the brain be fooled with ear "images"?

The answer to this question was binaural sound.

Binaural Sound

To record binaural sound, two microphones were used; they were spaced about seven inches apart, and each pointed away from the other. Often the two microphones were separated by a small globe that represented the head of listener, as shown in Fig. 1-2. The theory was that sound would reach the two microphones in exactly the same way that sound would reach the ears of an individual at that spot. As a result, the recorded signal should be exactly the same as what the on-

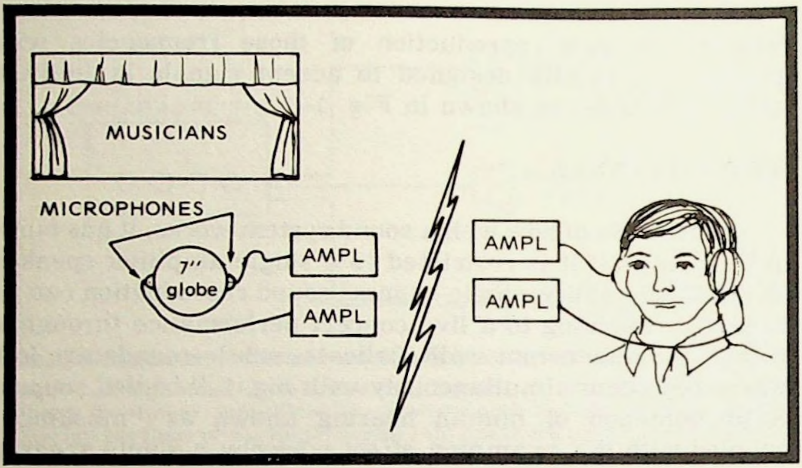


Fig. 1-2. Binaural sound was likened to binocular vision, where two enlarged eye images are combined into one by the human brain. Two microphones, serving as ears, record the concert sounds the way they'd be heard by an actual concert attender. On playback, the signals are fed to the listener's ears as two discrete channels.

the-spot listener would hear. On playback, of course, the signal recorded by the left microphone would be heard by an earphone at the listener's left ear; the right signal would be played back through the earphone at the listener's right ear.

The concept worked in fact as well as in theory, and "binaural listening" enjoyed a brief surge of popularity. There were a few problems, though, that kept binaural sound from attaining the enormous popularity that stereo was to claim. For one thing, listening required earphones. The differences between the output channels were very small, but earphone listening kept the differences significant enough to be noticeable. When the binaural signal was played back through a two-channel amplifier-speaker system, it sounded very much like a conventional monaural program.

Suppose there were a large wall between you and an auditorium where a concert was taking place, as in Fig. 1-3. If you could put one ear against a hole in the wall, you'd hear the concert in good quality, but monophonically. If there were a head-shaped mask built into the wall so that each ear had a hole of its own through which to obtain concert sounds, the signal would be lifelike and more or less three-dimensional. It would be binaural. But suppose now that you were to listen to those holes with your ears about ten feet back from the mask.

You can see that the binaural effect would be lost; the sounds through the two holes would merge and become one between the loudspeakers and the listener's ears. Masking was another problem with trying to hear binaural signals through ordinary loudspeakers. A soft subtle sound could easily be detected in one ear if that ear were protected from the dominating sounds present at the other ear. A nice easy job for headphones, but an impossible task for a pair of speakers without some electronic hanky-panky. There's no getting around the fact that the ear cannot hear a sound in the presence of a much louder sound of another frequency when both signals are phased together.

We'll talk about phase in greater detail later. For the time being, think of it as being synchronization of sound. Sound is vibration—up and down, to and fro, zigging and zagging—a movement one way and then the other. Two sound sources are in phase if they move together. If one source zigs while the other zags, the two are out of phase. Phasing plays an extremely important role in stereophonic reproduction, and an even greater one in quadraphonic reproduction. About the only types of reproduction where it is of little importance is monophonic and binaural sound.

Well, binaural sound drew a share of admirers from among the purist listeners, those who didn't mind the disad-

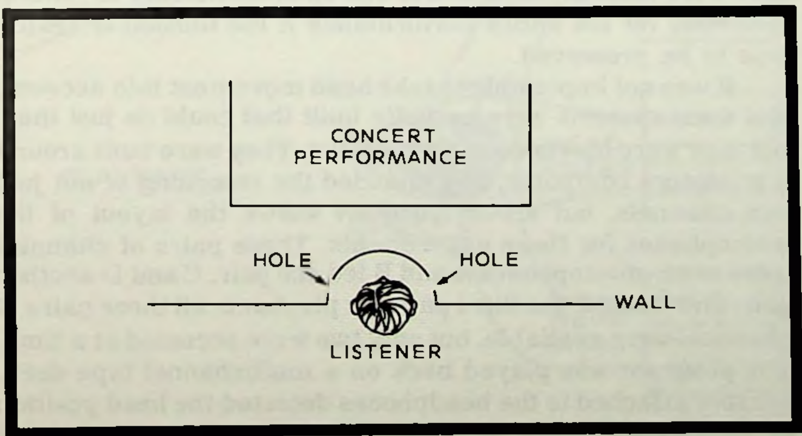


Fig. 1-3. Binaural sound is the equivalent of a listener hearing a concert through a pair of earholes; it takes earphones to reproduce the signal. If speakers are substituted for the earphones, the listener hears monophonically, as if he were standing back ten feet or so from the two earholes in the wall.

vantages of headphone listening. But the tendency of headphones to shut off the listener from outside disturbances proved as much a deterrent to enjoyment to some as it was an encouragement to others, and it reached a popularity peak that would be a disappointment to hi-fi manufacturers by the standards of today.

There was yet another huge disadvantage to binaural sound, even with earphone listening. As long as the listener sat, head riveted in a single position, the effect was soul-stirring, as if he were in attendance at a live concert. But when he moved his head, a very disconcerting thing happened: the performance moved with him. The stage always was positioned directly in front of his head, regardless of his own position. If he were attending a live performance, he would be able to glance from side to side, turn his head here and there, and with each head movement, his ears would receive slightly different signals—signals that would pinpoint the stage, localize the audience. To retain the lifelike reproduction, some method would be required to detect the listener's head movements, and supply a slightly different signal with each new position. Since the microphones had no way of knowing which way the listener's head was going to be turned and when, they just reclined helplessly in the position in which they were initially fixed, and it was up to the listener to remain immobile for the entire performance if the illusion of reality was to be preserved.

It was not impossible to take head movement into account, and some systems were actually built that could do just that; but they were horrendously expensive. They were built around a miniature computer, and included the recording of not just two channels, but six! Figure 1-4 shows the layout of the microphones for these experiments. Three pairs of channels were used; microphones A and B fed one pair, C and D another pair, and E and F the third pair. On playback, all three pairs of channels were available, but only two were accessed at a time. The program was played back on a multichannel tape deck. Sensors attached to the headphones detected the head position of the listener. When he turned his head to the left, signals from the A and B set were fed into his phones. With his head straight forward, C and D amplifiers were accessed. And a movement to the right caused the remaining two amplifiers (E and F) to feed the phones. The result was excellent, ac-

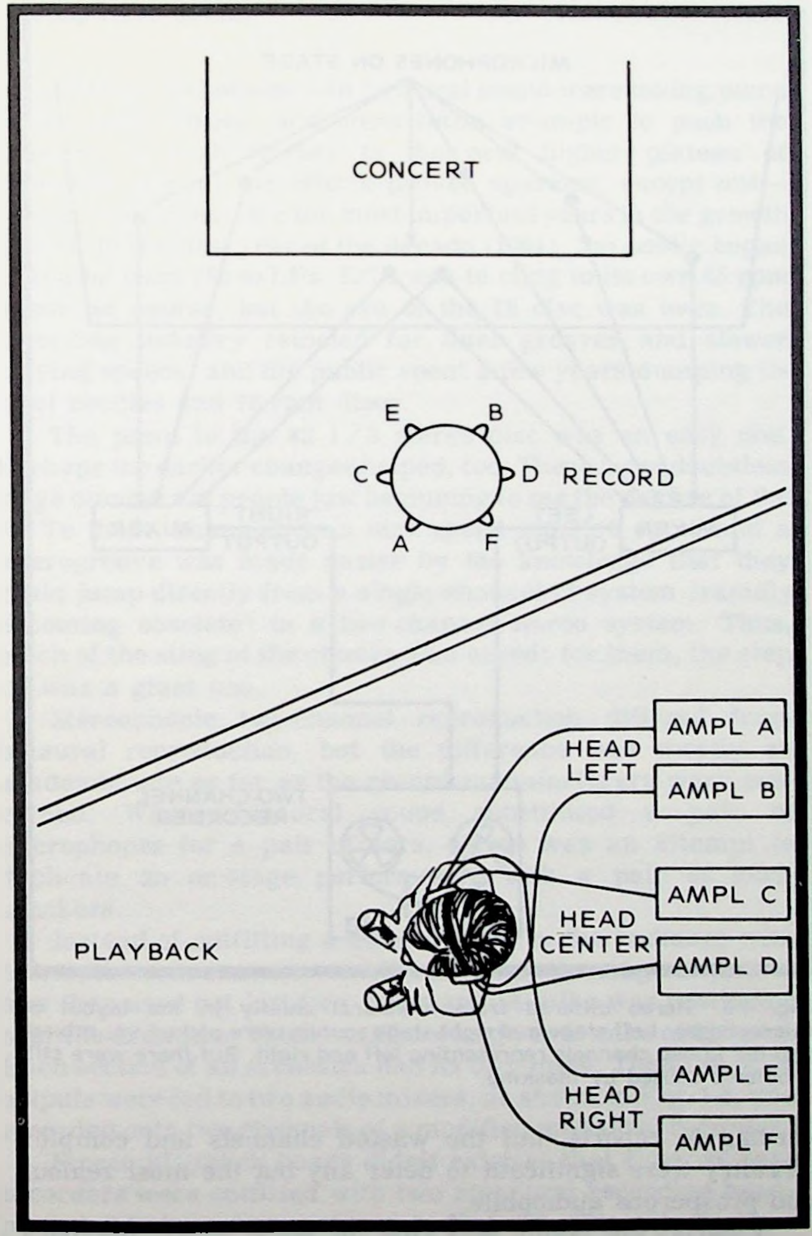


Fig. 1-4. One system of binaural reproduction involved the recording of six channels. The listener could turn his head left or right and receive signals roughly the same as if he were doing the head-turning at the live performance.

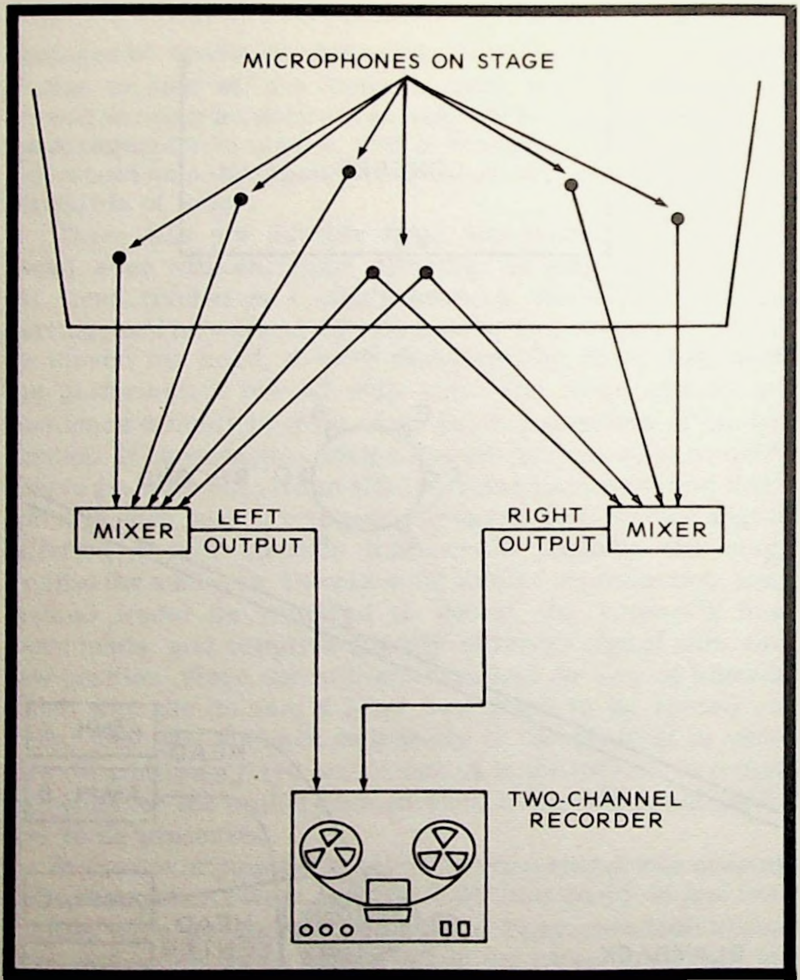


Fig. 1-5. Stereo differed from monaural chiefly in the layout of microphones. Left-stage and right-stage sounds were picked up, mixed, and fed to two channels representing left and right. But there were still problems caused by masking.

cording to reports, but the wasted channels and complex circuitry were significant to deter any but the most zealous and prosperous audiophile.

Clearly, the public was ripe for some music playback system that could offer both realism and a means for entertaining the entire family without the physical and social handicap of headphones.

All these dabblings with binaural sound were taking place in the early fifties, and were futile attempts to push the progress of high fidelity to the next higher plateau of evolution. But all the efforts proved spurious, except one—stereo. The fifties were the most important years in the growth of hi-fi. In the first year of the decade (1951), the public began changing from 78s to LPs. RCA was to cling to its own 45 rpm single, of course, but the era of the 78 disc was over. The recording industry retooled for finer grooves and slower playing speeds, and the public spent a few years dumping its steel needles and 78-rpm discs.

The jump to the 33 1/3 stereo disc was an easy one. Perhaps the earlier changes helped, too. There were doubtless large numbers of people just beginning to see the demise of the 78. To them, changing to a new speed (33 1/3 rpm) and a microgroove was made easier by the knowledge that they could jump directly from a single-channel 78 system (rapidly becoming obsolete) to a two-channel stereo system. Thus, much of the sting of the change was eased; for them, the step up was a giant one.

Stereophonic two-channel reproduction differed from binaural reproduction, but the difference was mostly an academic one as far as the record manufacturers were concerned. Where binaural sound substituted a pair of microphones for a pair of ears, stereo was an attempt to duplicate an on-stage performance with a pair of loudspeakers.

Instead of outfitting a dummy head in the audience with mikes, the sound men moved the mikes to the stage area. But now they used not just two, but many. A mike was positioned near the drummer. Each vocalist was given a mike of his own. Each section of an orchestra had its own mike. Then the mike outputs were fed to two audio mixers, as shown in Fig. 1-5, and recorded onto two channels of a modified monaural recorder.

Stereo playback heads didn't exist at that time, so tape recorders were outfitted with two monaural recording heads placed side by side, as shown in Fig. 1-6. The signals were staggered by the distance between heads, so the taped information for the right channel appeared earlier than that for the left. It was a "mickeymouse" approach, but it did prove

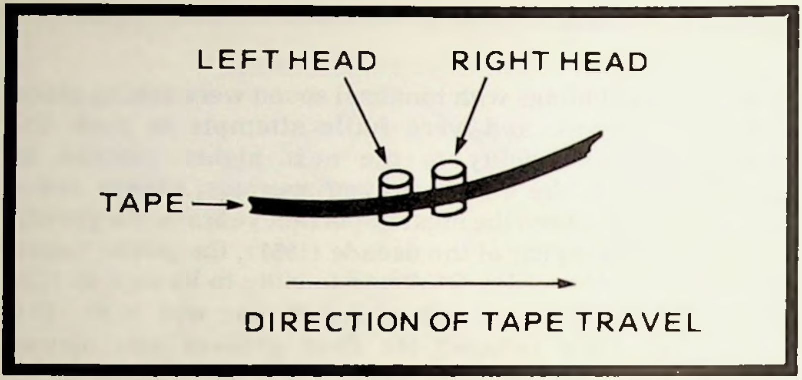


Fig. 1-6. "Staggered" heads recorded early stereo programs because there was no heads available capable of recording two channels simultaneously. To remain synchronous, the right track's signal appeared a few inches ahead of the left. The tape, traveling left to right, would pass over the left head first. On playback, the slightest variation in head placement caused degrading phase shifts; more serious misalignment caused an annoying "echo" effect.

workable pending development of heads with two-channel capability.

The multimicrophone-mixing technique proved viable too, though it did have some very serious shortcomings, the most prominent of which was the masking effect, mentioned earlier.

A series of important investigations were being conducted by major companies engaged in stereo recording during the fifties and early sixties. One of the curious phenomena observed in stereo's beginnings was that signal phasing plays an important part in the psychology of recreating mentally an original live program. Phased signals create easily located apparent sound sources between two loudspeakers. Unphased signals from two sources don't get "localized, " but they serve to "unmask" sounds of one channel that would otherwise be covered by the other.

The fruits of much of this research were to be useful later in enhancing the realism of stereophonic sound, and were to eventually form the basis for four-channel sound reproduction, but other developments seemed more important to music lovers, and acoustic research took a back seat temporarily to public demand.

Music reproduced through two amplifier-speaker systems sounded good. Cloudy melodies began to be discerned by careful listeners as individual instruments. The chief problem was not one of fidelity, for who really cared if a subtle nuance of sound were lost now and then if the overall clarity and trueness of tonal range were preserved? The main problem was a mechanical one: what could a music lover use for a source of sound other than a staggered-head tape playback unit?

The answer came just in time to save the phonograph record from extinction: the stereo disc. For a time, discs and tape competed heavily, particularly after the in-line two-in-one head was introduced. But the convenience of the record was virtually unchallengeable, so tapes were to lag behind turntables as a principal home-entertainment medium, at least for the time being.

With every development in stereo, sales in the industry pushed upward. People who were biding their time until they could determine whether or not stereo was "here to stay" were won over with each announcement of a new "breakthrough." There was something lacking, though: all stereo programs had to be played back mechanically by the listener.

MULTIPLEX

There were a few scattered events across the country that were responsible for winning large numbers of converts to stereo. Radio stations, AM and FM, cooperated occasionally to produce a show in stereo. Listeners were advised of the broadcast weeks in advance, and were told to place an AM receiver on the left and an FM receiver on the right, about five feet apart. Then, at broadcast time, curious people everywhere gathered around their AM-FM hookups to hear what the big to-do over stereo was all about. Many were highly impressed. Others, mainly those in areas of marginal reception, were disappointed. But the stereo story was being told in the only way it could be—by actual demonstration.

Later, broadcasters cooperated with local television stations to bring stereo telecasts with an FM receiver on one side and the TV set on the other. But all these goings-on were not to the liking of the FCC, which objected on the grounds that an owner of an FM set was being deprived of half the broadcast information if he was without a TV set. And TV audiences complained that they were cheated of half the music if they didn't own an FM receiver; they resented the fact that they were being "pressured" into buying a piece of electronic equipment they either didn't want or didn't need.

But Edwin Armstrong's invention was to come to the fore. In 1953 he was awarded a patent for stereophonically broadcasting two channels of information over a single frequency-modulated carrier. FM broadcasters could not rush headlong into full-scale stereocasting, though, because any modification to a transmitted radio signal involves communication with the FCC, which has the option of accepting or rejecting any proposed plan, depending on the relative benefit to the public.

To complicate matters, the FCC was being approached by proponents of several stereocasting methods; and only one of the methods was to be adopted because compatibility was one of the FCC's primary considerations. The system that finally won FCC approval was one that involved matrixing of the two channels in such a way that a non-stereo-equipped listener could hear the broadcast without suffering loss of either channel of information. This established the compatibility the FCC was looking for, and approval finally came. Manufacturers began full-scale production of multiplex adapters for use with existing FM receivers as well as all-in-one stereo-multiplexed receiver-amplifier combinations.

How It Works

Audio signals are alternating-current voltages just like the house current that operates your radios and toasters. With audio, the voltages are substantially smaller, though, and the rate of alternation varies constantly—in proportion to the audio pitch. With household power, the voltage is very large (117 volts or so) and the commercial power rate of alternation is fixed at 60 times per second (hertz) in the U.S.A.

An alternating current is called "alternating" because it reverses polarity many times per second. If you could stop the alternation at any time, you would be able to measure a direct-current voltage similar to what would be available from a battery. With stereo, there are two alternating-current voltages, one for the left channel and one for the right. These two signals are similar, because they belong to the same program, but they are not identical—otherwise both channels would have the same program information, which is nothing more than monophonic listening with two amplifiers.

If both channels are superimposed so that they are in phase with each other as shown in Fig. 1-7A, the output is a single signal carrying all the information on both channels. If both channels are superimposed so they are out of phase, the output is a varying voltage that represents the **difference** between the two signals (Fig. 1-7B).

In a multiplex FM transmission, the main FM radio carrier conveys the sum information so that people with monophonic receivers will be able to hear both channels with no degradation or loss of information. A 38 kHz subcarrier (a radio signal that is well above the range of human hearing) carries the low-level difference information. In the multiplex receiver, the two signals are superimposed again, and the difference channel is used to cancel the components of itself that exist on the main carrier. The resultant signal is reversed in phase and split into two signals. One of the two signals is amplified and fed to one loudspeaker system; the other receives a superimposition of the difference signal, which is now additive. This signal becomes the second channel of the stereo pair.

Antennas

Stereo multiplexing proved a new boon to the prosperous hi-fi industry, but FM owners who purchased multiplex adapters had to contend with a problem they had never been faced with before: they had to upgrade their antenna systems to enable the low-level subcarrier signal to be detected properly. Where a little hunk of wire dangling from the antenna terminals once proved satisfactory, now a complete antenna was required for the receiver. Where the FM stations were situated in a group at a good vantage point—a high building or nearby hilltop—there was simply the matter of splicing a couple of pieces of television "twinlead" antenna wire together, as shown in Fig. 1-8. But this antenna, though a good performer for reasonably close distances, proved unsuitable for ranges of 50 miles or more. And since it is a directional arrangement, it is totally unworkable when several stations must be received from various directions, unless it's mounted so it can be rotated.

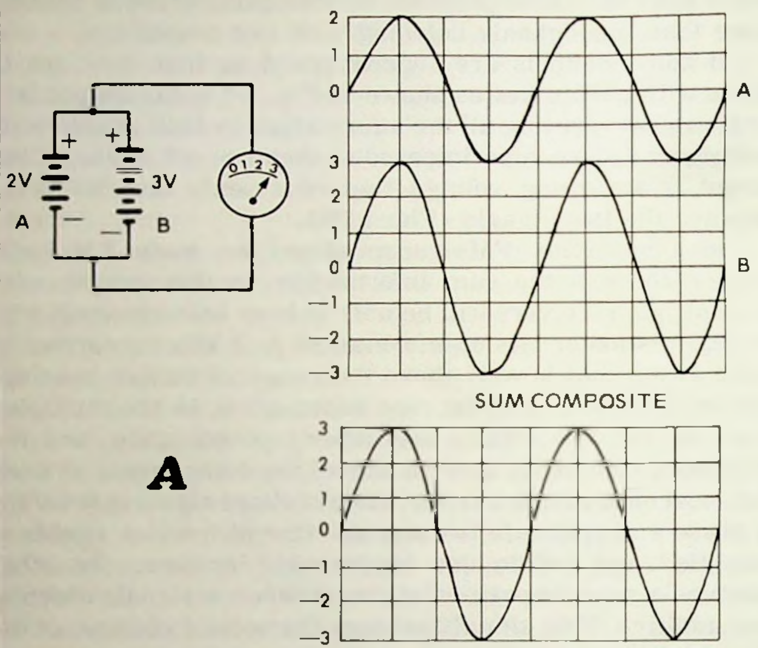
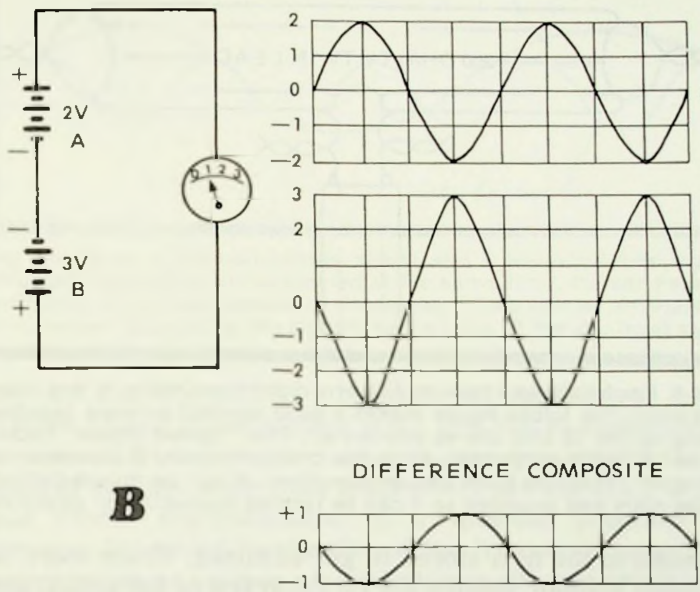


Fig. 1-7. When two audio signals are phased, the output is a single signal carrying total information of both channels. The battery analogy shows how two voltages can be superimposed to get a reading always equal to the stronger of the two cells. When two signals are **not** phased, the result is a signal carrying the information that is **not** common to both channels.

Those who minded least were the manufacturers of TV antennas. Since the FM broadcast band is situated smack dab in the middle of the VHF television band, there was little effort required for them to tool up for producing high-performance yagi-type antenna arrays.

Don't get the idea that a TV antenna will give you satisfactory performance on FM, or you're sure to be disappointed. It is true that the complete 88 to 108 MHz FM spectrum lies between TV channels 6 and 7; but the TV lower channels (2 through 6) are clumped between 54 and 88 MHz. The higher channels (7 through 13) will start well above the FM band. Common TV antennas have elements of two sizes,



The battery analogy illustrates why this subtractive method results in a lower signal voltage than the additive method. On playback, the difference signal is superimposed on the sum signal. The small voltage variations are subtracted from one channel through phase cancellation and added to the other by in-phase bolstering.

one for the low channels and one for the high. The FM band, with a wavelength between the two element sizes, cannot be covered effectively by either, except where the TV antenna is a log-periodic type, whose element size tapers gradually from that of channel 2 to channel 13.

SEPARATION AND LOCALIZATION

With FM now serving as a source for stereo programs, and with the ready availability of stereo discs for record players and tapes for stereo decks, the two-channel world was bristling with activity as audiophiles by the thousands

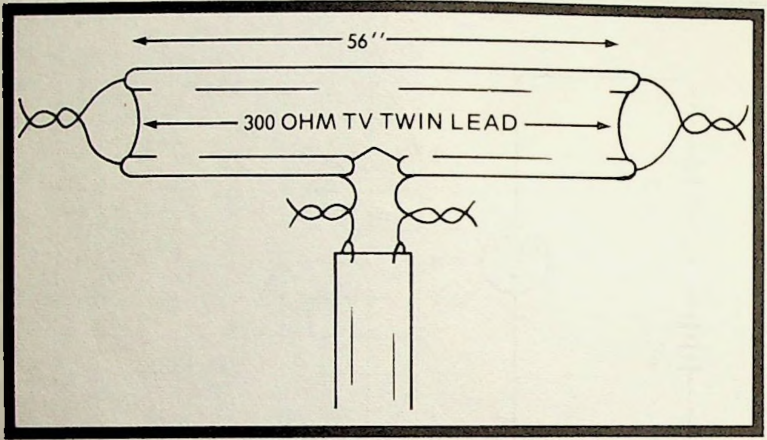


Fig. 1-8. For locations where an FM broadcast transmitter is less than 50 miles away, the folded dipole makes a good low-cost antenna capable of picking up the 38 kHz stereo subcarrier. This "folded dipole" receives best at an angle perpendicular to the cross member; it is worse than nothing for receiving from end-on directions. It can be thumb-tacked to wooden slats and mounted so it can be rotated manually for directivity.

swarmed to the hi-fi stores to get outfitted. There were still problems, though, despite the fact that few of the actual users had any knowledge of them.

One problem was that of direction (buyer education). Buyers hardly knew where to start with a component acquisition program. They could buy a preamplifier as one component, then purchase a separate stereo amplifier, a stereo multiplex tuner, and a pair of speakers. Or they could buy a combination stereo preamp and amplifier, then add a tuner later. Or they could buy a tuner-amplifier, which combined the three basic components (tuner, preamp, and amplifier) in one package. This was a compatibility problem more than anything else; a buyer of a separate preamp was out of the marketplace as far as the vendors of preamp-amplifiers and amplifier-tuners was concerned. It was to be a long time before a sense of direction in hi-fi and stereo manufacturing was to evolve. As a matter of fact, every possible combination of components is still available today; but at least there is now a general direction: the tuner-amplifier combination has finally achieved a "standard" status.

Another problem, and this was more serious than the first, was purely technical. It takes in-phase signals to give across-

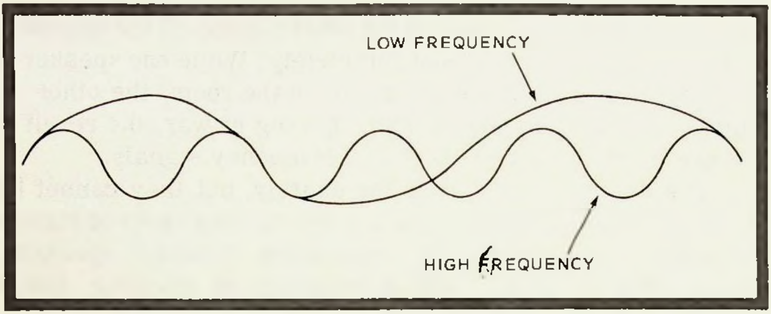


Fig. 1-9. When a high-amplitude signal and a low-amplitude signal of different frequencies are presented at the same time, the ear hears only the louder of the two because of "masking." This can be alleviated to a great extent by phasing the signals so that less of the low-level signal is masked by the louder.

the-wall sound localization between two speakers of a stereo system. Out-of-phase signals from two stereo speakers seem to emanate from two general areas, one over toward the right speaker and the other over somewhere toward the left. But a true stereo representation of a musical program must preserve the sound localization if the listener is to be able to hear sources at various places between the two speakers, not just dead center, but left of center stage and right of center stage as well. So in-phase playback seemed vital.

But in-phase signals introduce another problem—masking. Two signals occur at the same time and in the same phase, as in Fig. 1-9. If the frequencies of the signals are substantially different, the lower level signal will go unheard. One method of correcting the problem to some extent is by phasing the signals so they are not synchronous. If the low-level signal peaks occur during the high-level signal dips, the low-level signal can be unmasked to some extent.

But if the phasing changes are handled by the hi-fi listener, it boils down to a simple yes or no proposition: he either connects both speakers to the amplifiers in the same way to keep the phase the same, or he reverses one set of speaker leads to put one speaker 180 degrees out of phase with the other. What he has gained by unmasking, he has lost by sound cancellation of lower frequencies.

A speaker moves in two directions, fore and aft. When both speakers move forward at the same time, they can be said to be in phase. If one moves aft while the other pushes forward, they are out of phase. It is easy to see, then, that if one is connected one way and the other connected the opposite, some frequencies will be lost completely: While one speaker is pushing, trying to compress the air in the room, the other is pulling, trying to rarefy it. With this tug of war, the result is silence, particularly on very low-frequency signals.

There are compromises, fortunately, but they cannot be made at home—the answer is in varying the phases of signals to result in an optimum separation between speakers, localization of signals, and a minimum of masking. But a compromise is a compromise.

In the early days of stereo, both speakers were phased together, and what was recorded was exactly phased between the input microphones. Listeners could thus hear not only cute table-tennis matches with one speaker pinging after the other ponged, but dead-center clapping and other similar demonstrations. If the recording artist made a sharp sound that arrived at both channel microphones at exactly the same instant—and both microphones were phased identically—the stereo listener heard that sharp sound emanating from a point precisely between his two speakers.

But all that was sensationalism, and the public tired of it quickly. Before stereo grew whiskers, the mode became at least partially sophisticated. Record producers became concerned more with quality and oneness of sound and less with special effects. But not everybody was happy, because the new sound sometimes left a "hole in the middle" that listeners didn't dig.

Articles in stereo magazines appeared which showed special speaker interconnections to add a "phantom" channel between the speakers. It was crude, but it did the trick for many. The phantom was simulated by attaching a third speaker to the taps of both amplifiers, as shown in Fig. 1-10.

If the information applied to both amplifiers was in phase —which was most often the case—the signal level at both 4-ohm taps was roughly the same, either a positive value or a negative value. With two negative values applied to a set of speaker terminals, the speaker will see the least-negative signal as a positive value. But the least-negative of the two signals would still only be a very small value below the negative value of the other speaker, so the total signal applied to the phantom was considerably smaller than the value applied to either of the two sides (both of which received a legitimate positive and negative value). If both side signals were positive, the opposite case occurred, and the center speaker interpreted the least-positive signal as a negative value. In virtually all cases of in-phase signals, the center channel received but a fraction of the signal sent to the two main speakers.

During the time when all this concern for a center channel was taking place, several manufacturers hit upon the logical idea of doing the audio combining early in the amplifier. Harman-Kardon and others provided preamplifier outputs for a center channel; it was designed to be used with an add-on single-channel amplifier. This would have worked out very well, because now the center channel would have all the gain necessary to bring it into the perspective necessary to preserve the across-the-wall solidarity that everyone esteemed so highly. But it was a matter of too little too late, for

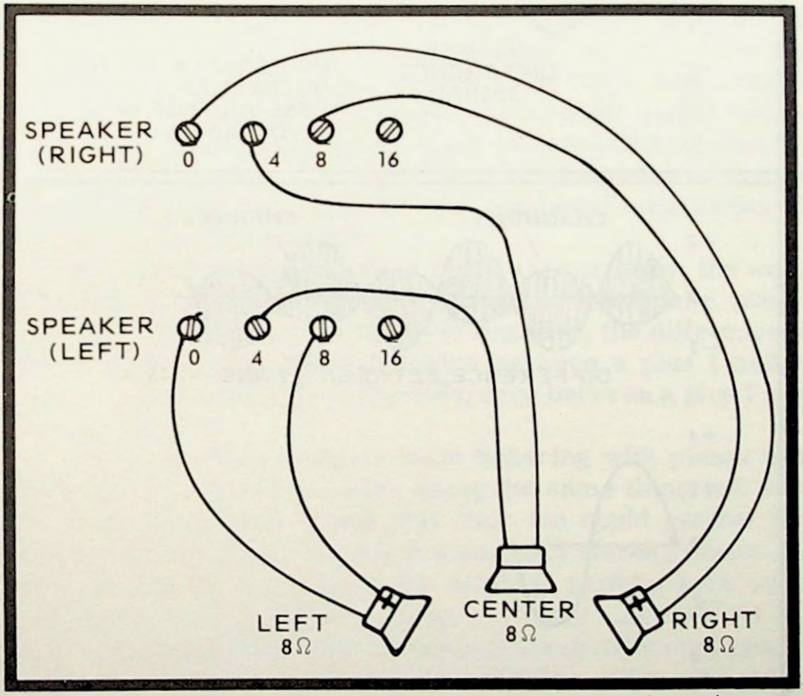


Fig. 1-10. By connecting a speaker between the two channels as shown, a center channel can be simulated. The volume of the center channel will be less than the sides because the signal is subtractive rather than additive; that is, it represents the difference signal between the two amplifiers. But it is an accurate representation of the material front and center.

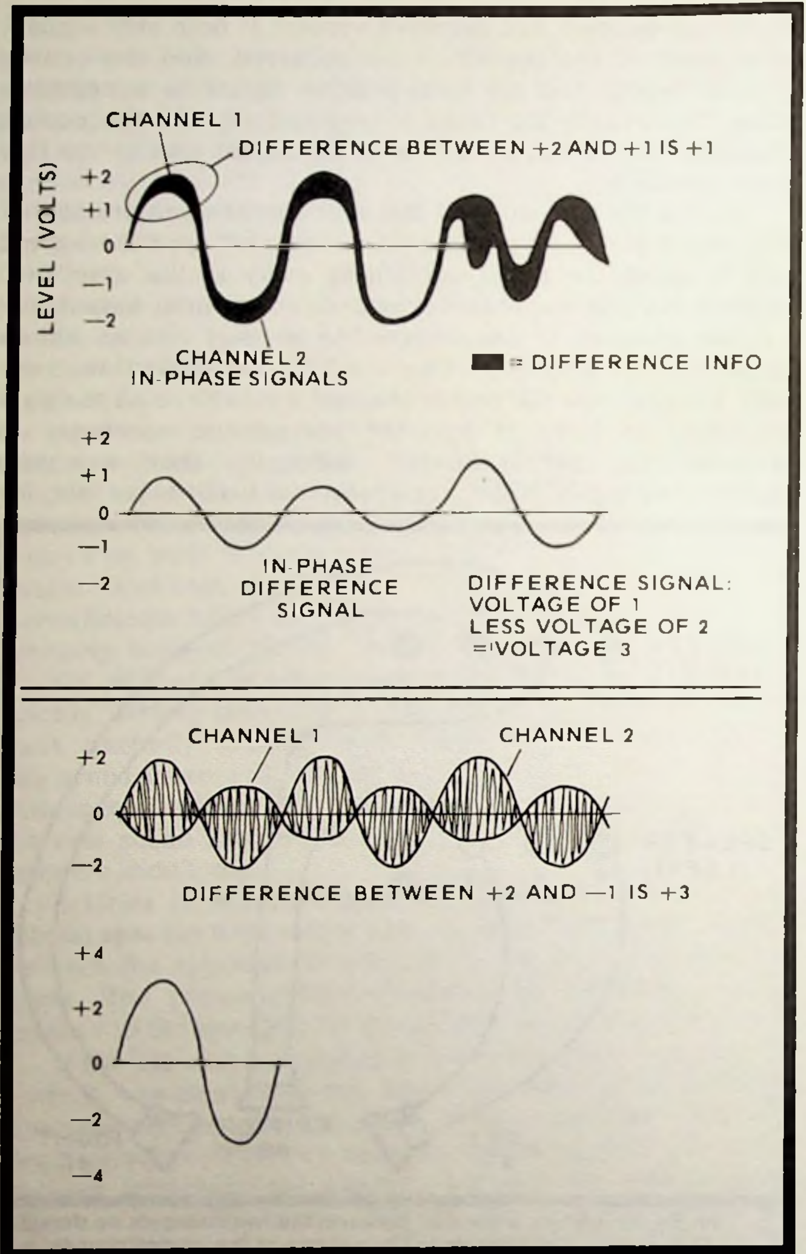


Fig. 1-11. "Difference" signals may be derived from two existing channels of information, and these signals can provide startling effects when coupled to loudspeakers in an existing stereo system.

the manufacturers had pretty well settled the question for themselves by producing stereo amplifiers rather than individual amplifiers. The person who wanted an additional amplifier to use for his center channel was out of luck unless he wanted to pop for a complete new stereo unit and only use half of it. It just never happened.

Even so, the concept of a phantom channel was to haunt a few diehard manufacturers and individual experimenters until they could turn it into something really worthy. Here were two individual channels, both with information that is interrelated with one another. With those two signals, another signal could be derived—that of the center channel. A little figuring shows that even **four** different signals can be obtained from **two**!

Take a left-channel and a right-channel signal, in phase with each other, as an example (Fig. 1-11). The voltage difference between the two signals represents a signal all by itself, lower in amplitude than either of the two. Now take the same two signals out of phase with each other. The voltage difference between the two results in another signal whose amplitude is higher than either of the two contributing signals. This signal is a difference signal, too, but it is referred to as a sum signal because it contains the combined information of the left and right channels.

The explanation is shown in Fig. 1-11; normally, the word "difference" implies that a subtraction must take place. However, when one of the values is negative, the difference is obtained by adding. The difference between a plus 1 and a minus 1, for example, is 2; the difference between a plus 1 and a plus 1 is zero.

While the experimenters were tinkering with pluses and minuses, record makers were doing the same thing but with different tools. They found that they too could control the presence or absence of sounds in a playback stereo system. All they had to do was record an original performance on a multitrack tape recorder, using as many channels of information as possible. Indeed, many of the professional record cutting specialists went to 16 tracks of information, and a few even went to as many as 24.

When it came time to mix the multitudes of channels down to a mere two, the recording engineer had the option of controlling the mixdown by delaying any track by a microsecond or two to avoid the masking effect. Of course he could increase the level of a vocalist or drop the level of the pianist or tuba player, too, if he wished. The point is, he brought a great deal of precision to the record-making business; and it was chiefly through his exercising of control over the phases of the recorded signals that enhanced the experimental work being done by Feldman and a few other audio researchers.

Look again at Fig. 1-11 to see why. The difference signal in the in-phase case is small. But suppose the phase of one of the two signals changes abruptly. Now the two outside speakers see the same level as before, but the center channel sees a signal louder than the other two. And the listener can localize that sound as emanating from the point where that speaker is situated.

Simple? You bet. And it is the foundation of four-channel sound.

Chapter 2

Surrounding Yourself with Sound

There is a vagueness about the source of a pair of out-of-phase signals. When a set of stereo loudspeakers is being fed out of phase—as, for example, when the supply leads are reversed at one speaker—the listener hears the sounds as emanating from nondescript areas rather than from across the front.

This characteristic is thought to be linked in some ways to a mechanism within the human hearing structure whereby the brain identifies in-phase sound with front sources and out-of-phase sound with rear sources. You can experiment with this phenomenon by listening to a stereo program with earphones (or with speakers placed at either side of your head, and aimed inward, as shown in Fig. 2-11).

When the speakers are phased, the sound will seem to come from a hemispherical area that describes an arc from speaker to speaker, and directly over your head. The same is true for headphones. When the polarity of one of the reproducers is reversed, the sound seems to move backward, and the arc melts away. With speakers out of phase, of course, there will be a noticeable loss of low frequencies because of cancellation, particularly where the distance between the two ears (around the head) is very short with respect to the wavelength of the transmitted signal, but the apparent rearward movement is just as distant.

All this variation in apparent sound sources depends to a great extent on psychology. The fact that you know there are two speakers beside your head may tend to hamper the illusion, optical or otherwise, but if you persevere, you'll find the "funny things" happening as indicated. (The frequency loss, of course, is no illusion at all, can can be measured objectively.)

Perhaps other manufacturers produced equipment to take advantage of the psychology of phased listening as well, but Dynaco's effort is certainly the best known. Dynaco produced

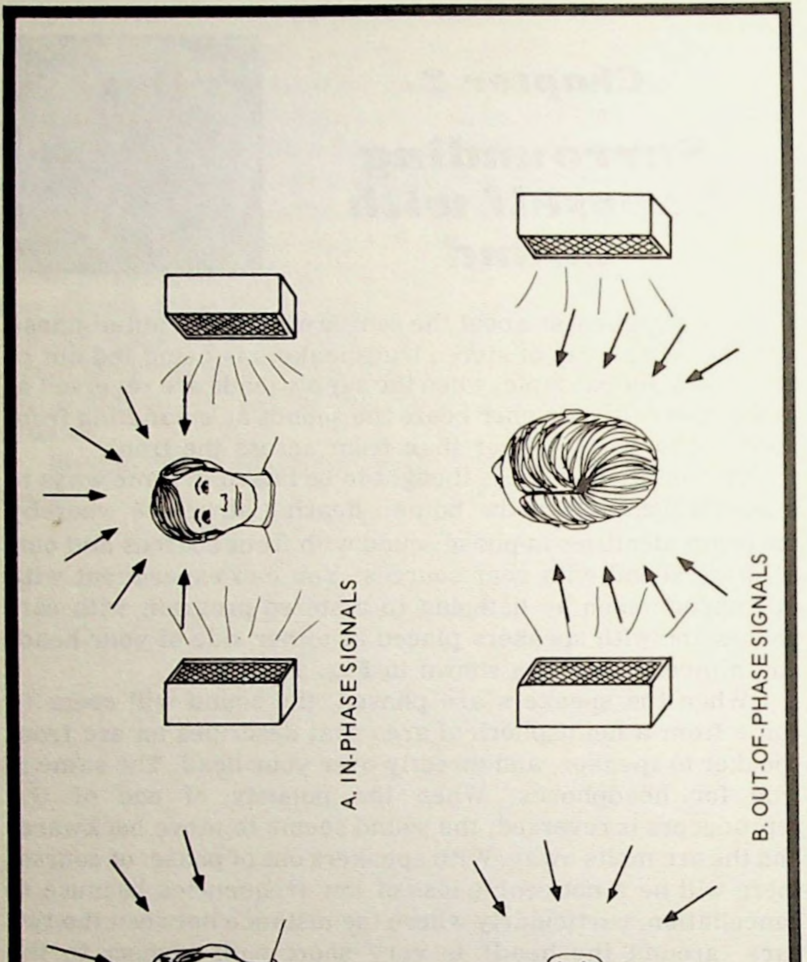


Fig. 2-1. In-phase signals from two sources (sketch A) seem to come from a 180-degree area that passes over the listener's head (and, with earphones, through it as well). Out-of-phase signals (B) seem to come from two rearward areas. (Arrows show "apparent" sound source.)

an amplifier with a blend control that was made to feed part of the right channel to the left side and part of the left channel to the right side. This "blend" control allowed phase corrections to be made by the listener, enabling him to improve the across-the-wall effect. But there was one other advantage, too: it allowed connection of a rear speaker to the front terminals so that cross-phased information could be utilized to add an extra dimension of spaciousness to the program material.

The concept is shown in Fig. 2-2. A stereo amplifier is connected to the front two speakers in the normal manner, and a third speaker, situated at the rear of the room, is connected to the "hot" terminals of both speaker outputs on the amplifier. This circuit arrangement works with any stereo amplifier, but under normal conditions the rear-channel level is very low. With a "plus" signal of approximately the same level applied to both sides of the rear speaker, the total voltage developed across the rear speaker voice coil is quite small. However, if an out-of-phase signal gets fed to either of the front two speakers, the voltage increase at the rear speaker is such as to make the sound particularly loud momentarily. This arrangement allows certain previously lost signals to

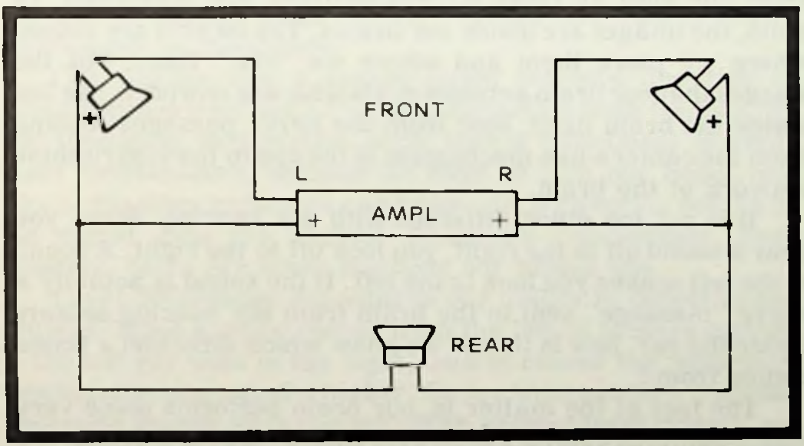


Fig. 2-2. When a loudspeaker is connected to the "plus" terminals of the two front speakers, and placed in the rear as shown, it reproduced an out-of-phase signal equivalent to the difference between the two front speakers. Dynaco's "blend" control allowed cross phasing between the two front speakers to some extent, which served to increase the level of the rear speaker. At optimum settings, some signals that seem totally veiled (unheard) can be heard distinctly in the rear.

stand out loud and clear, though they seem to appear from the rear of the room.

As mentioned earlier, however, the ear likes to attribute out-of-phase signals to the rear anyway, and the result is pleasant, and very similar to the sounds one might expect to hear in a large concert hall, where sounds reverberate and reflect back to the listener's ears a split second later than they arrive from the front. (A time delay is the same as a phase delay.)

Dynaco's unique "blend" control heightened the illusion even more because it allowed a certain amount of "dephasing" between the two front channels, thus intensifying the signal in the rear.

Probably now is a good time to pause briefly and learn why signal phase is such an important element in our hearing. It gets out of the realm of four-channel sound, and more into the realm of psychology, but it is very real and very universal.

THE DIRECTIVITY OF SOUND

When we see, we seem to perceive images spaced as far from our eyes as those objects actually are positioned. In truth, the images are inside our brains. The objects are indeed where we place them and where we "see" them, but the images that our brain actually deals with are nowhere else but inside the brain itself, sent from the nerve passages leading from the camera-like mechanism of the eye to the labyrinthine network of the brain.

It is not too much different with our hearing. When you hear a sound off to the right, you look off to the right. A sound to the left makes you look to the left. If the sound is actually a nerve "message" sent to the brain from our hearing sensors inside the ear, how is it that we know which direction a sound comes from?

The fact of the matter is, our brain performs some very highly complex computations every time we hear a sound. We use such unlikely functions as frequency, phase, level, and time (which can be different than phase, as we shall prove.) But more than that, we use comparisons of each function on a left-ear-versus-right-ear basis as well as combinations of functions.

It is easy to see that a person with hearing in one ear only has considerable difficulty in ascertaining the direction of a sound source, for he has no means for pitting one ear's perception against the other. He must rely on nothing more than combinations, such as frequency versus level, and if he can't compare this with a memory of a similar sound, he is simply helpless. Yes, the man who hears in only one ear can tell sound direction by memory comparisons, but it is a learned talent, and it sometimes takes years before he can develop it to a useful degree.

All audio waves are directional in character, but humans are more sensitive to the directional effects of high-frequency sound than to low-frequency sounds. If you have a hi-fi set, turn it on and increase the treble control all the way. As your ear passes in front of the speaker, you'll have no trouble at all in determining the location of the tweeter behind the grille cloth. If you back away some distance from the speaker, you can turn your head with your eyes closed and notice the rapid falloff of high frequencies when your ears are not oriented in a direct line with the tweeter. Plug one ear and you'll notice the effect even more. With your good ear turned toward the speaker, you'll hear whatever hiss or high-frequency scratches might be present on the program material. But turn your ear away, and the highs will just disappear. This is sound localization by frequency.

Can you see how a person might perceive direction of sound sources if he had no other information than frequency? But frequency information alone makes directional discernment cumbersome, because we have to turn our heads one way or the other before we can know which direction results in the signal with the highest frequency components. Fortunately, there are other methods our brains use.

Level differences are vital to our ability to detect sound sources. When a sound comes from the left, it is more intense in the left ear than in the right; and of course the frequency components are usually higher. So we immediately pinpoint the sound source as to our left. With normal-level signals, we cannot often consciously tell a difference in level, particularly if the sound signals (there are always two signals if we have two ears) are less than 1 dB different in apparent level. But even if we can't consciously tell the difference between an 8-watt signal and a 10-watt signal, our brains can. The nerve impulses that travel from our hearing sensors to our brain are directly proportional in level to the actual signal levels we perceive.

The two principal remaining determinants of direction in hearing are phase and time, and these are very closely related. A signal that is 180 degrees out of phase with another signal of the same frequency is either slower or faster than the other signal by a time corresponding to how far the sound travels during the period it takes to propagate one-half a wavelength at that frequency. That's pretty confusing, isn't it? Let's state it another way.

Every sound pitch has a wavelength all its own. The wavelength might be many feet or it might be inches. It is fairly easy to calculate the length of any sound wave, though, if you remember that all sound travels at a speed of some 1130 feet per second in air at normal room temperatures. The frequency of the sound tells you how long a wave lasts; and once you know that, you know how long it is by how long it takes for the complete wave to pass your ear. A 50-hertz (Hz) tone (which is the same as a 50-cycle-per-second tone) lasts one fiftieth of a second. That means it takes a fiftieth of a second to go past your ear, from start to finish. If you could stand back somewhere and watch those invisible 50 Hz sound waves, you'd see fifty of them leaving the loudspeaker in a one-second period. And of course the first one to leave the speaker would be 1130 feet from the speaker at the end of the one-second period. So each wave is one-fiftieth of that total distance of 1130 feet, and a 50 Hz tone (or an individual wave of such tone) is just over 22.5 feet long.

Now, if both speakers of your stereo are reproducing 50 Hz tones at the same time, and one of the two speakers is out of phase, it doesn't mean that one will arrive at the 1130-foot mark before the other, but it does mean that the two signals will appear different at that mark at that time; and they'll appear different at all other points along their path. This is because one speaker began the note by pushing the air in front of the speaker cone, and the other began by rarefying the air before the cone. The first hundredth of a second the first speaker pushed, and the second hundredth of a second, it pulled. But during the second hundredth of a second (one-half wavelength), the second speaker started pushing, and the two speakers remain forever doing just the opposite thing. If you could see the two waves, they might look something like the sketch in Fig. 2-3. Note that one appears to lag behind the other. Indeed, if the wave could be moved forward or backward just exactly one-half wavelength, both signals would look identical. But then there might be a time differential even though the speakers were phased.

The time and phase subleties are the factors that help us to know whether a sound originates behind us or in front of us. If the difference were strictly level and frequency (or level or frequency), we would always perceive sounds as being straight off to the left or straight off to the right. We wouldn't be able to tell when one sound was slightly behind us and above or below our ears, and we could not determine which sounds were coming from in front of us.

Let's look at some practical examples of sounds. Suppose you hear a truck passing by and you're standing somewhere near the road, with no obstructions. The truck generates a complex waveform that doesn't even come close to resembling the sine wave in Fig. 2-3. What we hear are a broad spectrum of tones impressed one upon the other, some of very low frequency and others of very high frequency. If both ears hear exactly the same signal, the truck is passing either directly in front of you or directly behind you—no ifs, ands, or buts. You'll know whether or not it's in front or in back by the clues you've received before the truck gets close enough to

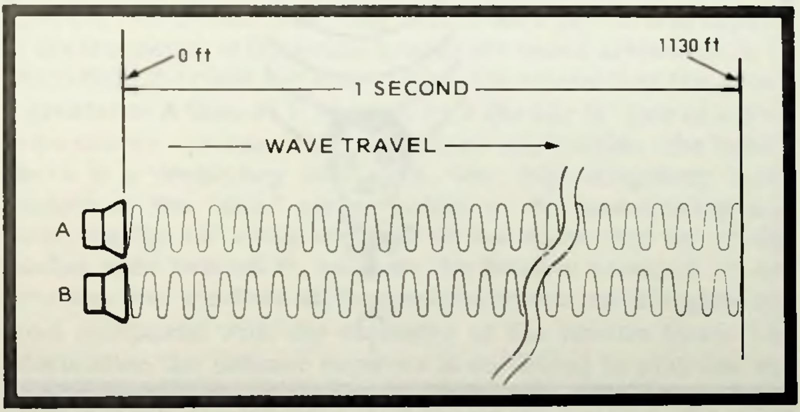


Fig. 2-3. When signals from one speaker are 180 degrees out of phase with the other, the result is a pair of outputs that are equal but opposite, as shown. If speaker A could be fed the same signal 1 / 100 second later than or sooner than B, the signals would be exactly the same, and perfectly phased.

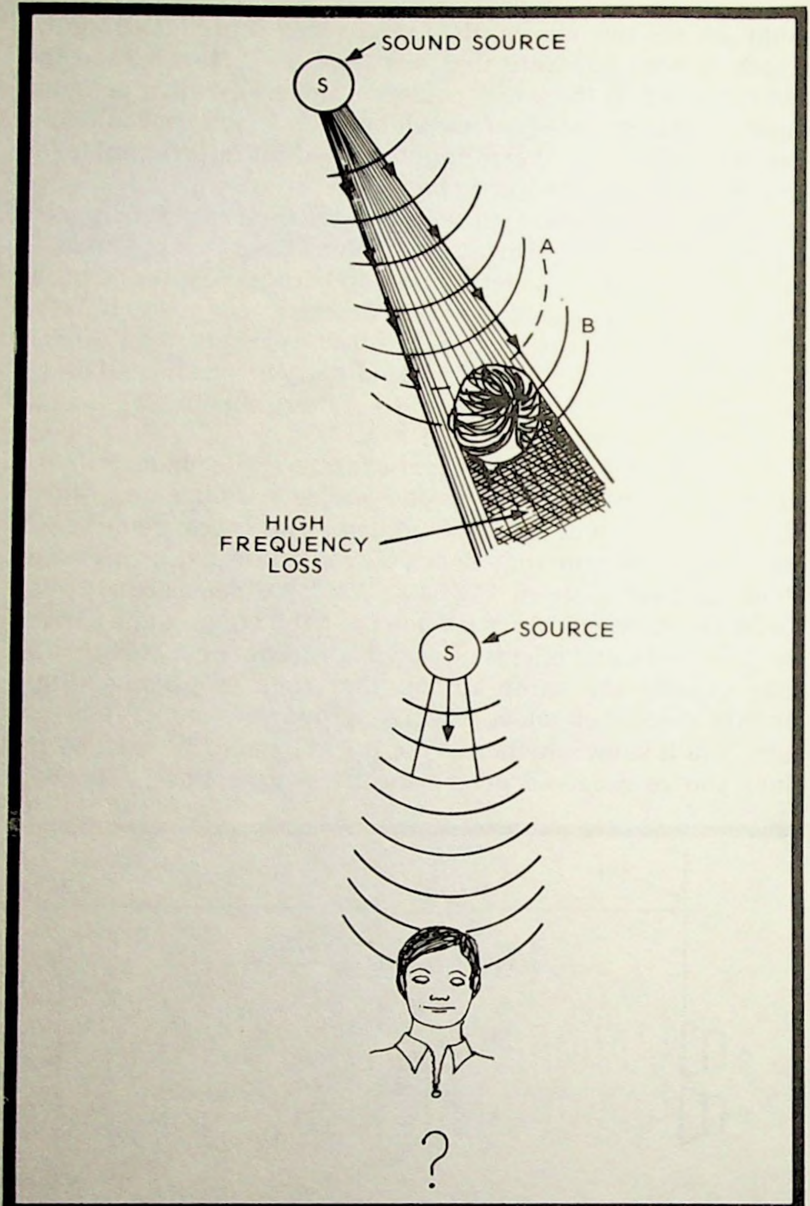


Fig. 2-4. The brain accepts information we aren't even aware of to tell us where sounds originate. These data elements include mental (unconscious) comparisons between sound levels, frequencies, phase, and timing. If we deprive the brain of difference material, it cannot decide on the source.

pass directly opposite you. If the truck was approaching from your left, and passed in front of you, your left ear would have heard the signal minute fractions of a second before your right ear, and it would have heard higher frequencies than your right ear received. If the truck passes in front of you, the right ear starts getting the high frequencies at a specific time period after the left ear gets them, and that time period is shorter than if the truck were passing behind you. Also, as the truck passes, the high frequencies get very intense if it passes in front of you, but the intensity drops ever so slightly if it passes behind you.

All along the way, your brain is calculating, computing, with lightning rapidity and below your level of awareness—and you get accurate data. Accurate? Not always.

Try this experiment. Close your eyes and sit in the center of a room. Have someone turn the volume up on your hi-fi to mask any other sounds that you might otherwise hear. Then, when you're well situated in the center of the room, have a friend stand immediately in front of you or behind you (or above you) and clap his hands sharply. Do you think you will be able to tell where your friend is standing?

Look at the sketches in Fig. 2-4. Each arrow represents the path of a sound wave at any frequency. The waves are represented by arcs. In the first sketch, sound waves reach point A first, and cause a hearing sensation in the listener's right ear; an instant later (the actual time period will depend on the frequency of the sound heard) the sound arrives at B, at which time the right ear hears. Also, the intensity of the sound is greater at A than at B because at A the ear is "line of sight" to the source, whereas at B, there is an obstruction (the head). There is a frequency loss at B, too; high-frequency components in the sound arrive easily at A, pressurizing and rarefying the air along the path of the wave—but not at the shaded area beyond B, because the head is creating an obstruction that shadows all frequencies whose wavelengths are short compared with the diameter of the human head. The information the listener receives is sufficient to pinpoint the source of sound to a high degree. There may be several spots on a vertical plane from which the sounds could be originating, but the listener's knowledge of his own environment provides clues, too, so there is seldom any question as to the source of the sound.

In the sketch, the sound source is directly above the head of the listener. This provides a very difficult decoding job for the poor fellow's brain because both ears are receiving the same information. The frequencies are the same, the timing and phase are identical. Both ears hear the same volume level. If the man is standing in a room with a low ceiling and the sound is that of another voice, he will automatically hear the sound as if it were originating from the rear (assuming he has his eyes open and can see no speaker before him). But if the sound is an unfamiliar one, a sound that he cannot compare with any memory, and if he is standing in the dark or had his eyes closed, he will be hard-pressed to describe the source. If you perform this experiment, you'll find yourself asking for the sound to be repeated again and again, and each time you'll be certain that with the next hearing you can identify the source. But the brain does need comparative information, and without it, sound has no source as far as the brain in concerned.

DIRECTIVITY INFORMATION IN MUSIC

There is, inherent in virtually every recording made today, certain features that allow "source" localization subjectively on the part of the listener. The only stimuli needed to extract much of this built-in data are the corner positioning of four speakers and some control over what part of a signal is fed to each speaker system.

Take note that no mention was made of amplifiers. We are talking only of extracting phase information from existing signals; specifically, from a pair of stereo signals. Since recordings are made now with a variety of instruments of varying phase relationships, it is a simple matter to effect some degree of control over where the signals are placed.

It should be understood at the outset that the tricks you can do with phasing and level and frequency control do not result in four-channel sound. Oh, you can hear sound seemingly originating from four discrete sources, make no mistake. But this is not four-channel sound in the sense that it can be. What you are doing is borrowing some of the information present on the two front channels and placing them in the rear, with the knowledge that at some point during the program, the relative phases of the front signals will be such that a sound will be louder at either the right rear or the left rear. And of course it will seem to be emanating from one of these two locations.

The listener's position is fairly important for these experiments, for the signals that seem to originate from the rear are present on the front channels, big as life, but at a slightly lower level than they appear in the rear. If you are much closer to the front than you are to the back, you might not hear the rear speakers at all; and if you are too near the rear speakers, it will seem as if you're listening to an entire program with the stage behind you. If you sit too near the right, you won't detect the subtle increases in loudness that come from the left rear. And that's the way it goes all the way down the line.

Nonetheless, you can demonstrate the almost eerie sensation of "surround sound" by connecting up a second pair of speakers to your existing stereo system as shown in Fig. 2-5. But there are a few ground rules you'll have to follow if you want good results. First, use a wirewound resistor to feed the common line driving the two rear speakers; this will offer an inductive load whose impedance varies in a manner not too unlike the voice coils of the speakers themselves. Second, use speakers that are not more efficient than the ones in the front. If they do not have the same frequency response, there will be nothing lost because the fidelity will be retrieved through the front speakers anyway; but if the rear speakers are of a higher efficiency, they may appear louder than the front speakers even during passages where they should not. The level in the rear should be low enough so the signals from the rear are not even noticeable most of the time. When the phase of recorded signals is correct, the signals will be localized properly, even if the level of the two rear speakers is kept deliberately very low.

The fact that the frequency response of the two rear channels need not be as good as the loudspeakers used for the two front channels can be a huge advantage. It means that you can adapt your present system to at least a simulated four-channel sound system with a minimum of expense and effort. Even with a matrix system, which we'll be discussing later, fidelity is no important criterion for the rear channels, particularly if the main interest is to retrieve the ambience of the original recording. (Ambience signals are signals in the air

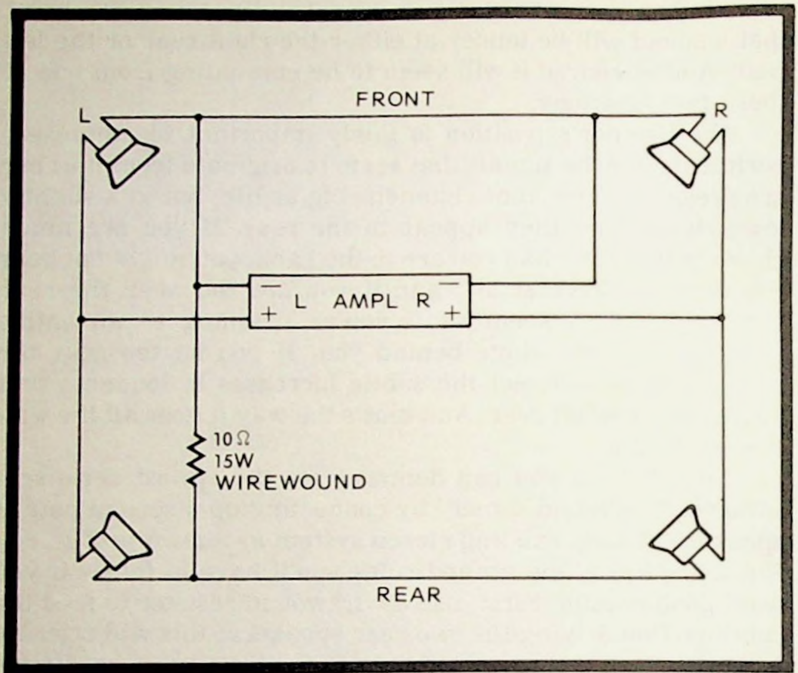


Fig. 2-5. An effective passive four-channel sound system can be simulated if you have a normal stereo system to begin with and a couple of extra speakers. Just connect the speakers as shown, and feed the common line through a series resistor. The resistor should be wirewound, though, because its impedance should be variable. A wirewound resistor offers an inductance that will provide some variation in impedances with frequency.

that arrive there through reflection or deflection from various surfaces in the room where the recording was made, and they are lower in high frequencies than direct signals because of their character.)

An excellent way to get started in four-channel sound is to set up the speakers for it, and connect them up passively as shown in Fig. 2-5. Your stereo system retains its full capability, all four speakers will be operational, you'll get added dimensional pleasure from listening to all your old stereo records, and you will be ready for whatever four-channel system you decide on later. Thus, there will be no performance sacrifice because you are preserving compatibility.

If you have a room with two free corners, setting up another pair of speakers will be a snap. Buy as many low-priced 16-ohm speakers as you can afford. If you shop around, you should be able to come up with a pretty good buy, because today, amplifiers are mostly transistorized types and are designed to work into an 8-ohm load. However, if you parallel a couple of 16-ohm speakers, you have an 8-ohm load anyway. And if you use more than two, you can continue to parallel them and series them in such a way as to maintain the low

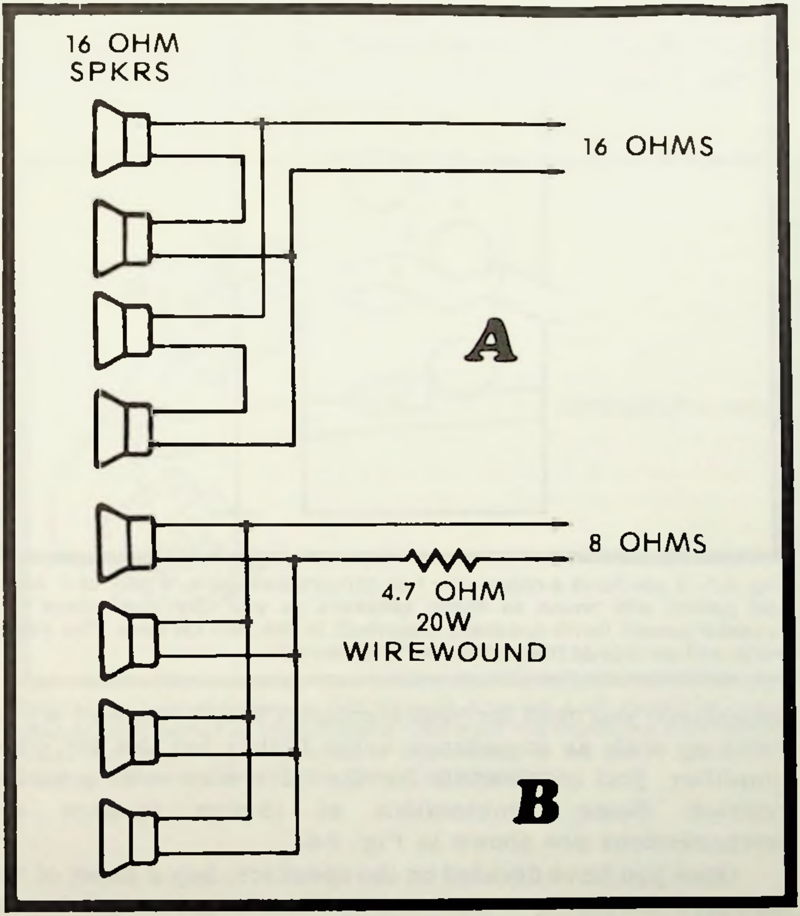


Fig. 2-6. A bank of low-cost 16-ohm speakers will make a good rear channel. You can series-parallel them as shown in A to get 16 ohms, or parallel them as shown in B and feed them through a series resistor to get 8 ohms. If you replace the resistor with a length of wire, the impedance will be 4 ohms, but this will be too low for most passive rear-channel systems because of the dangerous load on the amplifier at certain frequencies and loudness levels.

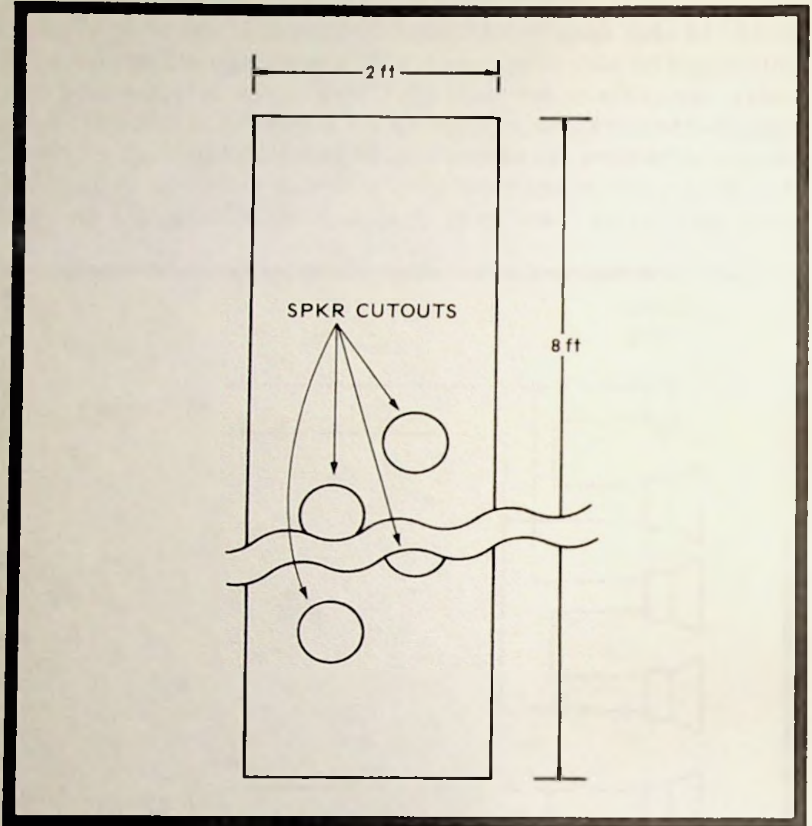


Fig. 2-7. If you have a room with two corners available, a pair of 2- by 8-foot panels will house as many speakers as you like; just place the finished panels (with speakers mounted) in the two corners. The room walls will service as the rest of the "enclosure."

impedance you need for your transistor amplifier. And if you come up with an impedance value that is too low for your amplifier, just compensate for the difference with a series resistor. Some combinations of 16-ohm speaker interconnections are shown in Fig. 2-6.

Once you have decided on the speakers, buy a sheet of ¾-inch plywood that is 4 feet wide by 8 feet long, and split it into two 2- by 8-foot pieces. Saw out the cutouts for the speakers, as shown in Fig. 2-7, then affix the speakers solidly with sheet-metal screws or nuts and bolts. Stand the panels up in the two available corners and cover them with grille cloth. Add a simple framework of regular molding if you like to give it a real finished look, and you're in business. If you want to do a first-class job of it, pack acoustical fiber behind the speakers before attaching the panel to the wall corner. This will dampen the sound somewhat so that the wall doesn't vibrate with every resonant note the speakers reproduce.

Tweeters will be no problem, either. Just use a couple of inexpensive hard-cone types ($2 from Lafayette and other suppliers), and connect them to the incoming speaker leads as shown in Fig. 2-8. You probably won't even need to buy the capacitors, since the hard-cone tweeters usually are packed with the proper driving capacitor in the first place. Use two tweeters for each speaker bank.

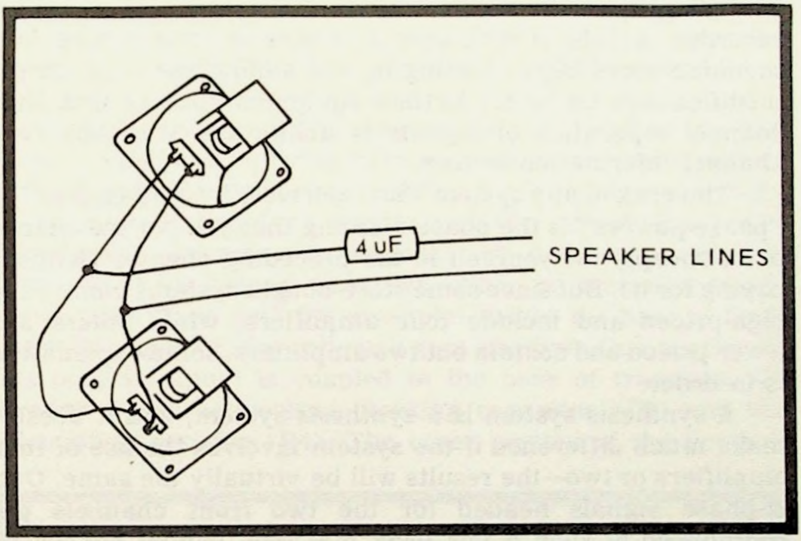


Fig. 2-8. A pair of tweeters, fed through a series 4 uF capacitor, can be connected together as shown and driven from the incoming speaker lines. Since the operating frequency is high, there is no worry about overloading the amplifier with too low a load impedance.

Chapter 3
One Times Two Is Four

An audiophile would have to be deaf and blind these days to be unaware of the existence of a four-channel synthesizing technique that virtually every stereo receiver-amplifier manufacturer is bragging about. Once it was established that phase differences inherent in conventional two-channel stereo recordings could be used to control the directivity of certain recorded sounds, it was only a matter of time before hi-fi manufacturers began touting it, and adding the very simple modifications necessary to their equipment lines so that four-channel separation of signals is achievable with any two-channel information source.

The crux of any system that "derives," "synthesizes," or "phase-powers" is the phase diddling that you got the chance to do cheaply for yourself in the preceding chapter (without paying for it). But since some store-bought systems come very high-priced and include four amplifiers, while others are lower priced and contain but two amplifiers, some explanation is in order.

A synthesis system is a synthesis system, and it doesn't make much difference if the system involves the use of four amplifiers or two—the results will be virtually the same. Out-of-phase signals headed for the two front channels get reproduced at such a low level that they cannot always be heard. But these signals are grist for the mill of the two rear speakers, for they can convert them to sound that can be as loud as you like. If you make speaker connections as shown in the preceding chapter, you can synthesize four channels using but one stereo amplifier; but this will increase the power burden of the amplifier, and if your system isn't up to the task, the level will drop too much for you to enjoy the programs.

LOW-LEVEL PHASE SHIFTING

The alternative is to juggle the phasing of the signals somewhere earlier in the system, say between the preamplifiers and the amplifiers. Here the two basic signals are low in level, so there is no need for big resistors of the type used for speaker phasing. There are several ways of accomplishing the necessary phase shift, but the most practical method, which retains the separation (or at least as much of it as possible under the circumstances) of the two front channels, involves the use of a simple one-transistor amplifier that provides phase inversion as a corollary to its amplification function.

With this method, of course, four amplifiers carry the audio signal to the speakers. But here the speakers are all connected in phase with each other. The signals themselves provide the inversion necessary to create the effect of four channels. Fig. 3-1 shows a basic phase inverter. If you're surprised by its simplicity, remember that it is really nothing more than a common-emitter amplifier; it is simply the nature of a common-emitter amplifier to invert all signals fed to it.

If you're up to building your own simulated four-channel sound synthesizer, and you have a couple of extra amplifiers kicking around, try the circuit of Fig. 3-2, which is a two-stage phase inverter. You don't need to limit the output signal with this circuit, because two signals, 180 degrees apart, are available at the output, as shown. When an incoming signal (from your "tape out" for example) drives the base of transistor Q1 negative, the collector goes positive. One portion of this positive signal is coupled to the base of transistor Q2 through the direct-current blocking capacitor (C2) and the attenuating resistor (R4). The other portion of the positive

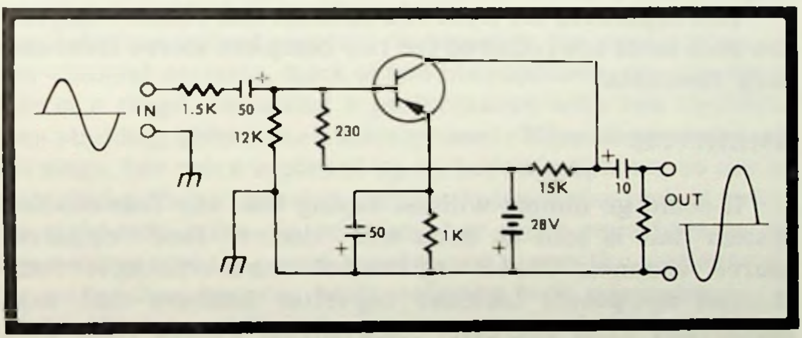


Fig. 3-1. This phase inverter is actually a very high-gain common-emitter amplifier. The resistor shunting the output deliberately limits the output so that the level of the rear channels will be no louder than the level of the front channels.

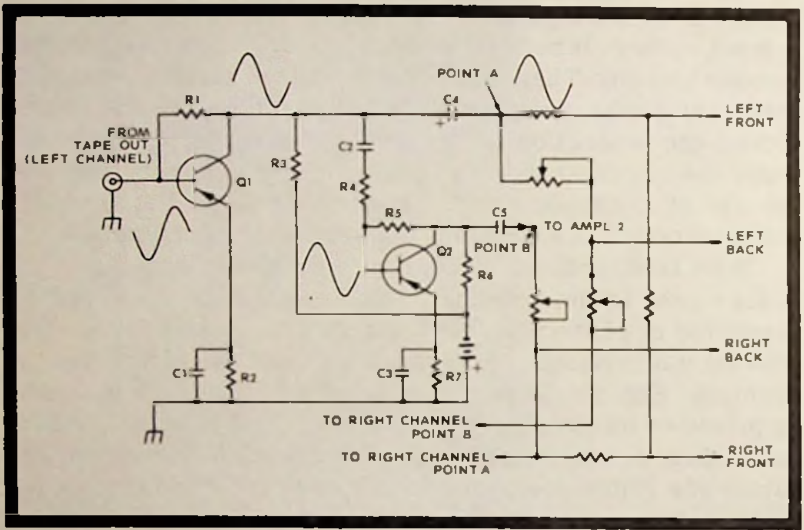


Fig. 3-2. This two-stage phase inverter provides two output signals that are 180 degrees out of phase. The C4 output is the drive for the rear channel of one side, and the C5 output is the in-phase drive signal for the existing front-channel amplfier.

signal is coupled through dc blocking capacitor C4 to the input of your add-on amplifier for the rear speaker for that channel.

The positive-going signal on the base of transistor Q2 causes a negative-going signal on the collector of that stage because Q2, like the first stage, is a phase inverter. This negative signal is coupled through dc blocking capacitor C5 to the "tape in" or "auxiliary" input of your existing stereo amplifier.

This separates the front and back for one channel only, so two such units are required for two complete stereo front and back channels.

AMBIENCE

It could go almost without saying that any four-channel system that is sold as such will "derive" four "apparent source" channels. That is why most of the advertising for four-channel equipment includes big-letter banners that say: RECOVER THE HIDDEN AMBIENCE FROM ALL THE STEREO RECORDS IN YOUR LIBRARY. The fact is, if any quad system has the capability of decoding information put there intentionally by the record manufacturer, it also has the capability of passively responding to all the information that appears there inadvertently.

It may be a little deceiving for the ads to state unequivocally that every stereo record has information that is totally unclaimable with only two channels. It would be deceiving for a manufacturer or salesman to state that the derived signals you hear from the two rear speakers represent the ambient sounds of the auditorium or studio where the recording was made. A more truthful statement would be that certain signals, when they appear at the original pickup transducers out of phase, are often either lost or suppressed by unavoidable cancellation when the record is cut.

Experienced, knowledgeable engineers who make the cuts have a great deal of control over what goes on the record, however, if a sufficient number of tape channels have been used during the recording of the master. They often do cause the phase of one signal to lead or lag another in order to make certain that as much information as possible gets recorded onto the tape.

What the quad ads refer to as "ambience signals" are those signals that are ostensibly reflected from the back wall of a concert hall, then back to the microphone, where they are duly picked up for transfer to a stereo disc, there to remain hidden covertly until that shiny day when some smart buyer outfits himself with a derived four-channel system and releases all that locked-in listening pleasure. To be sure, you can recover sounds that are not apparent with two-channel stereo, but there is a "pipe-dreaminess" about synthesis the way the admen tell it. Expect an improvement over two channels, but caveat emptor! To simplify the explanation of four-channel deriving, think of two microphones, one on either side of a stage, recording a performance with two vocalists (one standing before each microphone). When the vocalist at left sings, her voice is picked up by her microphone, so she is identified with that channel. But her voice is also picked up by the right-side mike. In addition, her voice reverberates to some extent and the sound reaches out across the audience to the walls, then bounces back, entering both microphones.

The first vocalist's voice is phased perfectly with the other vocalists' voice, but her voice may not be phased perfectly with the other mike's pickup of her own voice; and it may or may not be phased with some of the reflected incoming signals arriving at both mikes. Certainly some of the signals will be in phase.

The same basic description applies to the second vocalist, too, whose emitted sounds are sensed by both mikes, though not equally, and not necessarily in the same phase. The out-of-phase signals will constitute the rear-channel information. On playback, all that is necessary is to play the left channel into the right channel in phase opposition. Then all the signals in the left side that are right-channel signals of the same phase are canceled, leaving only in-phase left and out-of-phase right. And all the signals in the right side that are left-channel signals of the same phase are canceled, leaving only in-phase right and out-of-phase left.

As you can see, there is no means for eliminating all left-channel signals from the right channel, nor is there any means for eliminating all right-channel signals from the left. What is possible, however, is to get four channels with different combinations of the two basic signals. And since we can sense each combination as a separate signal, we are influenced by the total playback to the extent that we think we hear four individual channels.

As shown in Fig. 3-3, there are eight basic directions from which sounds can originate (of course there are actually an infinite number of sound source directions, but for the purposes of discussion, we will consider just the eight), consisting of left front, center front, right front, dead right, right back, dead back, left back, and dead left. A pair of signals equal in phase and amplitude appearing in the right and left front channels creates a sound that seems to come from center front. A pair of equal-volume signals of the same phase between any two adjacent speakers will result in a sound that seems to come from a point midway between the two sources.

With this information, it is easy to see how front, rear, left, and right signals can be simulated. Getting the other four signals (the corners) is even simpler. Disregarding phase, if any channel is 3 dB louder (twice the power level) than any other of the speakers, the sound will seem to emanate from that source.

BASIC APPROACHES TO QUAD SOUND

In a derived four-channel system, no one has any control over what bit of information is fed to what speaker, because no thought went into the recording of the two basic channels onto the disc in the first place. So it doesn't really matter whose system is used to derive the four channels. Sansui uses one approach, CBS another. There are also methods used by Dynaco, Electro-Voice, and others. The chief factor is that all these systems do have the capability of breaking down a pair of incoming signals into four signals of the same information but with different quantities of it.

It would also only seem fair to say that most of the manufacturers' approaches we mention are designed to decode records that have been specifically encoded to contain four channels. The fact that these systems can be used to derive four channels of information from existing two-channel stereo records is little more than simple good fortune.

Sansui System

Not all manufacturers of decoders or decoder receivers rely on the basic straight-through phase shift to bring about the reproduction of the four channels. Sansui has been doing considerable advertising about its "special" system (Fig. 3-4) that treats the signals fed to the rear channels to enhance their apparent volume level and increase the ratio of front-to-rear signal level. Quite frankly, the approach seems viable. The concept is simple enough to understand, and easy to explain. I haven't had the pleasure of listening to Sansui's system, but I have looked over reports from professional observers who have. From indications, the Sansui system is about as close to four-channel discrete reproduction as you can get without having four discrete channels. (Sansui, of course, says its system is even better.)

The Sansui synthesizer (Fig. 3-5) has a matrix system that operates in much the same manner as other matrixes. It separates the 180-degree-out-of-phase signals from the two primary channels; but instead of processing them through amplifiers to drive the rear speakers directly, the Sansui unit puts the signals through 90-degree phase shifters.

The left rear channel is moved forward in phase by 90 degrees, and the right rear channel is delayed by 90 degrees. As shown, this combination results in the two rear channels appearing in phase with each other (usually) as well as delayed but in phase with the front. The two rear-channel signals are then fed to a phase modulator, which is used to

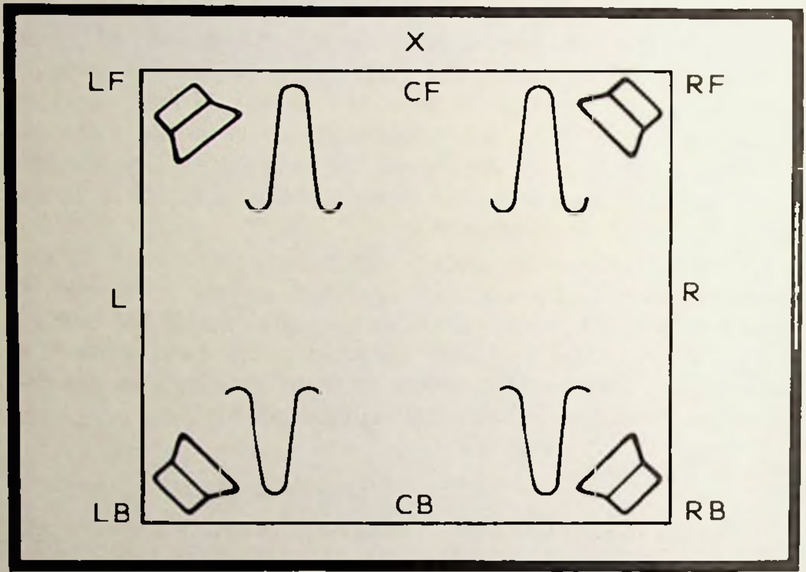


Fig. 3-3. The apparent source of a sound (X) is at a point midway between two speakers if those two speakers are emitting sound of the same level and phase. In this sketch, the sine waves of the two front speakers are shown phased together and higher in amplitude than the rear. When an out-of-phase signal appears on the front channels, its amplitude might be doubled at the rear (depending on the configuration of the derivation circuitry).

provide a continuously varying phase relationship between the front and the rear channels.

As the rear-channel sound components are phase-modulated from 0 to 180 degrees, sounds emanate from the rear with varying phase differences. As Fig. 3-5 shows, the apparent sound volume at the rear left or rear right dramatically increases in accordance with the basic phasing of the signals on the record but at a considerably greater magnitude than would be obtainable without the added "discrete" doctoring. The rear channels will now arrive at the listener's ears with indiscernible (except to the brain, as we discussed earlier) time differences, and the end effect roughly approximates the acoustic effect present in the hall or auditorium where the recording was made.

It seems hardly necessary to mention that these derived four-channel techniques appear a good deal more convincing with live-performance recordings. The effect of any derived four-channel system can be impressive with "live" program material, but the Sansui QR-6500 (Fig. 3-7) should offer even further enhancement because of the differences in levels it manages to bring about between any one channel and all the other channels.

Sansui achieves this level control by incorporating circuitry that senses the gain of the signals fed to all four chan-

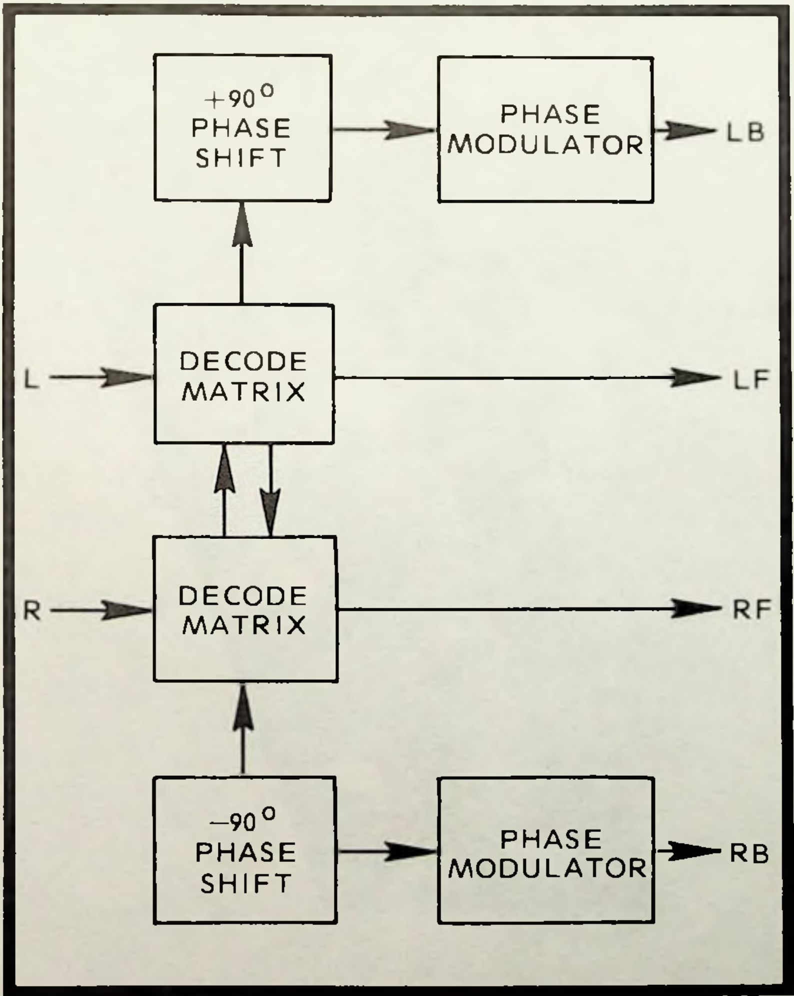


Fig. 3-4. Sansui's QR-6500 takes the rear-channel sound components, shifts the signals (up for left and back for right) to get them in phase, then uses them to drive a pair of phase modulators. The rear-channel outputs then appear as random-phased, well-separated signals that are as much as 20 dB away from their front-channel counterparts.

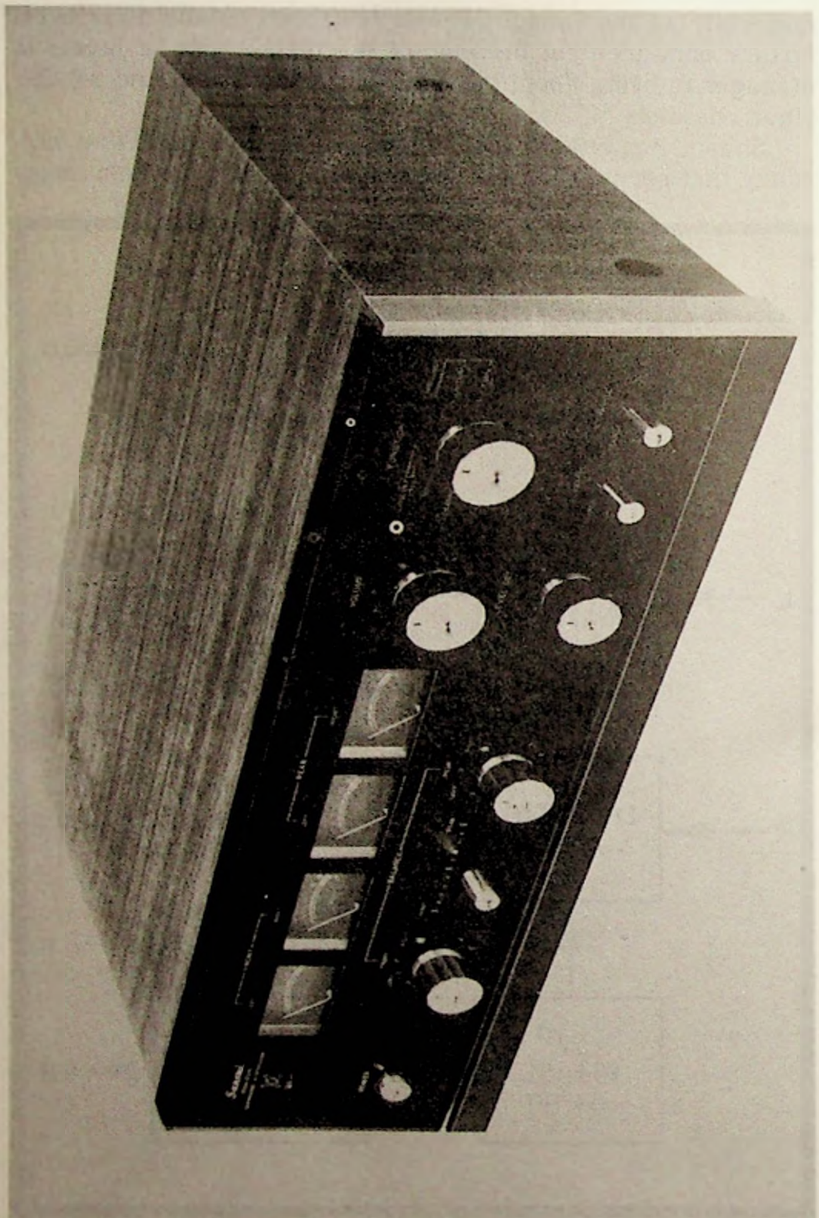


Fig. 2-5. The QS-1 Quadraphonic Synthesizer unit shown here is Sansui's deluxe synthesis system; it does not contain the company's sophisticated four-channel matrix decoder. (Courtesy Sansui Electronics Corp.)

nels. When any one of the four corner signals is predominant, the channel diagonally away from that one is cut off completely. But even though it is shut down during the high-volume-level period of its diagonal opposite, its inherent level is continuously sensed. When the level goes up or when the diagonally opposite channel loses volume, the clamp is removed, and the gain goes back to its "normal" position.

The Electro-Voice System

Electro-Voice began a couple of years ago with a system it purchased from Feldman and Fixler, and the system incorporated a great deal of front-to-back separation with an up-front separation that was a degraded version of a conventional stereo arrangement. The combining of the two channels to get

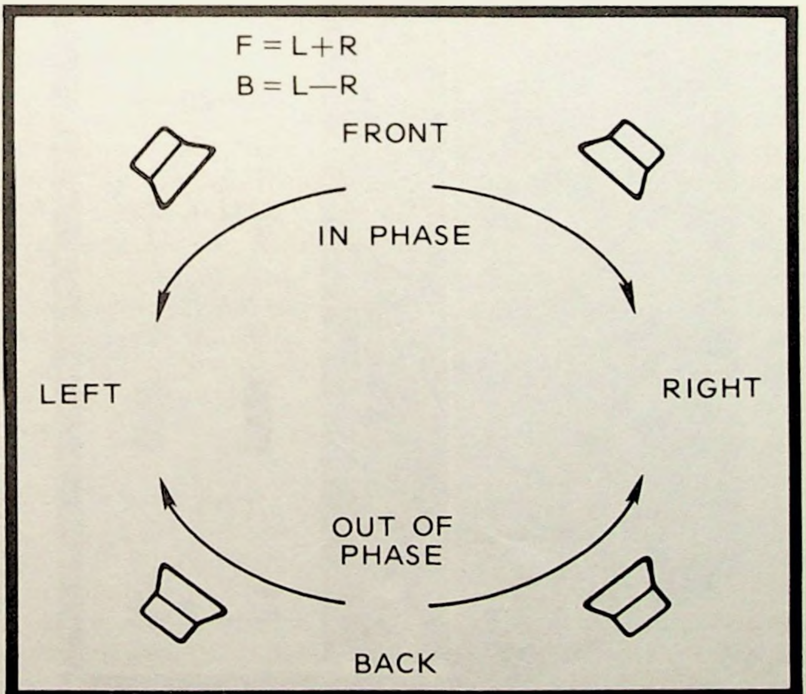


Fig. 3-6. The apparent back-channel volume increases or decreases according to the phase of the information recorded there. With a fully matrixed four-channel record, the out-of-phase components come from the back, but as the phase changes (approaches that of the front), the apparent source follows the two lower arrows, moving forward.

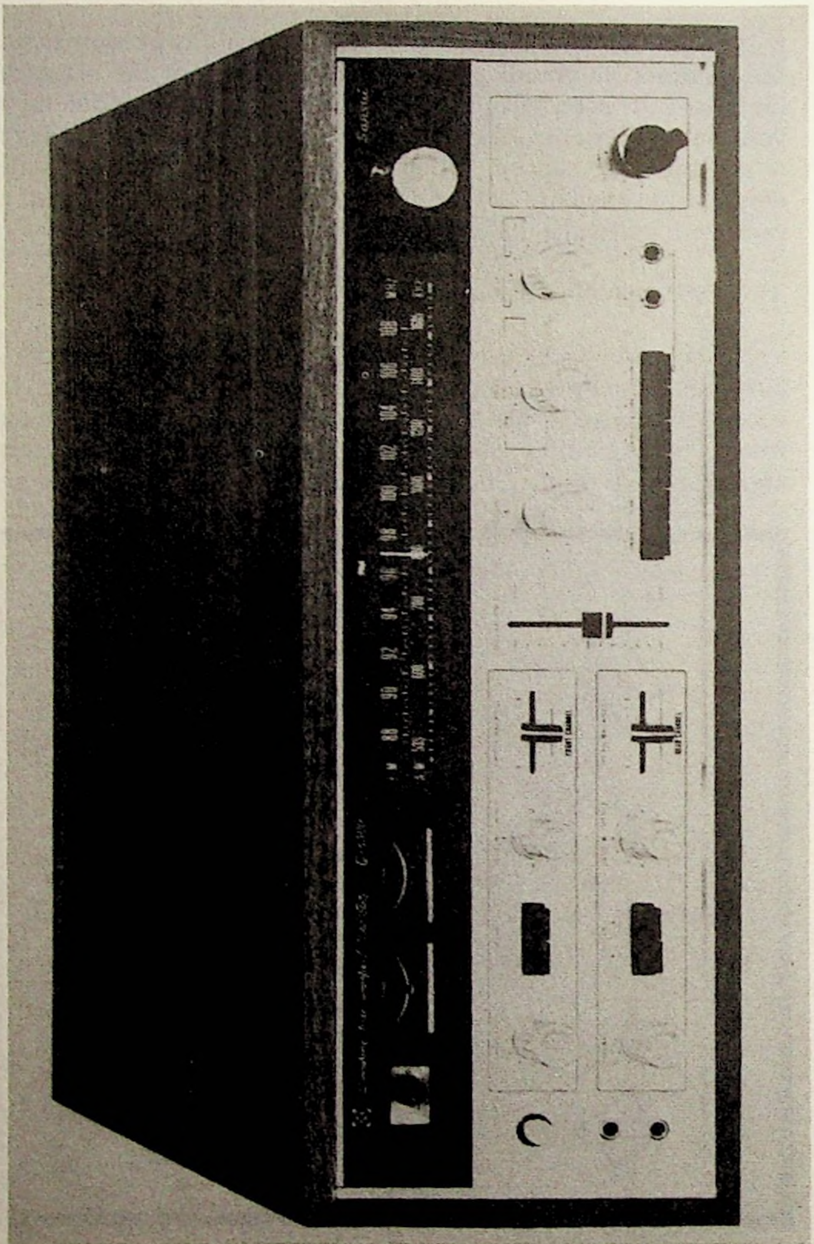


Fig. 3-7. The QR-6500 incorporates a logic circuit to shift the matrix so that separation is optimum under all conditions. Matrix shifting is an exclusive function of Sansui, made possible by the symmetrical layout of the matrix. Also see Fig. 3-4. (Courtesy Sansui Electronics Corp.)

phase cancellation, of course, resulted in a deterioration of the two front channels' basic individual integrity; but this was considered of minimal importance, because even 3 dB of difference between two channels results in good separation to the ear, and retention of the phase similarities between the two front units resulted in very superior front-center information (an area where some of the manufacturers seem to fall short).

With a growing number of systems becoming available, however, and each system differing substantially from the others in the two basic areas (phase of each channel and level of each channel's opposite-channel content), Electro-Voice employed the services of a computer to design a system that would be compatible with all the other units being marketed. (The idea was to make a universal decoder for decoding intentionally matrixed records of various types, which we'll discuss in the next chapter.)

The ultimate system incorporated by Electro-Voice did give the desired overall compatibility sought by the company, which included enhanced left-right up-front separation, but only at a compromise of the front-to-back capability. Leonard Feldman, one of the early proponents of four-channel sound, described the "apparent" spatial difference between the basic four-channel approaches in a 1972 article that appeared in Stereo Magazine (Billboard Publications). His observation was that the apparent room size was wider, but not as lengthy as with the original E-V system. Still, the approach does have a number of things going for it, not the least of which is the latent compatibility with all the other diverse approaches currently being marketed by various foreign and domestic firms.

Dynaco's Systems

It is interesting to note that Dynaco's early system was actually made specifically for the enhancement of two-channel sounds, and not for the decoding of four channels from two. But with the capability of being able to create the illusion of four separate channels from two, the company thought it would be a good idea to control the amplitude and phase of various program portions. On playback, then, the proper proportions of the controlled signals would be properly distributed to the four speakers in the room. It was the same

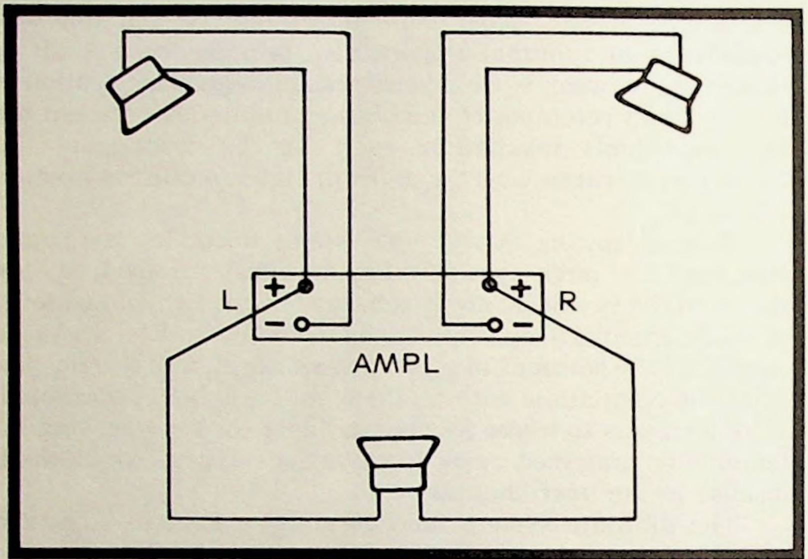


Fig. 3-8. Early Dynaco method of "surround sound" derivation was simply this three-speaker system, which depended on polarity (phase) variations of the two front channels to reproduce an "indirect" sound from the rear. With near-the-floor front speakers and an elevated rear center speaker, the effect of presence on two-channel material is astounding.

logical conclusion made by other firms at about the same time that made four-channel sound an important medium.

Dynaco's first system was purely passive. That is, all the phase plotting was done with no additional amplification, and at the speaker terminals. And the approach was very similar to that described in the preceding chapter. Dynaco adopted a three-speaker system that was extremely effective and beautifully simple. It consisted of connecting a rear speaker to the two front channels as shown in Fig. 3-8.

When the signal on the left side is polarized with the one on the right, both "plus" terminals see a voltage of the same polarity, either plus or minus. But one of the two will be greater than the other because the two signals are actually different. If the left channel sees five volts and the right channel sees four volts, the rear channel will develop but one volt, the difference between the two. But if the polarity of either of the front two speakers is reversed, as often happens with recorded material, the left and right still "see' the voltages they are fed with, but the rear channel will now

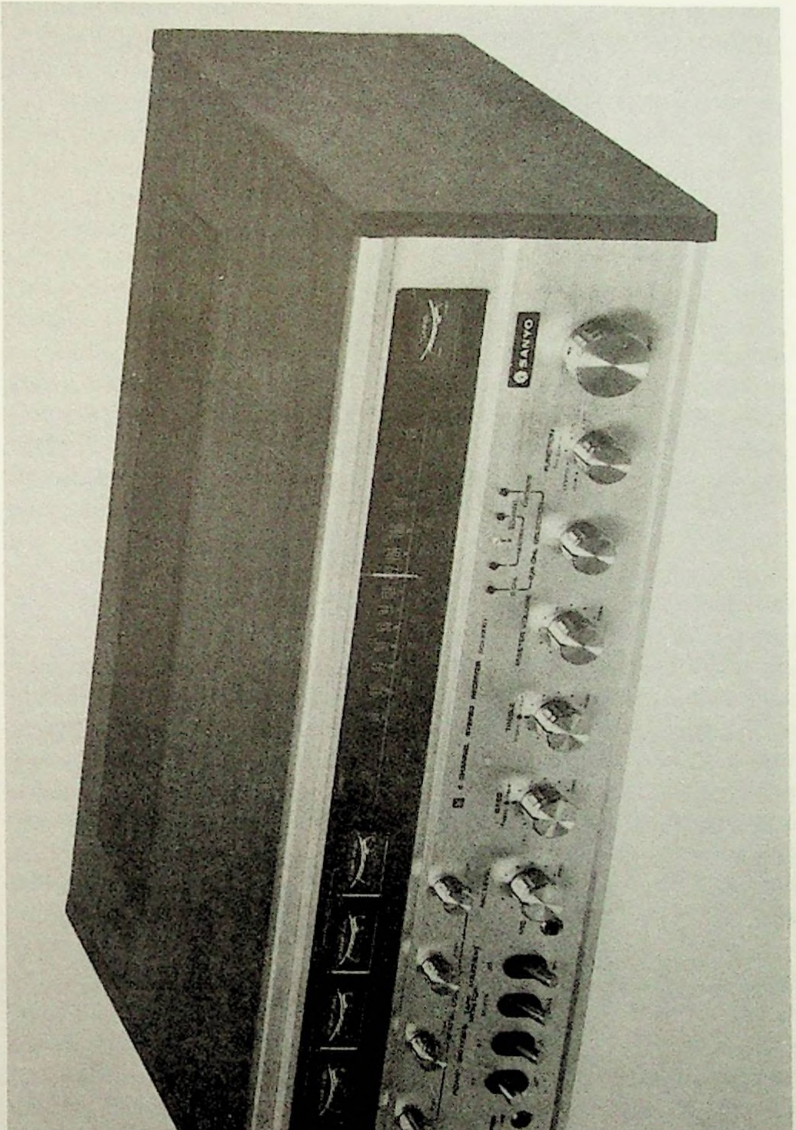


Fig. 3-9. This Sanyo DCX-3300K receiver contains no less than three decoding matrices; one for the Sansui system, one for the Electro-Voice system, and the third for CBS-SQ. (Courtesy Sanyo.)

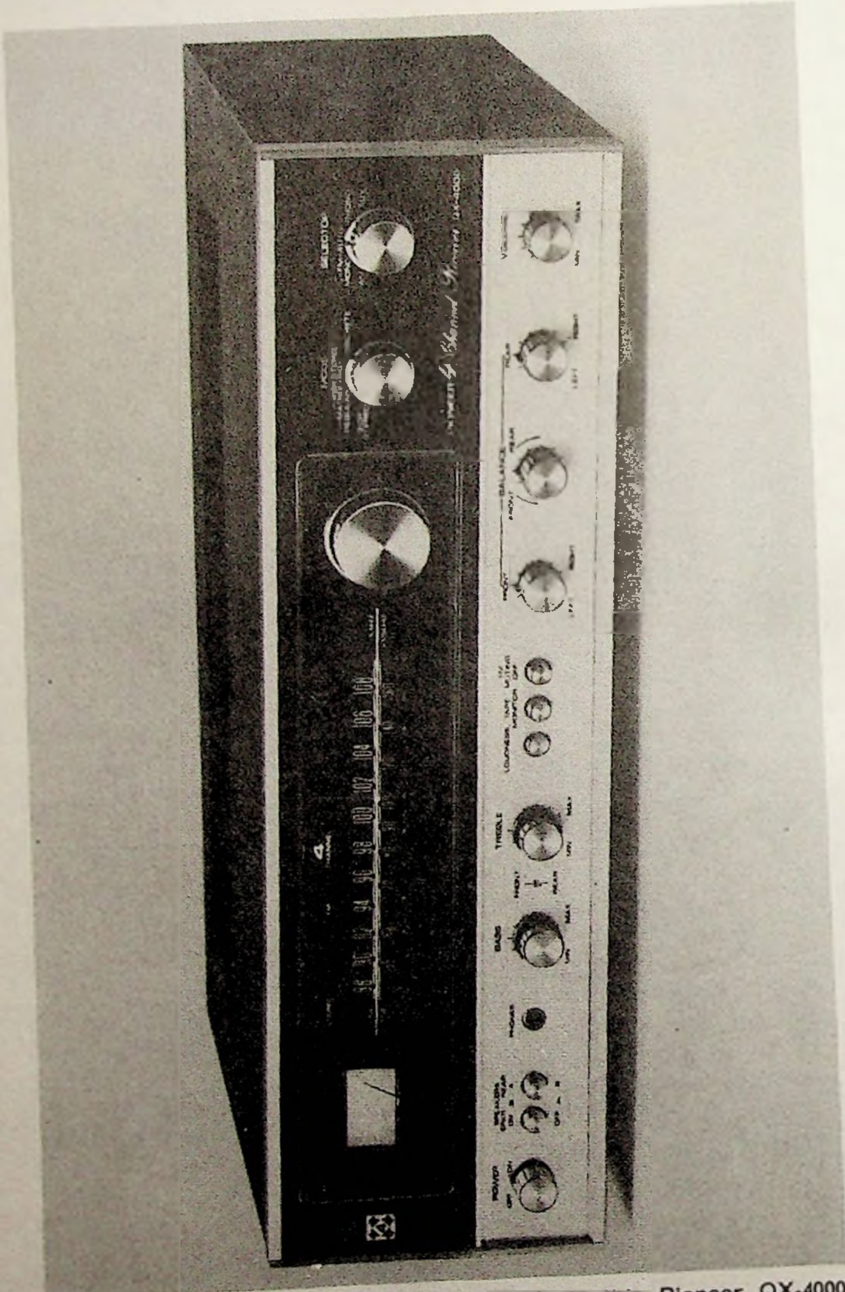


Fig. 3-10. Some consumer equipment, such as this Pioneer QX-4000 receiver, contains "dual" matrices. In this case, the matrix is switchable from Sansui to CBS. (Courtesy U.S. Pioneer Electronics Corp.)

develop the combination of the two, because the negative side of one signal effectively appears in series with the positive side of the other channel.

When the rear speaker is elevated above the head of the listener and to the rear, and the two front channels are low, near to the floor, the overall effect can be quite striking.

Later versions of Dynaco's phase differentiation technique were quite similar to those described in Chapter 2, where two speakers were used in the rear, but fed through a resistive network to alter the level of one rear channel or the other. When Dynaco made its signal-breakdown scheme available to record manufacturers, some began to produce discs with varying levels and phases for deliberate decoding. Not surprisingly. Dynaco's method proved highly compatible with Electro-Voice's, and owners of one playback system found they could reproduce music encoded with either system with equally good results.

CBS System

The manufacturers mentioned thus far can be recognized as being in the hardware business. Columbia Broadcasting Systems, however, is not. Nonetheless, CBS was among the first to see the potential merit in a system of reproducing four sound channels from two, and the company became deeply involved in researching various methods to achieve this goal without compromising the quality of discs in current production. Little can be said of CBS' approach with regard to derivation of random ambience of two-channel records because CBS' main concern is in the intentional recording of these signals for playback on equipment designed to undo what was done in the recording labs.

Many (probably most) of the equipment manufacturers who are not involved in recording themselves have adopted the mathematics of the CBS approach, and their equipment is made for decoding four-channel "SQ" (Stereo Quad) matrixed discs. Nonetheless, without exception, all the equipment produced for this purpose can be used to derive or synthesize four channels from a conventional stereo disc. Those manufacturers currently using the CBS matrixing format include Lafayette, Allied Radio Shack, Sony, Pioneer, Concord, Kenwood, Sanyo, and others. Some of these, such as Sanyo (Fig. 3-9) and Pioneer (Fig. 3-10), include several matrix decoders in one package.

Chapter 4
Matrix Versus Discrete

Some years back, one major manufacturer of records noted that certain phase-manipulating techniques could result in the playback of two "apparent" channels with a regular monaural recording, provided the monaural recording was properly cut. The process was heralded, during the early days of stereo, as "Duophonic Sound!" or some other such title.

This story is pertinent to what is happening in four-channel sound today, because it involved complaints, cross complaints, hard feelings among manufacturers, and disappointment on the part of some consumers. In essence, the record manufacturer claimed to have a process whereby two channels of information could be extracted from a monaural cutting. And, as we have already learned, this is an easy possiblity because of the way the ear uses phase information to localize sound sources. But those manufacturers who were then in the all-new business of cutting stereo records weren't having any of it. Their cry was understandable; one manufacturer was pushing phased mono discs and raking in cash from consumers who expected stereophonic programs; other manufacturers had gone the full stereo route and thought it unfair that the "bad guy" record maker could get away with such an underhanded exploit. As far as they were concerned, every consumer who would inadvertently buy one of the "Duo" discs would be disappointed, and would then turn off to stereo altogether, thus killing their own chances to woo the customer.

Ultimately, the Federal Trade Commission stepped into the act, and they brought down the gavel against the "Duo" producer. Their decision was that the signal-splitting mono cutter was giving the consumers a mono signal that had been tampered with rather than a pair of actual, dyed-in-the-crepe discretes.

If you aren't yet beginning to see the significance of all this, consider the fact that there is now a very vociferous—though relatively quiet—battle going on within the world of four channels. There are proponents of one system or another making unbelievable claims with respect to signal separation and individuality. There are proponents of another system debauching all those claims and shouting "Fake! Fake!" Perhaps the Federal Trade Commission will ultimately settle this question, too; but you, as a consumer (or at least a potential one), should be advised of the differences between the claimant's packages.

On the one hand, there are manufacturers who produce equipment—tape playback and recording equipment, phonograph records, and special amplifiers (with the required ancillary equipment)—designed specifically for the purpose of reproducing four individual (discrete) channels of information. These four channels may be related to each other, or they may be four separate programs—with four discrete channels—it makes little difference.

At the same time, there are other manufacturers who are producing equipment designed to create the **effect** of four discrete channels, and there are those who manufacture equipment designed specifically to split two stereo channels into four **apparent** channels. Unfortunately, all systems are being sold as "four-channel" or "quad" systems, although reputable manufacturers of "synthesis" equipment do bill their line according to what it does.

But there are still the two groups of claimants to the "four-channel" title. One group, consisting of those who make equipment that breaks down phase- and level-coded stereo signals into four channels, includes the "matrix" proponents. The other group, made up of the manufacturers who have found ways to give buyers four real, honest-to-goodness, isolated channels of separate information, is referred to as the "discrete" set.

Matrix

To clear the air, a matrixed four-channel record is to quad what the "Duo" record was to stereo; it is not the real thing. It is good. It is interesting. It is exciting. It can surround you with music and can allow you to hear sounds you've never heard before. It may open up a dimension to music listening that you never dreamed existed. But it really isn't four channel, at least not in the sense that the channels are actually different from one another. With a matrix system, you could never reproduce four individual programs (one for each channel) at one time. Perhaps you will never even want to; but the fact is, you couldn't even if you **did** want to, because that's the nature of the approach. Because four matrixed channels are still two sound channels that have been modified in phase level (or phase and level) to the extent required to make some of the sound seem to come from the right front, some of the sound come from the right rear, some come from the left front, and some from the left rear.

Matrix systems permit the use of conventional stereo media, such as discs and tapes, and so give the greatest potential for consumer use in the future. It should also be noted that they can be adapted to the present FM multiplexing broadcast without requiring a major addition of equipment at the broadcasting facilities.

The encoder is an instrument to compress four channels of signal into two, and the decoder attempts to expand the two channels of signal back to four. The difference among various matrix systems is in the method of mixing the four channels of signal when compressing them.

Discrete

The discrete approach is just what it appears to be. It means you get four individual channels of information, and each channel is as discrete as an existing stereo is in a conventional two-channel system, and sometimes even more so. There are advantages with four-channel material of the discrete class that cannot ever be duplicated with any system that involves simulation, and one of these advantages is separation. In terms of actual use, separation means depth—the perception of that elusive dimension that makes an 8 x 10 ft room sound like an auditorium. And however exciting or good or fascinating or enjoyable or awesome and wondrous a matrix system is, the four-channel discrete system will top it. If you understand this at the outset, you will never expect more from a matrixed system than it is possible for you to get.

The discrete system uses a high-frequency signal called a subcarrier to cut the four separate channels of signal in the left and right walls of a record groove. It utilizes frequency modulation to modulate the four signals with the subcarrier, and is similar to the FM multiplexing system in that sense. But the music recorded by this system cannot be broadcast by the present FM transmitting equipment without major changes.

Several proposals have been made for discrete four-channel broadcasting on FM, but all proposed systems require modifications of the current broadcasting and receiving equipment. If a technique can be found to make the discrete four-channel disc system and the discrete four-channel FM broadcasting system equivalent, playback equipment for the consumer need not have separate circuits for the two program sources.

The common deficiency of most matrix systems has traditionally been the lack of sufficient separation among the four reproduced channels, meaning that a part of the sound in one channel leaks to adjacent channels. As far as regular matrix systems are concerned, however, this had been largely overcome by the development of new, improved decoders by Sansui and a few other manufacturers which give outstanding channel separation. This has added greatly to the advantages of the regular system, and is expected to accelerate the rate of popularization among consumers.

THE BATTLEFIELD

It is difficult to discuss the relative merits of the various approaches to four-channel sound without getting caught up in the subjective pitfalls of the controversy raging incessantly among the many axe-grinding "best-system" advocates. One top-ranking audio engineer who pooh-poohs a matrix approach will say it is impossible to get ten pounds of garbage in a five-pound bag. Another, in defense of matrixing, will tell you that a quart of orange juice comes in a six-ounce can; it isn't the basic signal that gives you the separation—it's how it is processed by the end equipment.

The Big Battle began some time back, but I think it was brought to a head in April 1972, when Lou Dorren, staunch advocate of discrete four-channel sound and inventor of the discrete FM broadcasting system, prepared a paper to be delivered at an Audio Engineering Society convention. He pulled no punches in his writeup, and said the blatant limitations of matrixing seem to proclaim "that matrixed sound is neither discrete nor quadraphonic, but merely a clever advertising stunt perpetrated by mercenary manufacturers to bilk an enthusiastic, four-channel-bound public of millions of dollars."

In his paper, Dorren quoted from advertisements placed in various media by manufacturers of matrix four-channel equipment, and pointed out that they were overoptimistic (his words actually were a bit stronger) with regard to the degree of separation achievable.

The big blow, however, was leveled directly at CBS, when Dorren managed to dig out an old technical paper prepared several years earlier by Ben Bauer, who later became vice president of CBS. The technical paper was discussing sound quality, and the matter of phase relationship in particular, and it appeared to place CBS in the awkward position of disclaiming the very technique the company was now backing. Bauer's old report, as quoted by Dorren, said, of out-of-phase recorded signals:

"This is a most unnatural situation which has no counterpart in the normal hearing experience. First, there is a reduction of response at the low end, and with some observers, a feeling of 'pressure in the ears.' One can only conclude that integrity of phase relations must be carefully maintained in stereophonic sound reproduction." (Ed. note. The quote was from IRE Transactions of Audio, Jan.-Feb., 1962; Vol. AU-10 No. 1; pps 18-21.)

CBS, of course, took the first opportunity it could to retort to the charge of "turncoating." The scene was the 1972 convention of the National Association of Broadcasters, and the subject of Bauer's comments was raised again.

The CBS man at the convention defended Bauer's remarks by stating that they had been quoted out of context. He acknowledged that the "information theory" equation that describes a matrix system does "have a phase component to it," but that Bauer had not been discussing anything related to the manner in which four-channel matrixing is achieved today. "I commend that paper in the IEEE Journal to the reading of anybody here," he said, "rather than taking the statement out of context. I would like you to read everything it has to say—but recognize one thing: a matrix recording is NOT an out-of-phase recording."

His dander was up, and perhaps had he a chance to think it out he would not have offered this tidbit about four individual channels: "The discrete disc is a lab curiosity."

RCA, in league with Japan Victor Company, an affiliate, had been working on the development of a four-channel discrete reproduction system for years. And now, after investing perhaps millions in research, trial and error, and avenues that led nowhere, the company has a system that is not only workable but representative of major technological thrusts into the future of recording in general.

These two companies worked together to get a disc that would hold a 45 kHz signal without degradation after many playings. They had to design and develop a special stylus and cartridge that would be capable of sensing that signal (three times the maximum frequency most people can hear). They had to have a disc that would resist the dust-sucking static charges that play havoc with high-frequency signals. In short, RCA and JVC, acting in concert, did more to advance the state of the art of reproduction than any other single venture in the history of high fidelity.

It would seem natural, then, for the conference attendees who advocated the discrete approach to take offense at the remark of the CBS man, who also leveled other indictments as well.

"We have been hearing about a discrete disc coming from Japan now for two years," he said. "We are still hearing it is 'going to come.' It has yet to arrive..." and "...a discrete record...has to be recorded about 5 dB lower in level than a standard LP, and if it is a 'brighter' recording, the degradation might be as much as 11 or 12 dB. And what about playing time? So far we are only getting fifteen minutes on discrete discs. Shall we charge the customer for two records for the same program, or what movement of the symphony will we leave out?"

The speaker for CBS was Emil Torick, and he knew the ins and outs of discrete recording from a unique vantage point—CBS was probably the first recording firm in the world to produce a discrete disc. And the problems Mr. Torick spoke of were undoubtedly the snags his company had run up against in its own abortive development effort. But he was painfully unaware of the recent strides made by RCA and JVC, whose representatives or spokesmen were only too pleased to point them out.

The first to counter the CBS man's claims was Lou Dorren, who himself had developed the four-channel discrete method of FM broadcasting. "I am surprised to hear the comments Mr. Torick has made about the discrete disc," he said, "because all of the problems he has brought up have been more than solved."

Dorren pointed out to those present that the amplitude level of the RCA discrete record "is 4 dB below NAB standard zero level," rather than 5-12 dB, as was indicated by Torick. "The levels are not dependent upon whether the program material is dull or bright, and this (4 dB down) is the level it is being cut at presently."

The signal-to-noise ratio of the discrete disc would seem to be below NAB standards with a lower audio level recorded on this disc. But the people producing four-channel (discrete) programs are aware of the potential problem in this area and have been exercising extraordinary care in the production process. The result is a signal-to-noise ratio that is no different from that of standard stereo.

The playing time of an RCA discrete disc is 30 minutes per side. And Dorren just happened to have a selection of demonstration records on hand to prove his point.

And then it was time for counter charges, which gave the serious NAB 1972 conference the atmosphere of a CBS-RCA battlefield. Dorren said that the discrete disc was compatible with mono equipment, but that the matrix disc was not. "...if you go down and buy 'Switched On Block,' and play cut four on side one by Lynn Anderson, and play six of the ten cuts by Barbra Streisand, and Paul Revere & The Raiders, and play Indian Reservation or the other major hit on that disc, you will find in most of the cuts where there are background voices, when you go to mono the background voices completely disappear..."

And so it went, with everything happening short of name-calling. Nothing was settled in the sense that "discrete" people were won over to the matrix method; nor were "matrix" advocates converted to "discretism." If anything was resolved, it was simply a general recognition on the part of broadcasters that two competing approaches are being offered to the public; and it appears that both approaches will continue to be around for a while. Both approaches have advantages that can't be matched by the other, and both have built-in disadvantages. In the meantime, quad is "what's happening" in hi-fi today, and everybody wants a piece of the action. Trouble is, there aren't many manufacturers who are sure of just which way to go. The public is a fickle entity, and second-guessing it always proves a fool's game. Still, there are those who have everything to gain and nothing to lose.

THE QUAD BANDWAGON

Hi-fi is one confused industry these days, what with audiophiles chomping at their bits, just waiting for some surefire sign of which way the wheel is going to turn. Many are afraid to tool up with a roomful of matrix equipment for fear that discrete will make it obsolete. Others would like to try for discrete but they see matrix gaining in popularity and they're afraid perhaps the discrete people will leave them hanging with lots of equipment and no music to use with it.

Their fears aren't really all that unfounded, either. CBS claims to have "in excess of 50 records out encoded with the SQ format." But few are the stores where they abound. The selection is certainly limited, of that there can be no rebuttal. Too, the SQ-encoded discs sell for a $1 premium above the normal two-channel fare.

A representative of the Sansui Company told a group of broadcasters that records encoded using the Sansui matrixing scheme would be no more costly than conventional stereo records; presumably, records made the E-V and Dynaco way would follow the same lead. But a check with local record shops will quickly prove that quad-matrixed records cost more, and it makes little difference whether they've been encoded with CBS, SQ, Sansui's QS, Dynaco's, or Electro-Voice's format.

RCA promises to combine its entire record library into one —and that one will be four-channel, complete with stereo and mono compatibility. As a matter of fact, the promise was made nearly a year ago, and the projected date for the unification of the library has long since gone past. RCA also said its discs would carry the same price tag as the previous two-channel stereo records. According to company spokesmen, 115 records have already been released, and "there are more being released every day." Be that as it may, you'll probably be hard-pressed to find even one lone RCA four-channel discrete disc, unless it is offered as a premium "demo" when you purchase your JVC or Panasonic discrete setup. It may well be that RCA has pressed thousands of discrete discs, but it does the buyer very little good unless it is made available to him in the local record shops.

The situation may change in the future, but for the time being even the record shops that sell discrete and matrixed four-channel gear can't seem to get records to play on the systems they're selling. Salesmen are forced to push the least of the equipments' features, that is, the ability to recover ambience from two-channel sources.

When stereo had its heyday, things were the other way around; stereo discs appeared before they were even in hot demand. With quad, and the concomitant confusion resulting from the various approaches, nobody—including the company that produces the records—seems to know which way to invest.

But there are those who can use the term "four channel" to advantage without getting taken up by the skirmishes between the matrix people and the greater wars between Matrix and Discrete.

Headphones

One of the least likely candidates for the four-channel bandwagon, one would think, would be the folks who make the headphones. A person has two ears, and a set of headphones accommodates these two beautifully. A casual observer might well wonder how an earphone manufacturer could possibly take advantage of a trend that is based on the deployment of four speaker systems.

The Koss people, always alert to changing trends, were actually one of the first on the scene in quad sound, and they have several models of headphones designed specifically for use with quadraphonic systems. The unit pictured in Fig. 4-1 is Koss' "Quadrafone," a headphone set that incorporates four separate driver elements (two in each earcup), and contains two completely interwoven cord and plug assemblies.

The unit shown isn't the company's only four-channel model, either; Koss has quad phones ranging in price from $39.95 to $85. According to the manufacturer, the sales have been gratifying. Even when used with conventional two-channel sources, the phones have a little something extra to offer: On two-channel material, the drivers can be paralleled, so what you'll have is nearly double the bass radiating area and a considerable increase in efficiency over the full audio range.

Of course, it was easier for Koss to commit itself in some respects than it would be for manufacturers of equipment that fit in a little earlier in the audio chain. It matters little to speaker or earphone people whether the industry eventually goes matrix or discrete—the audio still must be heard, and that means speakers and earphones will still be salable without the risk of obsolescence.

Fig. 4-1. It may look a lot like stereo headphones, but in reality this is Koss' four-channel "Quadrafone," which incorporates two acoustic drivers in each cup for reproducing four honest-to-goodness channels in any format, discrete or matrixed.

Fig. 4-2. United Audio's Dual 1229 has the extra-quality features required to play back a disc recorded with RCA's discrete method. Low tracking and arm-to-record angle errors are important for this, as are precision antiskate and stylus pressure adjustment.

Turntables

There are other firms who have been benefiting from the drive to quad, too. Turntable manufacturers have reported an upsurge in sales, ostensibly attributable to the quad trend. It makes little difference which of the four basic matrix methods eventually wins out over the others (if indeed one will ever "win out"), as far as the platter spinners go. The matrix approaches all use a two-channel stereo record as the vehicle for the four encoded channels, and all of the turntable manufacturers can boast of full compatibility. Indeed, even if a discrete approach eventually dominates, the higher-quality turntables will still be in there holding the playback cartridges that can respond to the ultrasonic FM carrier encoded on the discrete disc. Typical units that can handle either kind of job (matrix or discrete) with no difficulty whatever are the Dual 1229, Miracord Mark II, and Garrard Zero 100.

The Dual 1229, shown in Fig. 4-2, has an extra long tonearm that has virtually no tracking error and a precision accurately incremented weight-setting feature for the arm. The arm's attack angle remains absolutely parallel with the record for single-disc play, and moves up to parallel the third record in a stack when multiple discs are loaded. It is easily capable of tracking at stylus pressures even below one-half gram whether the turntable is horizontal or vertical. A feature that should prove particularly beneficial for discrete is the antiskate control, with two calibrated rate adjustments. One of the two is ideally suited to the Shibata stylus, which traces the groove wall with a narrower edge than a spherical stylus and rides considerably deeper into the record grove than either the spherical or the elliptical type.

The Garrard Zero 100 has absolutely no tracking error, which is the reason for the "zero" in the name. As shown in Fig. 4-3, an articulating tonearm maintains a constant angle through the playing of a disc, which assures recovery of even the very highest audio frequencies in that near-the-spindle area where most turntables can't quite hack it. Like the Dual, it has precise antiskate and stylus force adjustment, two necessities for discrete quad reproduction.

Benjamin Miracord's Mark II (Fig. 4-4) features adjustable antiskate, calibrated stylus-pressure adjustment, cueing, and a dynamically balanced tonearm that allows precision tracking regardless of levelness of surface. It has a relatively long tonearm that contributes little toward high-

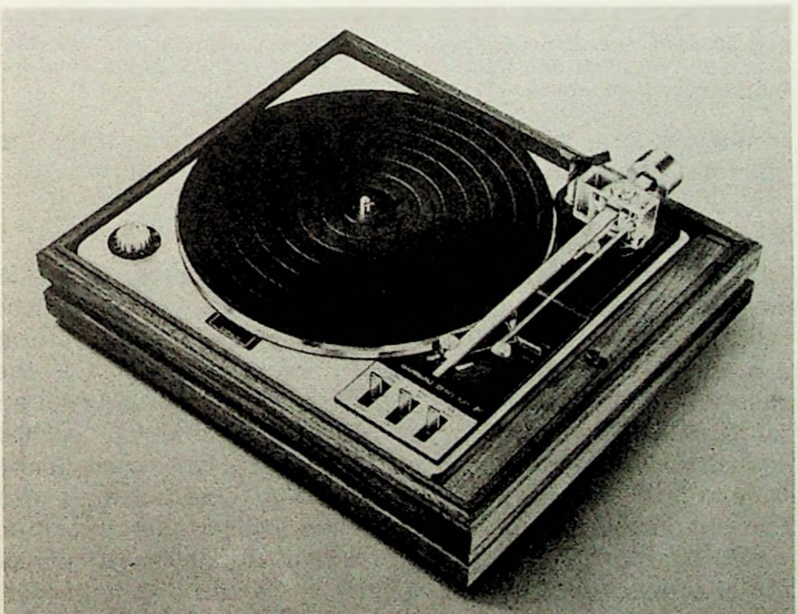


Fig. 4-3. Garrard's "Zero 100" features an articulated tonearm that offers no measurable tracking error, plus antiskate and stylus pressure adjustments.

Fig. 4-4. The Elac-Miracord Mark II, in the same price class as the Dual and Garrard models, has the same basic features necessary for discrete-disc play. This unit and those in Figs. 4-2 and 4-3, will prove ideal for matrix-record playback.

frequency loss, so it would be capable of recovering the discrete disc's subcarrier throughout the disc's play.

Receivers

You may think the matrix-versus-discrete squabble couldn't extend beyond the adapter-amplifier component types, but if so you couldn't be further from the truth. Putting a matrixed four-channel signal out on the FM band is the easiest thing in the world to do if the station is already set up for stereo. The requirement? Nothing. Just play a record encoded in four channels. Everything that is to be done can be done at home. Stations that would like to generate their own four-channel material, as from a live concert, for example, would have to be outfitted with equipment from one of the major suppliers of matrixing gear, but even that isn't difficult for most of them are only too anxious to get broadcasters to use their systems. Sansui (and probably others) have offered the "extended loan" of encoding gear to broadcasters

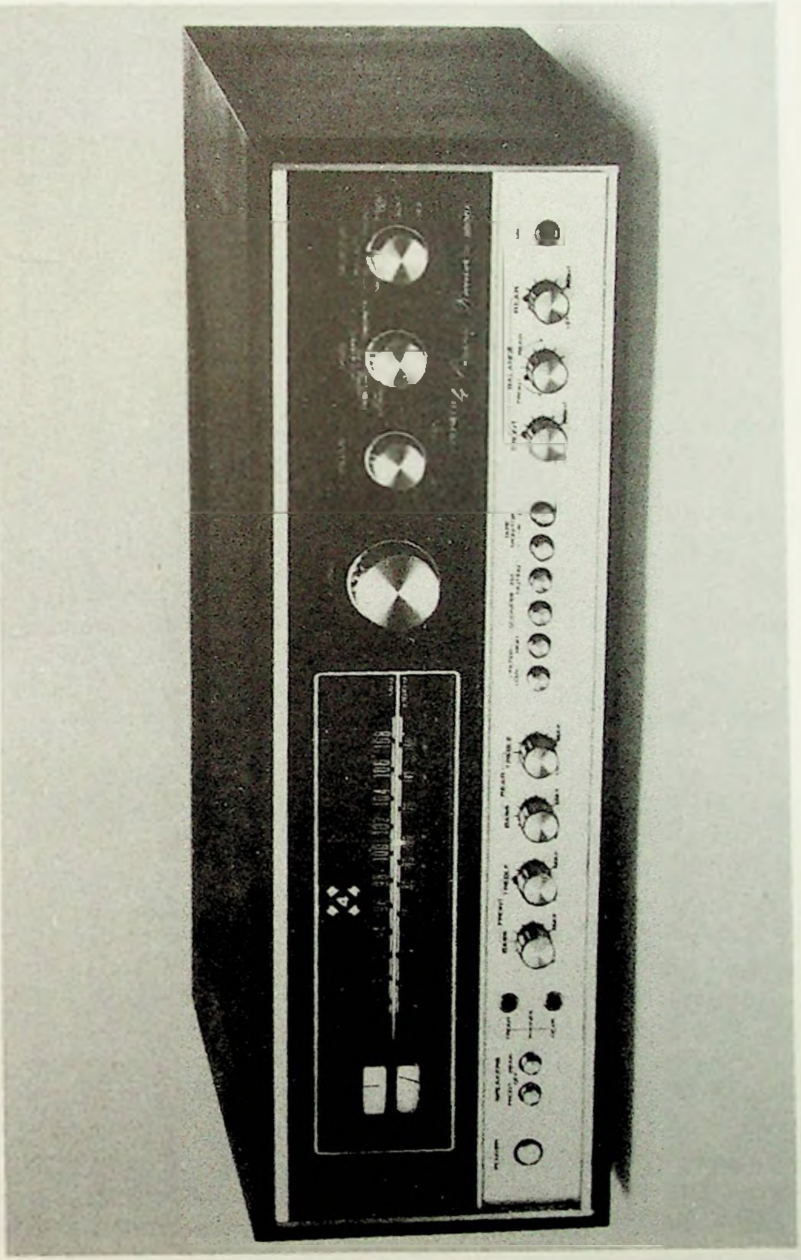


Fig. 4-5. Pioneer's bid for the quad market is a receiver that is ready for anything; the QX-8000A unit decodes matrix in two formats as well as discrete.

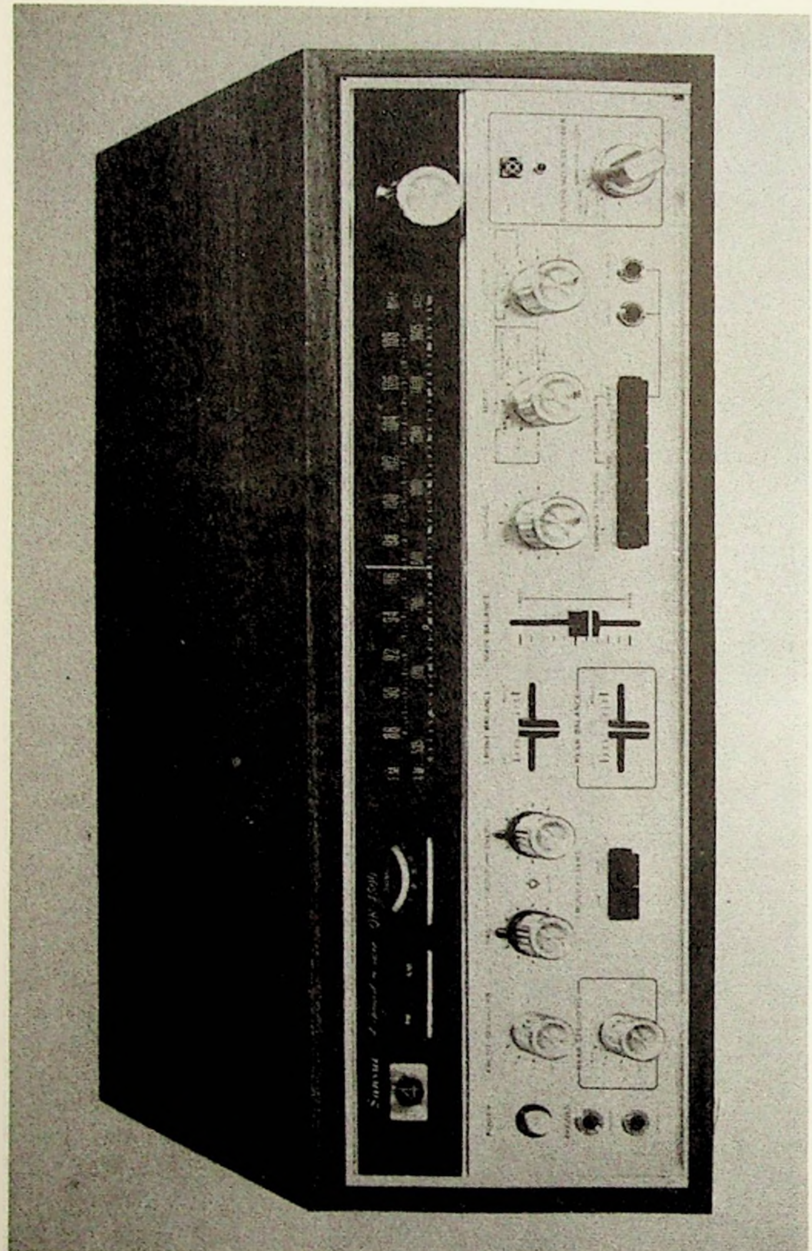


Fig. 4-6. Sansui elected to specialize in its own matrixing encode-decode system. Still, the QR-4500-4 receiver pictured has a sufficient degree of control that matrix formats of competing systems can be synthesized.

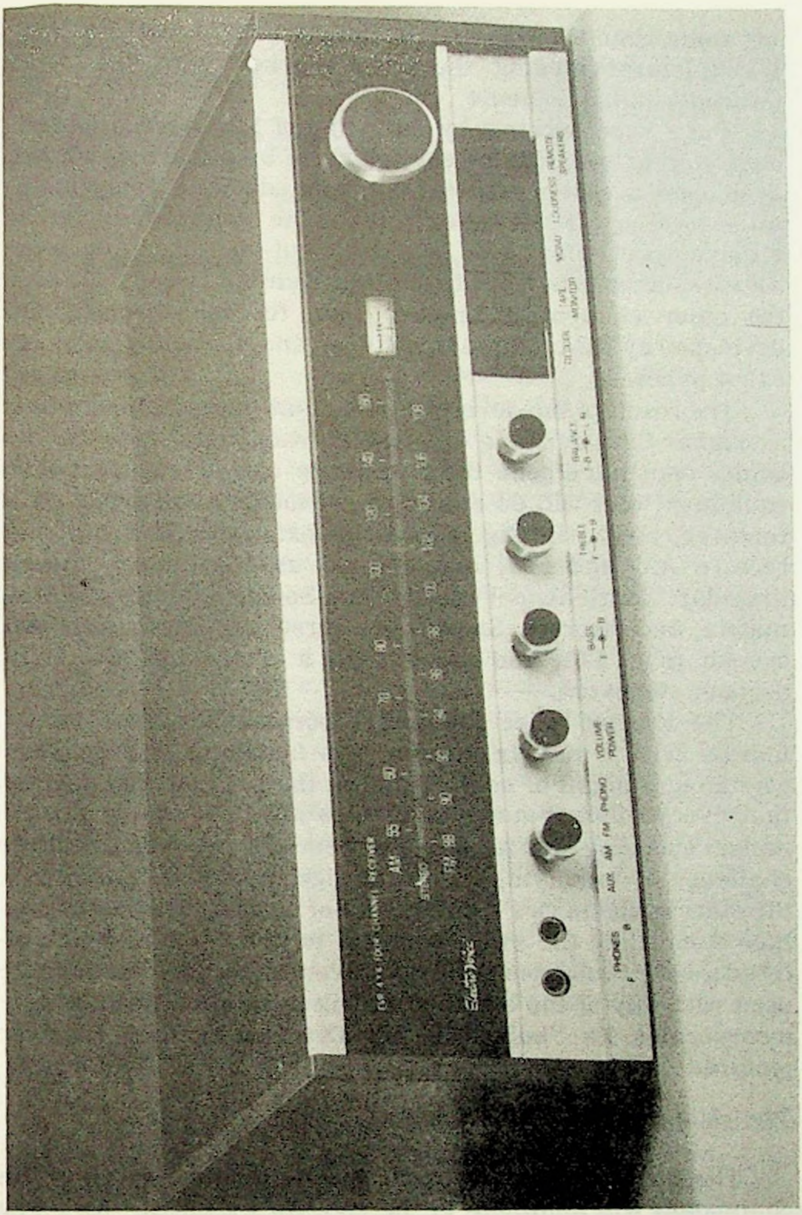


Fig. 4-7. The Electro-Voice 4 X 4 receiver is a high-quality AM-FM unit that comes equipped with the company's "top of the line" four-channel decoder system built in. A "separation enhancement" circuit in the decoder section cuts the level in the rear to enhance front-to-back separation when equal phase and level appear at the front speakers (indicating presence of a front-and-center audio source).

for some time, because the ultimate benefit will be to the manufacturer holding the card marked with the "most desired" matrix scheme.

But a young upstart by the name of Lou Dorren tossed a stick of dynamite into the easy world of broadcasting when he developed a system for transmitting discrete four-channel sound over one FM carrier without the requirement for additional spectrum space. So it isn't cut-and-dried any more. The consumer is faced with another element to consider—and the other element is another plus for the discrete disc developed by RCA. The race is still on and discrete gains a few extra paces.

The result of the development is that manufacturers have a tougher time deciding upon which way to go. A few of the bolder ones have gone the safe route already by producing equipment that will do everything. Pioneer's QX-8000A quad receiver, shown in Fig. 4-5, is an example. This unit can receive AM and FM, and can be used in mono, stereo, "regular" four-channel matrix (the Sansui system), CBS-SQ matrix, and discrete. Sansui, of course, gambled on its own system (Fig. 4-6), and brought out a whole line of matrix-decoder receivers.

The Electro-Voice Company, perhaps the first on the market with a matrix system, ran headlong into an overwhelming amount of competition in the field. It just seemed that every other manufacturer was out to design a private matrix encoding and decoding system. Electro-Voice met the challenge by modifying its own design to be compatible with the other systems (we'll get into all of these a little later), and then submitting the whole problem to a computer, which offered a very sound design for a universal decoder that can be used with any of the existing matrix systems. The company incorporates its "baby" in the 4X4 four-channel receiver pictured in Fig. 4-7.

The Shape of the Future

There aren't many who will profess to know which of the contenders will dominate in the end. As we noted, there are four basic competitors for the encoder-decoder market in the matrix field, and one strong company with a great deal to offer in the discrete field. It's time to examine both modes and see what they have to offer.

Chapter 5
Discrete Four-Channel Reproduction

Now that "quad" is a magic word in hi-fi circles, there are equipments of every description appearing on the market, and a variety of approaches to achieving the same end. The buyer is placed in the discomforting spot of trying to decide which system shows the most promise of being king of the mountain when all the other contenders for the title have fallen. With a little insight into what the various systems involve and a look at the manufacturers who are vying for the number-one slot, you should be in a better position to judge the merits of the various techniques.

To recap, there are two basic approaches—discrete and matrix—and four currently popular techniques for deriving four channels using the matrix approach. Discrete means four independent channels of information, none of which need carry any of the information on any of the other channels. Matrix means mixing the four channels so they fit onto two channel carriers, and decoding the two carrier channels on playback so that the original four are obtained. But the matrix system cannot decode with 100 percent efficiency, so each of the four channels contains the information on all the other channels, but at a level that is considerably lower than the information designed for that particular channel.

DISCRETE TAPE

It is no problem for a tape unit to reproduce four discrete channels of information simultaneously. The only requirement is a tape deck with twice the number of "electronics" normally found in stereo decks, plus a single head with four discrete (separate) gaps, such as the unit pictured in Fig. 5-1. Heads such as the type required for four-channel sound reproduction have been available for years—at least since the advent of the 8-track cartridge player. As Fig. 5-2

Fig. 5-1. Four channels can be easily handled by tape units, which can handle either discrete signals or four matrixed channels on two tracks.

shows, cartridge units themselves have entered the quad world, and are just as strong contenders as any other medium going at the moment.

Recording is no problem for the tape unit, either; just connect one microphone or musical instrument to each of the four inputs of a cartridge or reel-to-reel quad tape unit and turn on the machine. When you've made your tape, rewind and play it back on four separate amplifier-speaker systems. Just like that!

But with phonograph records, the story takes a whole new turn.

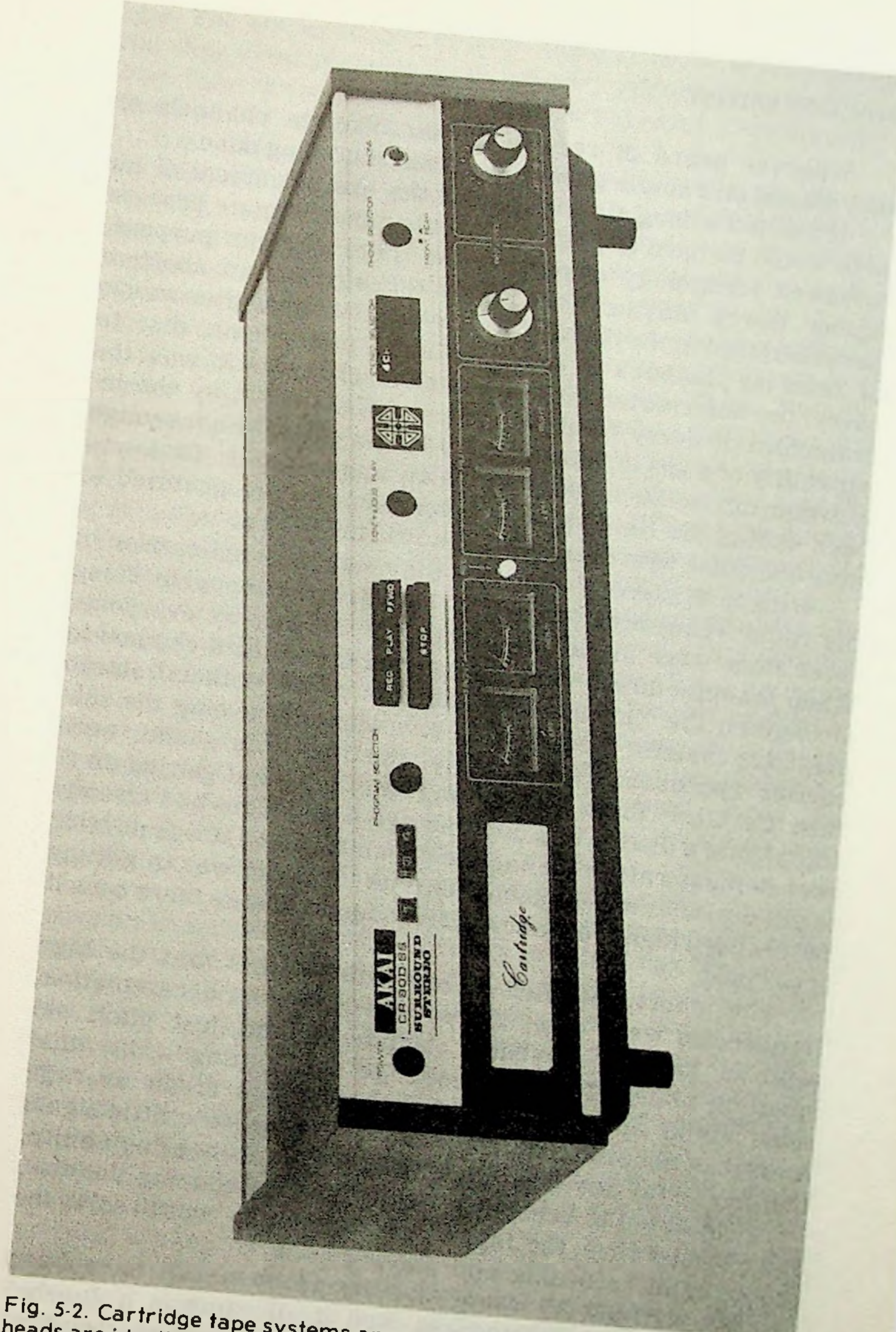


Fig. 5-2. Cartridge tape systems are made to order for quad. The 8-track heads are ideally suited for 4-channel record and playback.

DISCRETE DISC

Whoever heard of recording four discrete channels of information on a single-groove disc? But it is being done.

It started a long time ago, and the exact moment of its birth would be hard to pinpoint, because the ultimate process borrowed technology developed for quite another purpose. Author Harry Maynard attributes the birth to an abortive stereo attempt in the fifties, whereby an FM subcarrier was to be used on playback of a conventional monophonic disc to carry the information for the second channel. It was the brainchild of Jerry Mintner, who was frustrated by the incapability of discs in those days to take a carrier high enough in frequency to be modulated by an audio signal. That was back during the time recording companies were gratified to whip out a disc with even one clean 10 kHz signal on it.

With 45-45 stereo recording, the need for a subcarrier in the radio-frequency region was obviated, so thoughts along those lines were discarded or buried. But not by everyone. When manufacturers were trying to create a third channel to strengthen the "phantom" middle of a conventional stereo playback system, there were a few who kept eyeing the subcarrier approach. Theoretically, the idea was sound, even then. Certainly there was nothing limiting about getting an rf signal onto a disc in this day and age. Cartridges had already been demonstrated that had the capability. And it was no trick to get a disc to accept such a carrier. The rub was in getting the encoded high-frequency carrier signal to stay there once it is pressed.

The unfortunate fact of life with discs is that the high frequencies wear away rapidly through dust accumulation, wear of inferior playback equipment, and just plain old handling abuse. The big challenge was getting a disc that could stand up under the strenuous rigors of an average American household and still deliver the delicate little signal that lies so far above the upper limit of man's hearing ability.

RCA opted to accept the challenge of producing the disc, and its sister firm, the Japan Victor Company, was to solve the problems of recording and playing back.

The answers all came and the problems all have been watered down to nothingness, and it all came in a gunshot series of breakthroughs over a remarkably short period of time. The achievements of RCA and JVC were so sensational that they drew plaudits from industry reporters at the same time they drew jeers from competing—and disbelieving—giants of the recording industry.

To retain compatibility with mono playback equipment, the RCA-JVC disc is mixed in a manner not too dissimilar from the way two-channel matrixed signals are combined. The combined signals for all four channels make up the "sum" content. Channel-to-channel difference levels make up the "difference" content. The difference signals are considerably lower in level than the sum signals, of course, so they are prey to deleterious noise-with-signal conditions. Thus, the difference signals, at the time the recording is made, are processed by a specially designed noise reduction circuit, where the difference between the level of the noise and the level of the signal is increased drastically. After being purged of the noise accompaniment, the difference signals are used to frequency-modulate a carrier in the 45-50 kHz region.

The sum signals are processed by a phase-control circuit, which delays the signal so that it can be mixed again with the noise-free difference signals. The delay here allows the same signal to be extracted later, without being forever melted in with the difference signal. The resultant composite signal is the one that is actually pressed onto the stereo disc. Figure 5-3 contains two photomicrographs that show both stereo and four-channel discrete disc grooves. Note the apparent complexity of the "discrete" grooves.

The new discrete four-channel record was test-marketed in Japan, its birthplace. The idea was to get a preview of the possible problems that might crop up in actual use. The record reportedly sold well in stores, even though it carried a price tag $2 more than conventional stereo records. And problems **did** show up. JVC found that discs played on "normal" equipment deteriorated, and decided that people who used the disc with their ordinary stereo equipment were unwittingly carving off the high-frequency subcarrier. JVC had to stipulate that the records were to be played only on the company's specially designed four-channel sound equipment.

This basic lack of compatibility was a bad thing, and seemed to have the potential of killing the unit's success even before it could be realized. To keep the program in the running, JVC went about the task of designing a stylus and cartridge that could be used with conventional stereo equipment,

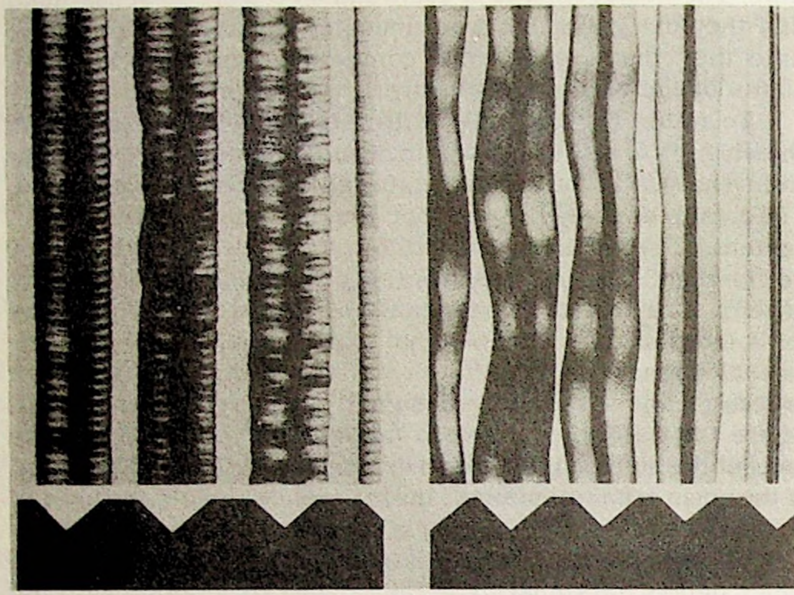


Fig. 5-3. The grooves of a discrete disc are used to contain the composite signals of four channels without changing the basic groove configuration. Photo A is an enlarged view of a four-channel discrete disc above a cross-section view of the grooves. Photo B is an enlargement of an ordinary stereo disc.

but one that would be good to the fragile subcarrier with its all-important information. At the same time, RCA in this country began to get serious about getting an improved disc ready for marketing.

The Shibata Stylus

The Japanese came through first, and the result of the effort was the now-famous Shibata stylus. From outward appearances, the Shibata looks to be no different from an ordinary modern elliptical stylus of current manufacture. But under the microscope, some enlightening differences show up readily. The Japanese stylus (Fig. 5-4) comes to a gently curved point, whereas the conventional elliptical stylus has the appearance of a rounded cone with two sides shaved off. The Shibata's configuration allows it to seat firmly into the record groove, making contact even when elliptical styli are forced up and nearly out of the record grooves.

The Shibata stylus, incidentally, is available for general use, but dealers report they are having trouble keeping up with the demand. According to JVC, the enhanced high-end response—it will respond to signals of 50 kHz and beyond!—will improve the performance of ordinary systems; and further, that the new stylus will cause considerably less groove deformation than existing elliptical styli.

Reports like these are reasonably common from overenthusiastic ad agencies, whose job it is to sell the wares of the companies they represent. Before buying the whole package, I took the trouble of citing a few of the maker's claims and sending them to Shure (a domestic manufacturer of high-quality cartridges and styli), along with a question as to how that firm intends to answer the challenge of Shibata's obviously superior-performing stylus and cartridge assembly. The company sent me literature describing its products and photos of its better units (which are very good, by the way,

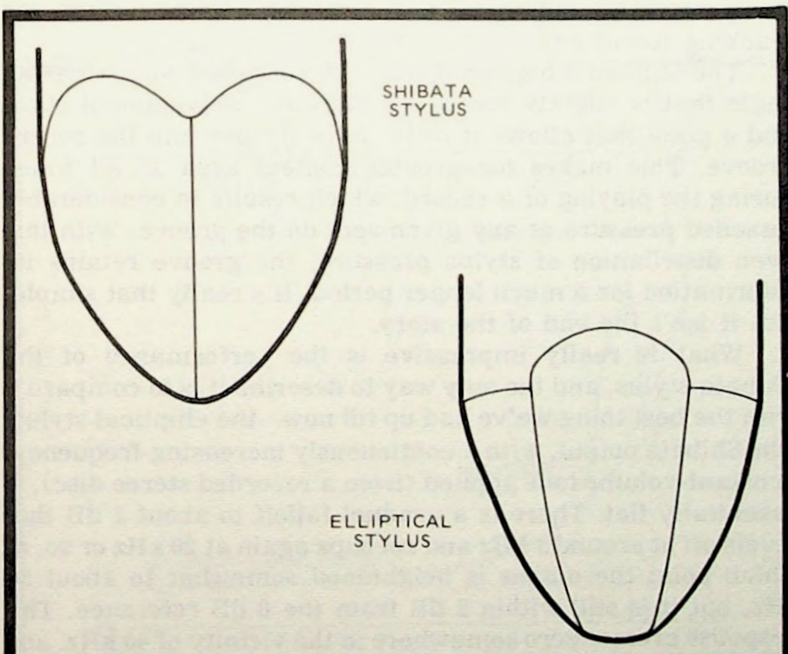


Fig. 5-4. The Shibata stylus seats better into a stereo groove than an elliptical stylus. As a result, there is less pressure on any given portion of a record surface; therefore, the disc lasts longer and retains its information better than at first thought possible.

and are products that I use for my own personal audio system) —but no mention was made of Shibata, the JVC cartridge, or any pending new developments at Shure. Nonetheless, it seems fairly safe to assume that no on-the-ball manufacturer of Shure's reputation and status in the field is going to sit comfortably while the world moves on around it.

If the question of why domestic stylus makers haven't already produced a stylus capable or responding to radio frequencies occurs to you, I'd like to suggest that you think not harshly of these firms. After all, the key feature of the Shibata is its ability to play records with extremely high frequencies without destroying them. Until now there have been no records with such high frequencies on them, so there has been no need for a stylus that would preserve them after many playings. Indeed, the elliptical and the spherical styli in current use, when used with good equipment and when installed properly, will allow records to keep their sound content for practically unlimited periods, particularly where the tracking forces are down in the one-gram region.

The Shibata's biggest edge is its configuration. It has an angle that is slightly less steep than our conventional styli, and a point that allows it to fit down deeper into the record groove. This makes for greater contact area at all times during the playing of a record, which results in considerably lessened pressure at any given spot on the groove. With this even distribution of stylus pressure, the groove retains its information for a much longer period. It's really that simple. But it isn't the end of the story.

What is really impressive is the performance of the Shibata stylus, and the only way to describe it is to compare it with the best thing we've had up till now—the elliptical stylus. The Shibata output, with a continuously increasing-frequency, constant-volume tone applied (from a recorded stereo disc), is essentially flat. There is a gradual falloff to about 2 dB that levels off at around 5 kHz and recoups again at 20 kHz or so, at which point the output is heightened somewhat to about 30 kHz, but it is still within 2 dB from the 0 dB reference. The response crosses zero somewhere in the vicinity of 40 kHz, and hangs in there nicely to well above 50 kHz. A typical performance curve and comparison plot are shown in Fig. 5-5.

As shown, the elliptical stylus performs similarly up to about 30 kHz, but the output degrades seriously and rapidly beyond this point, and at 50 kHz, the output is actually more than 10 dB down. Even this figure would make the elliptical stylus usable for the discrete records, but there are other complications, such as cochannel interference, or crosstalk.

It is important to keep one of the two stereo channels of a record 20 dB or so below the other. We have already learned that it is virtually impossible to separate the two signals totally, because they are both picked up by a single stylus. But it is possible to keep the signals far enough removed from one another in volume level that the left channel does not appear to exist on the right channel and vice versa. With a conventional elliptical stylus, the adjacent-channel rejection begins to deteriorate rapidly at frequencies above 10 kHz. As shown, there is some improvement between 20 and 30 kHz, but the deterioration beyond that frequency is sudden and devastating.

The Shibata stylus, on the other hand, has a smooth, relatively constant performance in this regard. The second channel stays well below the first, regardless of how high the

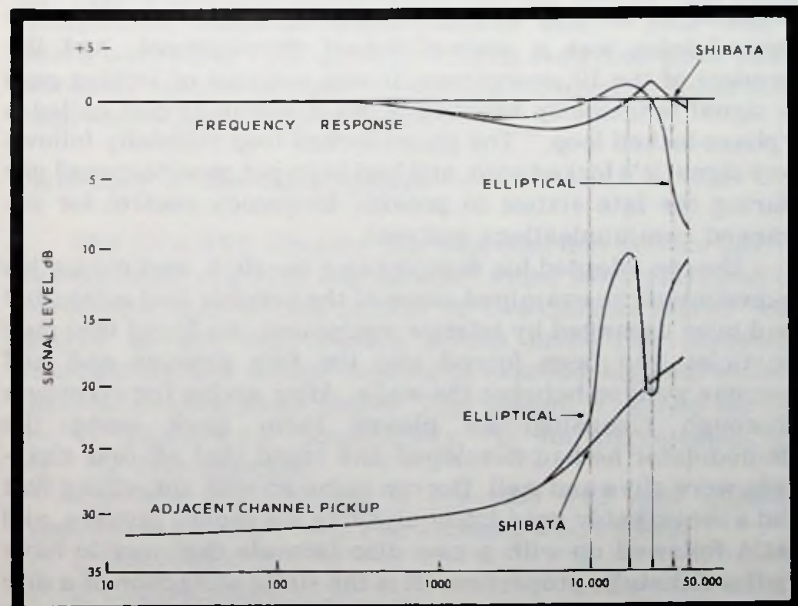


Fig. 5-5. The Shibata stylus really begins to show its colors in the region above 20 kHz, where elliptical styli start their downward thrust. Crosstalk gets very bad at higher frequencies with elliptical styli, too; and the Shibata holds excellent separation all the way up to 50 kHz.

frequency it is reproducing. There is almost 15 dB of separation even as far out as 50 kHz.

The RCA Disc

While JVC was making audio news with its cartridge and stylus, RCA was doing things back here in the States. In the first place, the company had made some experiments with first-generation prototype discs and had been disappointed by the results. As noted previously, the records that had been test-marketed showed what appeared to be irretrievable degradation after having been played on inferior equipment.

RCA felt that if the demodulator—the unit that recovers the high-frequency signal and converts it into audio—were more sensitive, the degraded signal could still be used to supply the information needed to produce the four channels.

Lou Dorren was still working with his four-channel discrete FM broadcast system in California, and was using a demodulator that he had designed, which was purportedly superior to the one in current use by RCA and JVC. His demodulator was a state-of-the-art development, and the product of the IC generation; it was capable of locking onto a signal tenaciously because of an in-vogue circuit called a "phase-locked loop." The phase-locked loop faithfully follows any signal it's locked onto, and had been put to widespread use during the late sixties to provide frequency control for advanced communications systems.

Dorren adapted his demodulator for RCA, and during his experiments he examined some of the records that ostensibly had been damaged by inferior equipment. He found that dust particles had been forced into the fine grooves and had become wedged between the walls. After giving the records a thorough cleansing, he played them back using the demodulator he had developed and found that all four channels were alive and well. Dorren came up with something that did a remarkably good job of cleaning the record grooves, and RCA followed up with a new disc formula that was to have better antistatic properties. (It is the static attraction of a disc that makes it so susceptible to accumulation of dust particles in the first place; and elimination of static is the first and most logical step in eliminating the dust problem that all audiophiles are plagued with constantly.)

The goings-on in the discrete disc end of the industry are still happening, and much of the information included in this chapter reflects the changing events of just the past few months. As 1972 closed, RCA admitted to the existence of small snags, but they are not significant enough to keep the discrete disc from being made generally available. One problem is in the recording process: to be certain that all frequencies are encoded and recorded faithfully, RCA cuts the record at half speed, which drops all recorded frequencies by a full octave. On playback, however, the speed goes back up to 33 1/3, and all the encoded frequencies are recoverable in their original state.

Early in the development effort, RCA promised to end up with a disc that would sell for the same price as its currently available two-channel type. With the extra time involved in recording, it appears that this will prove a real challenge, but even as recently as the 1972 NAB conference, RCA was holding to the pledge. As a matter of fact, one of the statements made by Lou Dorren in defense of RCA at that conference was that RCA plans shortly to turn its entire library over to "one single, compatible mono-stereo-four-channel disc." This was in sharp contrast to CBS' position with its matrixed discs which, like other matrix-format records, sell for about a dollar more than conventional stereo types.

Summary of The CD-4 System

The RCA-JVC discrete disc reproduction system has the compatibility of giving four-channel sound when played on the associated reproducing system or two-channel stereo when played on the conventional stereo unit. The associated reproducing unit is also capable of reproducing conventional discs as a two-channel stereo. This system is named the CD-4—"C" standing for compatibility, "D" for discreteness, and "4" for four-channel.

The new CD-4 system is provided with four qualities considered essential in a four-channel disc record:

1. Discreteness
2. Compatibility
3. Economy
4. High Fidelity

In this particular disc, independent sources of four channels are reproduced separately from respective channels

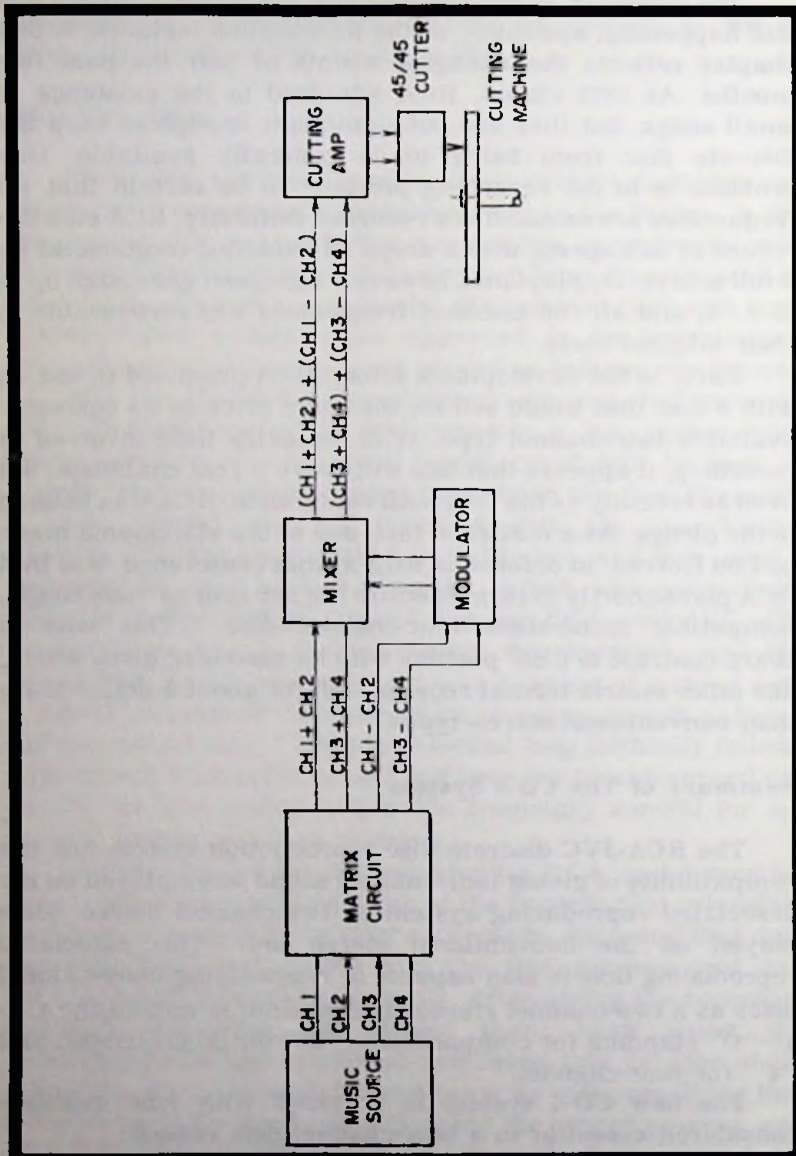


Fig. 5-6. Discrete signals employ a matrix, too, but the only similarity between this and the "matrix quad" unit is the name. Four discrete channels are algebraically encoded so that two sets of output signals remain. The difference signals are used to modulate a high-frequency carrier, and then all signals are mixed into two. One side of the stereo groove will contain all the left-side material, the other contains all the right-side material.

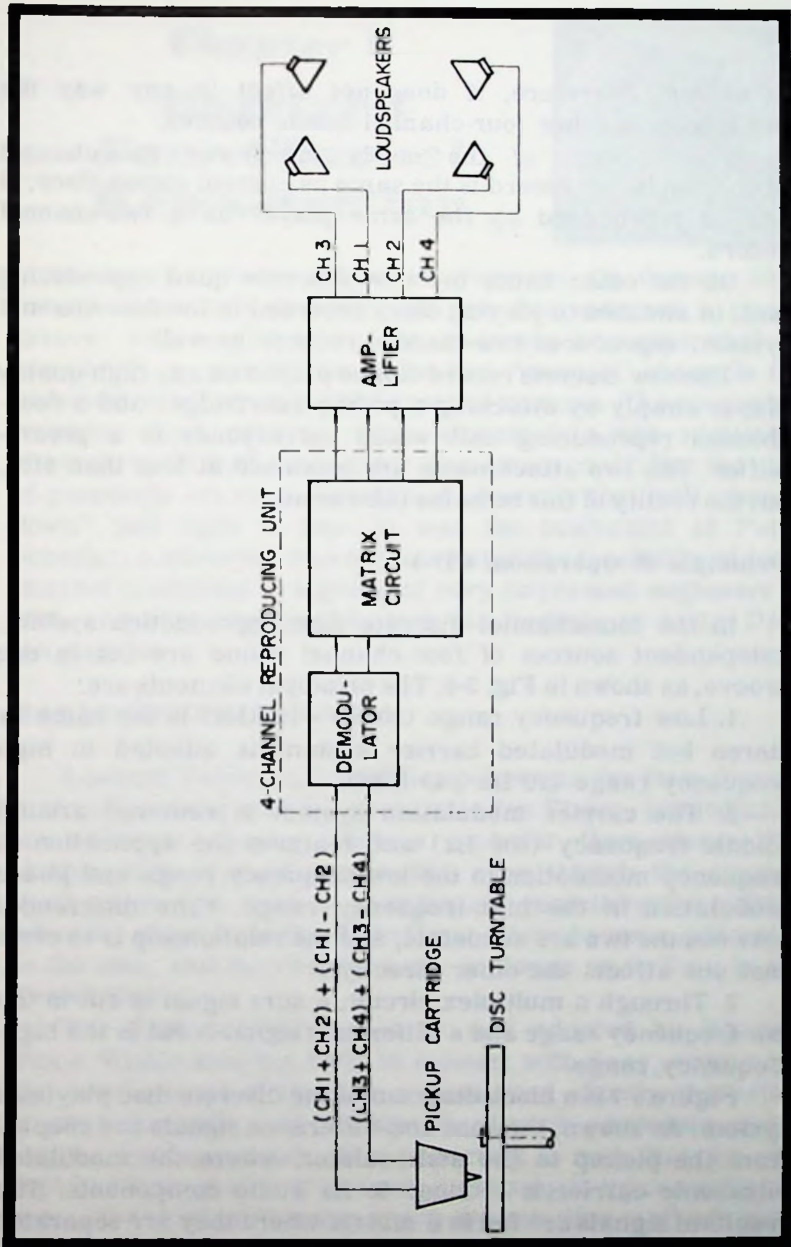


Fig. 5-7. On playback, the demodulator separates the audio from the high-frequency encoded signal, and a matrix decodes the algebraic resultant sums and differences down into the four discretes.

in unison. Therefore, it does not affect in any way the production of other four-channel music sources.

Tone quality of the newly introduced four-channel phonograph disc record is the same as current stereo discs; it can be reproduced by the same player as a two-channel record.

On the other hand, the new discrete quad reproducing unit, in addition to playing discs recorded in the four-channel system, reproduces two-channel records as well.

The new discrete record can be played on any high quality player simply by attaching a pickup (cartridge) and a four-channel reproducing unit which corresponds to a preamplifier. The two attachments are promised at less than $100, but the reality of this remains to be seen.

Principle of Operation, CD-4

In the four-channel discrete disc reproduction system, independent sources of four-channel sound are cut in one groove, as shown in Fig. 5-6. The principal elements are:

1. Low frequency range (30 Hz - 15 kHz) is the same as stereo but modulated carrier system is adapted in high frequency range (20 Hz - 45 kHz).

2. The carrier modulation system is centered around middle frequency (800 Hz) and features the application of frequency modulation in the low-frequency range and phase modulation in the high-frequency range. (The differences between the two are academic, and the relationship is so close that one affects the other directly.)

3. Through a multiplex circuit, a sum signal is cut in the low-frequency range and a difference signal is cut in the high-frequency range.

Figure 5-7 is a block diagram of the discrete-disc playback system. As shown, the sum-and-difference signals are coupled from the pickup to the demodulator, where the modulated ultrasonic carrier is reduced to its audio components. The resultant signals are fed to a matrix where they are separated into their original discrete formats. Four amplifiers give the required gain and the signals are passed along to the speakers to complete the system.

Chapter 6
Matrixed Four-Channel Reproduction

Matrixing is a double-edged development. On the one side, four channels are encoded onto the two sides of a stereo groove; and on the other, the channels are separated as completely as possible from the two "carrier" channels for distribution to four amplifier-speaker systems. The concept of breaking down existing stereo signals into two additional channels is old stuff, even older than stereo itself. But the idea of purposely creating a couple of channels that will "break down" just right is new. It was the brainchild of Peter Scheiber, a musician who demonstrated the feasibility of four-channel matrixing to a group of very impressed engineers in 1970, and who subsequently took his findings to Audio Data Corporation.

EARLY DEVELOPMENTS

Leonard Feldman, a prolific spokesman for four-channel matrixing, along with his associate, Jon Fixler, immediately set about to develop such a unit for themselves. The mathematics seemed reasonable, even though the concept itself seemed to have a paradoxical something-for-nothing unreality about it. At the same time, others became interested in the idea, and the race was on—in Japan as well as in the United States.

The Feldman-Fixler scheme was adopted by Electro-Voice. Within months, CBS, in concert with Sony, announced the development of its own system, and shortly thereafter several domestic and foreign hardware producers made similar announcements. It is important to note that the concept in all cases of four-channel matrixing is intrinsically the same; the chief differences are in the encoding methods used. The concept is based on time and level. The differences—how much time between primary and secondary recorded signals and how much of a level difference exists—determine the

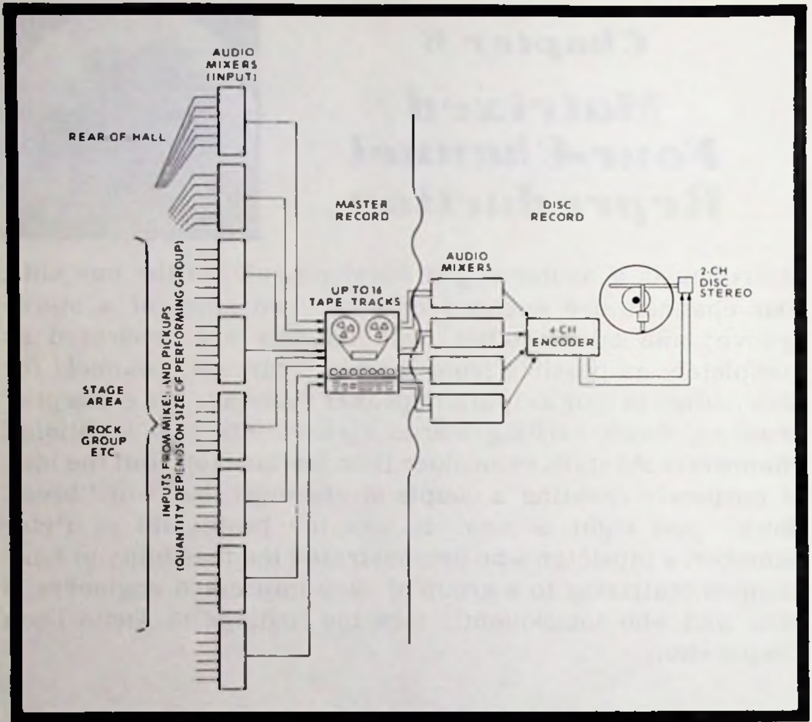


Fig. 6-1. Modern stereo and four-channel recording is a mixing down process. The tape recorder sees as many as 16 inputs, any number of which might be from mixers. The playback is matrixed through a four-channel encoder, which produces two output signals.

proportion of the remaining three channels inherent in the desired channel, and the separation of the desired channel from those other channels.

FOUR-CHANNEL MATRIX RECORDING

The overall scheme of recording in four channels using a matrixed format is shown in Fig. 6-1. This arrangement is applicable to all current matrixing systems, which differ in circuit subtleties rather than in basic operational theory. Usually, at the time of the original recording of a musical program, as many audio mixers are employed as there are tracks on which to record the signals, then several microphones or music instrument pickups are fed into each mixer. The recording engineer can monitor the signals at individual input mixers, but once the mix has been accomplished, there is nothing he can do to withdraw one mixed signal from another. This is one of the main reasons audio engineers like to mix as few signals into one as possible during the initial recording. The best possible approach would be to use one mike or instrument pickup for each tape channel to be recorded, then accomplish the mixing at some stage further down the line. The size of some music groups—such as orchestras or choral groups—precludes this, however, and the alternative is premixing.

The mixed signals are spread out onto as many tape tracks as the recorder will accommodate. Again, the philosophy is that ultimate sound mixing can be accomplished later, in the relative calm of the sound lab. Instruments that are too loud, vocalists that have been obscured by low level for one reason or another, off-key sounds, audience coughs, and extraneous noises can all be mixed out later.

The playback of the master tape is where the artistry of the audio engineer must come through, for here is where he performs the function of editor, musical director, and technician. He now mixes down the tape tracks, as many as sixteen, but often less, depending on group size and other factors.

He can feed all sixteen tracks into one channel of another audio mixer if he chooses; then, monitoring the content of that mixer, he can selectively adjust the proportions of each input, on a percentage basis. He will only feed four signals to the matrix encoder, but each of those four signals will contain perhaps elements of all the channels picked up by every mike and pickup deployed at the original performance. The percentage of what he incorporates for the channel which will eventually become "left back" might be composed of 60 percent signals picked up by the instruments and mikes feeding the mixer positioned at the left rear of the original recording studio or auditorium, 20 percent signals picked up by adjacent mixers, and 20 percent signals that are an aggregate of the signals on other tape tracks.

The encoder delays the signals of some tracks, and provides the summation and differencing necessary to be able to extract the encoded signals later. The signals thus matrixed are combined onto two stereo channels, and recorded in a 45-45 format. That is, one of the two coded output signals modulates

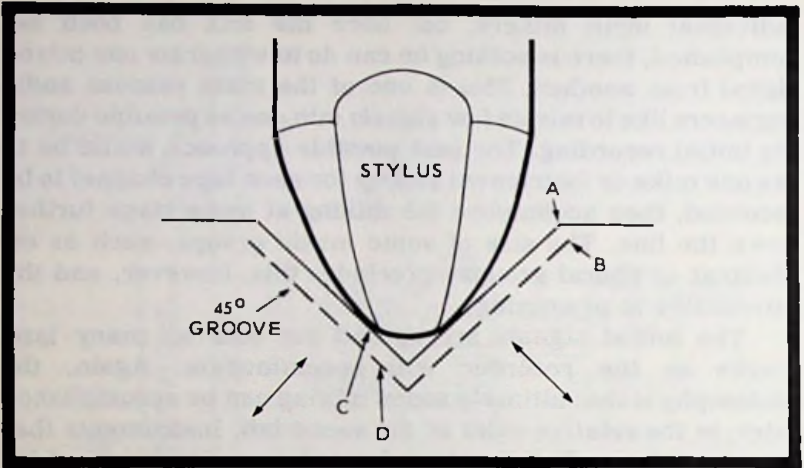


Fig. 6-2. The stylus moves in two planes—one between A and B, which represents the information for the left channel. The other groove wall is curved to allow modulation between C and D. In the cartridge, two magnetic fields exist. The A-B motion has little effect on the C-D magnetic field and vice versa. If the A-B motion and the C-D motion are similar, with one occurring before the other, a helical motion of the stylus results. The helical motion is the rear information—clockwise for left, counterclockwise for right.

one 45-degree groove wall, and the other coded output signal modulates the other groove wall.

Signals destined for the left rear channel impart a helical motion to the recording stylus. The motion is clockwise as you stand facing the stylus with the record feeding toward you. Signals for the right rear channel impart a counterclockwise helical motion. The modulation concept is pictured in Fig. 6-2.

The helical motion in both directions is an unimportant factor, and is brought about as a natural result of coding one channel's information slightly before or behind the other channel. Look at the drawings of a record groove shown in Fig. 6-3; note the similarity between the modulations of the record groove walls and sine waves. If both signals are similar in character—as from a single music program—the wall modulations will appear similar. Now, if the "undulations" of one wall are delayed even the minutest fraction of a second, the recording stylus can turn slightly and modulate the information while it is still modulating the opposite groove wall. The result is as shown in the sketches—one groove will appear to be a repeat performance of the other wall, but delayed by no more than a portion of an "undulation." Since either groove wall can be made to lag behind the other, this is the same as one wall being made to lead.

The leading groove wall causes the stylus to react earlier than the other wall, forcing the stylus up, but toward the other groove wall, which allows the stylus to fall slightly sooner toward the first wall. The overall motion is helical, and may be either clockwise or counterclockwise, depending on which of the two groove walls leads and which lags. But there is motion on two other planes as well: the two perpendicularly oriented 45-degree up-down planes characteristic of the stereo disc.

It is easy to see that there are still only two **basic** channels recorded, and only two basic channels appear on playback. The perpendicularly separated undulations of the two groove walls are the two basic channels to which your conventional stylus and cartridge will respond. The juxtapositioning of those two channels, however, which is precisely the same as controlling the phase of those two signals, creates **level differences** of the same program.

Phasing, of course, can be used with monophonic recordings to create the illusion of stereo. Two fully phased cuts of a vocalist, both at the same level, will place the vocalist front

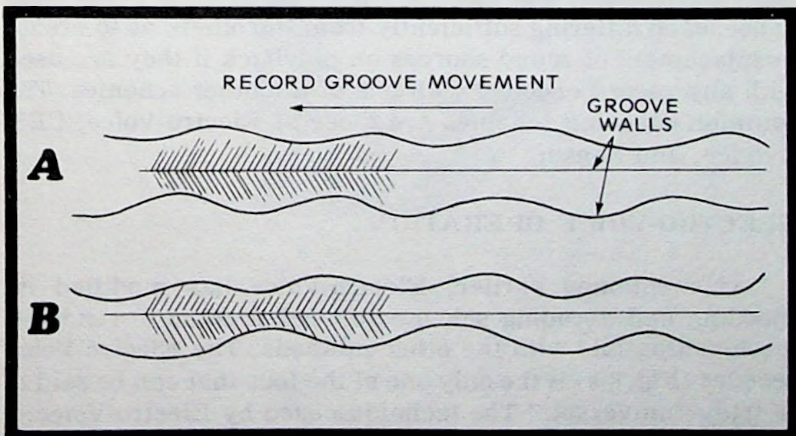


Fig. 6-3. View of record groove, looking down from top. In sketch A, the bottom side of groove lifts the stylus, which falls toward other side and drops, to be lifted again by lower side. This results in a counterclockwise stylus motion that represents the information to be fed to the right rear channel. In B, the upper groove wall's modulation leads the lower wall, and the result is a counterclockwise stylus motion, which represents left-rear-channel information.

and center on playback through a stereo system. But the engineer who is making the stereo cut from a monophonic array of mixed inputs can increase the level of a musical instrument here or decrease the level of another there, and the listener, on ultimate playback, will be able to localize all the intended sources, even though each channel has all the information of the other channel. The difference is purely one of proportion—nothing more. But the results of those proportions warrant some closer attention.

Quad sound is not in even the strictest sense a reproduction of four monophonic signals (or channels). With two additional loudspeakers (and the proper decoder) added to an existing stereo setup, you obtain the same effect as you would with six perfectly synchronized stereo systems operating. Sounds in a full 360-degree area around you can be reproduced, mixed in the air, and localized at will to approximate the original sound field.

One of the unfortunate things about having a number of approaches to one problem (the one of getting four channels out of two) is the matter of how best to accomplish the objective. Each approach is somewhat different from the others. Currently, there are at least four prominent methods in use, each differing sufficiently from the others as to create misplacement of sound sources on playback if they are used with any record encoded with one of the other schemes. The common encoding schemes are those of Electro-Voice, CBS, Dynaco, and Sansui.

ELECTRO-VOICE OPERATION

As mentioned earlier, Electro-Voice has modified its encoding and decoding scheme somewhat so as to render it more compatible with the other methods. The Electro-Voice decoder (Fig. 6-4) is the only one of the four that can be said to be truly "universal." The technique used by Electro-Voice, I should add, is not the same one presented to the company by Feldman and Fixler in 1970; rather, it is the result of E-V's incorporation of all the basic formats and placing the problem into the hands of an electronic computer, which helped to design a best-combination system that would be applicable to all the matrixing methods.

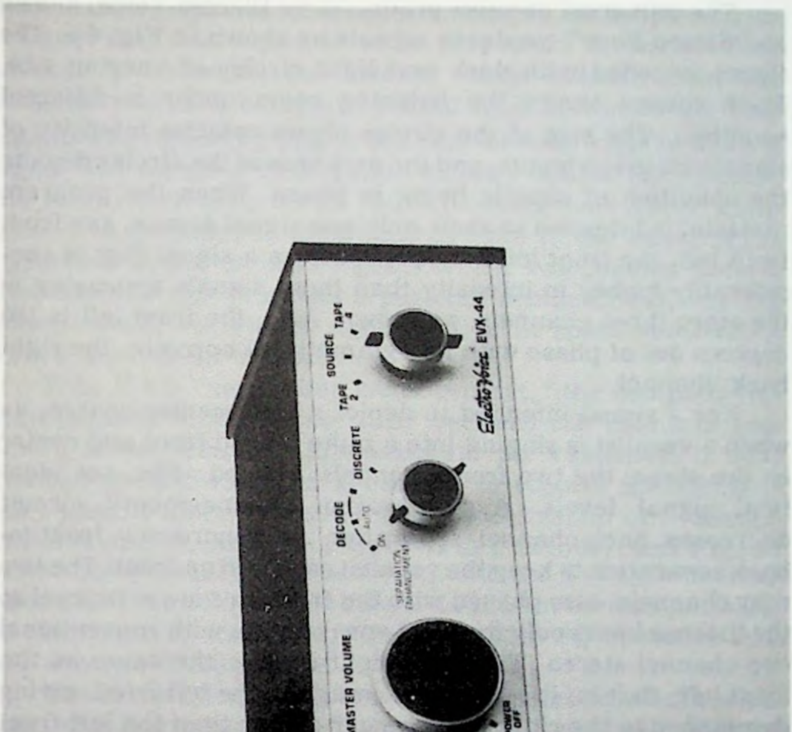


Fig. 6-4. Electro-Voice's EVX-44 quadraphonic adapter incorporates a single integrated circuit to decode all standard four-channel matrixed signals. The unit plugs into any standard stereo amplifier, and must be coupled to an additional stereo amplifier to produce the four quad channels.

The universal decoder produced by Electro-Voice, known as "Stereo-Four" produces signals as shown in Fig. 6-5. The figure is coded with dark and light circles of varying size. Each square shows the listening room under a different condition. The size of the circles shows relative intensity of signals for given inputs, and the darkness of the circles depicts the condition of signals being in phase. When the program material is intended to show only one signal source, say from front left, the front left channel receives a signal that is considerably higher in intensity than those signals appearing in the other three channels, as shown. Also, the front left is 180 degrees out of phase with its symmetrical opposite, the right back channel.

For a signal intended to depict a front-center source, as when a vocalist is singing into a mike placed front and center on the stage, the two front channels, phased alike, see identical signal levels. A "separation enhancement" circuit decreases back-channel separation and increases front-to-back separation to keep the vocalist centered up front. The two rear channels, also phased with the front, are lower in level so the listener hears only the front speakers, as with conventional two-channel stereo. The front-right case is the same as the front left; that is, it is a mirror image of the left front, giving dominance to the right front corner rather than the left front corner.

Side signals are no problem. For the left side, the front and rear speakers of the left side see the same level and phase, while the two right channels are lower in level and cross-phased with each other. The opposite case holds for right-channel dominance.

Rear-channel dominance—left, right, and center—depends a great deal on the psychology of the listener. The rear corners receive a slightly greater signal than any of the other channels, but the level is not substantially lower than the up-front signal for whatever side is being localized. The opposite side receives signals that are out of phase with the localized side, however. Center back dominance is shown by a straight out-of-phase condition, and all four channels appear at the same volume level. The ear tends to detect out-of-phase signals as originating from the rear, as noted in a preceding chapter.

The classical disadvantages of E-V's original basic decoding system, when used with E-V encoded material, have been in the interpretation of signals destined for the rear. With a signal that was supposed to be pinpointed at the left rear corner, for example, the listener would actually hear the program source as being just behind dead left (and right rear signals as originating just to the rear of dead right). When a signal was intended to originate directly behind the listener, the level would drop on all four channels—sometimes completely to nothingness momentarily, due to the exact phase opposition of the encoded signals on the two-channel record. The effect in this case was a puzzling nebulousness, and the source would sometimes appear to be in space somewhere above the listener's head!

With E-V's "separation enhancement," the company has incorporated subtle phase shifts between the channels (not shown in the sketches) to correct for the vagueness in rear channel localization, and according to reports the result is very good. Figure 6-6 shows signal localization of a Sansui-encoded disc through the improved E-V decoder, and Fig. 6-7 shows signal localization achieved with the E-V system when used with a CBS SQ-encoded disc. As shown, the compatibility is very high. Results using Sansui or CBS SQ decoders with EV encoded discs are not so impressive.

The Electro-Voice universal four-channel decoder has yet one thing more going for it: simplicity. A very basic decoder can be built inexpensively, using a single integrated circuit developed especially for this use. While some equipment manufacturers have gone to the trouble of incorporating multiple decoding circuits to preserve a compatibility with as many encoding schemes as possible, the E-V method seems to be the only one extant with an automatic applicability to all encoding arrangements without component switching.

SANSUI QS DECODER SYSTEM OPERATION

The basic Sansui system is characterized chiefly by symmetry. While some combinations give the effect of long rooms, and others the effect of wide rooms, the Sansui decoder (Fig. 6-8) maintains the apparent shape of a square room. In earlier models, this meant compromising the separation between adjacent channels, but a rapid series of improvements in decoding equipment—into which some very sophisticated electronics has been incorporated—has resulted

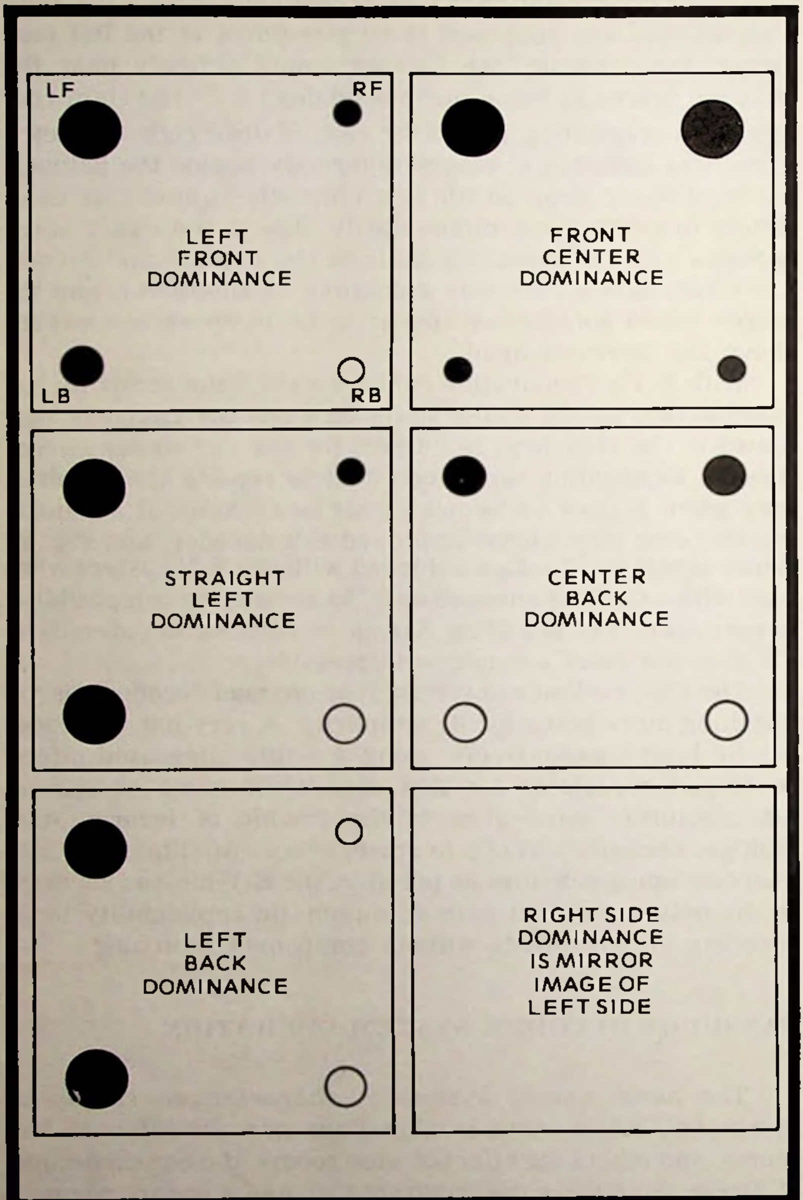


Fig. 6-5. Electro-Voice decoding of E-V matrix. The size of the circles shows signal intensity. The dark circles represent in-phase signals and the light circles show out-of-phase signals. As shown, E-V's weakest area is "center back," which represents input signals that are 180 degrees out of phase with each other.

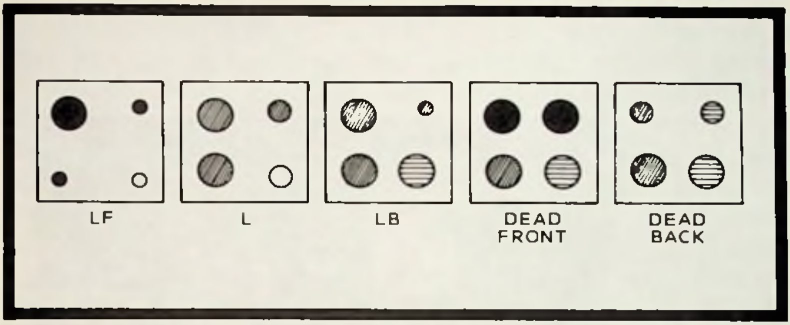


Fig. 6-6. Sansui disc played through E-V decoder shows excellent compatibility. Phasing is shown by lines. Close-spaced lines are phase leads; wide-spaced lines are phase lags. Dark is in phase, white is 180 degrees out of phase.

in an apparent increase in left-to-right as well as front-to-back separation.

Sansui likes to discuss four-channel stereo in terms of sound "fields," which is really not too bad an idea, because the listener is indeed exposed not to a point source of sound at any time, but a field of sound that originates from (or reflects from) a number of sources. The object of the Sansui system is to re-create, as nearly as possible, the total sound field present at the time of the original recording.

Speaking before a group of recording specialists, FCC men, and broadcasters, Sidney Silver, one-time spokesman for Sansui, said the company has been able to achieve more than 20 decibels of separation between any two channels of its properly encoded and decoded system. This is surely a significant achievement if it is on the level, because a

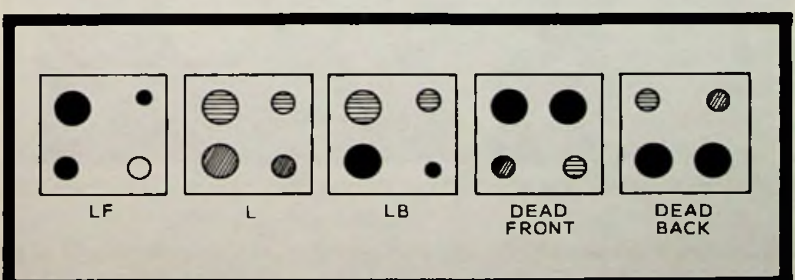


Fig. 6-7. CBS-SQ encoded disc played through E-V's universal decoder shows a high degree of compatibility. (Lines and circles represent functions described in earlier sketches.) CBS-SQ decoder does not possess such a broad compatibility.

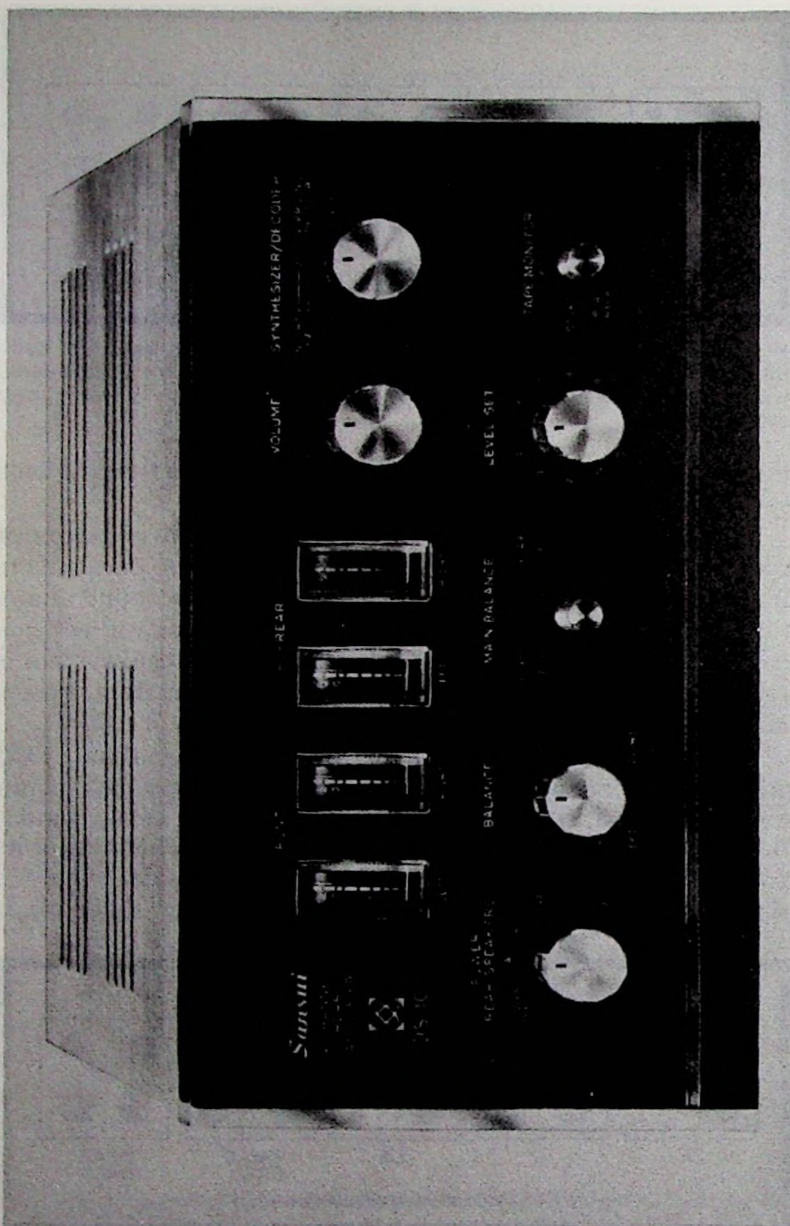


Fig. 6-8. Sansui decoder unit is the only matrix formula to maintain symmetry across all channels. This decoder, one of Sansui's many models, includes a pair of amplifiers for the rear speakers. It is intended for use with an existing stereo system.

separation of 20 dB represents a power difference of 100 times. In other words, if the right front speaker were being driven with 10 watts, no other speaker in the system would be driven with more than 0.1 watt. This type of performance would put a matrix system in the same league as a discrete system. And everybody who knows anything about mathematics, and particularly the "information theory," has been saying "it couldn't be done" for a very long time. One thing seems certain—Sansui's second-generation decoders do indeed achieve 12 dB adjacent-channel separation, which tops any figure quoted by manufacturers as of this writing. The 12 dB figure puts adjacent channels at 6 percent the power level of the dominant channel when a signal is intended to originate at a corner.

Wheher or not Sansui has managed to achieve separation to the extent claimed by Mr. Silver remains to be seen, though 20 dB is as believable as 12 dB . The company does have some very impressive things going for its system, regardless of the separation figures. Of particular significance are two developments: a unique 90-degree phase shifter (discussed briefly in an earlier chapter) and phase modulator circuit, and a volume monitoring system that totally suppresses the channel diagonally opposite the dominant one. Both these techniques can't help but enhance smoothness of reproduction and apparent separation.

The original matrix technique designed by Peter Scheiber incorporated a system that is very close to Sansui's straight-through approach, and suppressing the channel in diagonal opposition to the dominant channel was part of his original plan. Sansui started here, but went beyond, to the extent of automatic level-increasing for dominant channels and phase-shifting to enhance the spatial sound field.

Figure 6-9 is a room diagram showing the volume level of all speakers and the phase relationships of all channels for the basic conditions—front corner, rear corner, straight lateral, dead front, and dead back. Note that no signal ever exists in the channel diagonally opposite the corner from where the sound is meant to predominate.

In the sketch, the circle sizes depict the relative level; the solid circles represent in-phase signals, while the white circles are out-of-phase signals. Signals that lead in phase are shown with close-spaced lines showing the number of degrees (90 degrees would be a horizontal line). Signals that lag are shown

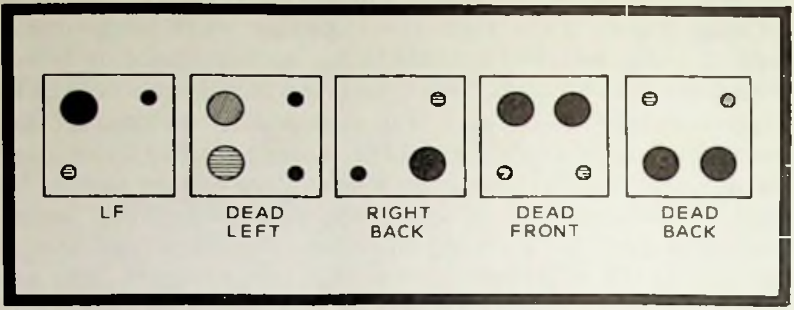


Fig. 6-9. Sansui matrixed quad program is shown here for five room conditions. Until recently, smaller circles represented signals at half-power level of that channel. Sansui's new "second generation" line maintains a channel separation as high as 12 dB, which represents an adjacent channel power difference of 16 to 1.

as wide-spaced lines in the circles. The right front, right, and left rear channels are not shown; these are mirror images of left front, left, and right rear channels, respectively.

COLUMBIA (CBS) SQ SYSTEM OPERATION

CBS is the Big Name in four-channel sound. Since CBS makes records and lots of them (Columbia label), it has a voice that gets heard when it speaks. Through sheer giantism, CBS has managed to let is encoding and decoding systems seep through framework and bulkheads of a large number of firms engaged in the manufacture and sale of stereo equipment.

The most notable characteristic of the CBS matrix approach is the diminished front-to-back separation. CBS apparently feels that left-right separation is of paramount concern, and thus the company deliberately sacrifices the one in favor of the other. The side-to-side separation is good, but the soundness of the compromise remains open to question.

The room diagrams in Fig. 6-10 illustrate the various levels and phase under varying input signal conditions. When the system is intended to reproduce a sound source in the front left corner, for example, there is infinite separation between the two front speakers. The two back channels, however, receive the same signal 3 dB down. Since the two back channels are phased in reverse, the sound from the rear is not well localized, and the listener's attention is drawn toward the dominant front-left speaker by means of the psychoacoustic phenomenon of directivity by phase and volume.

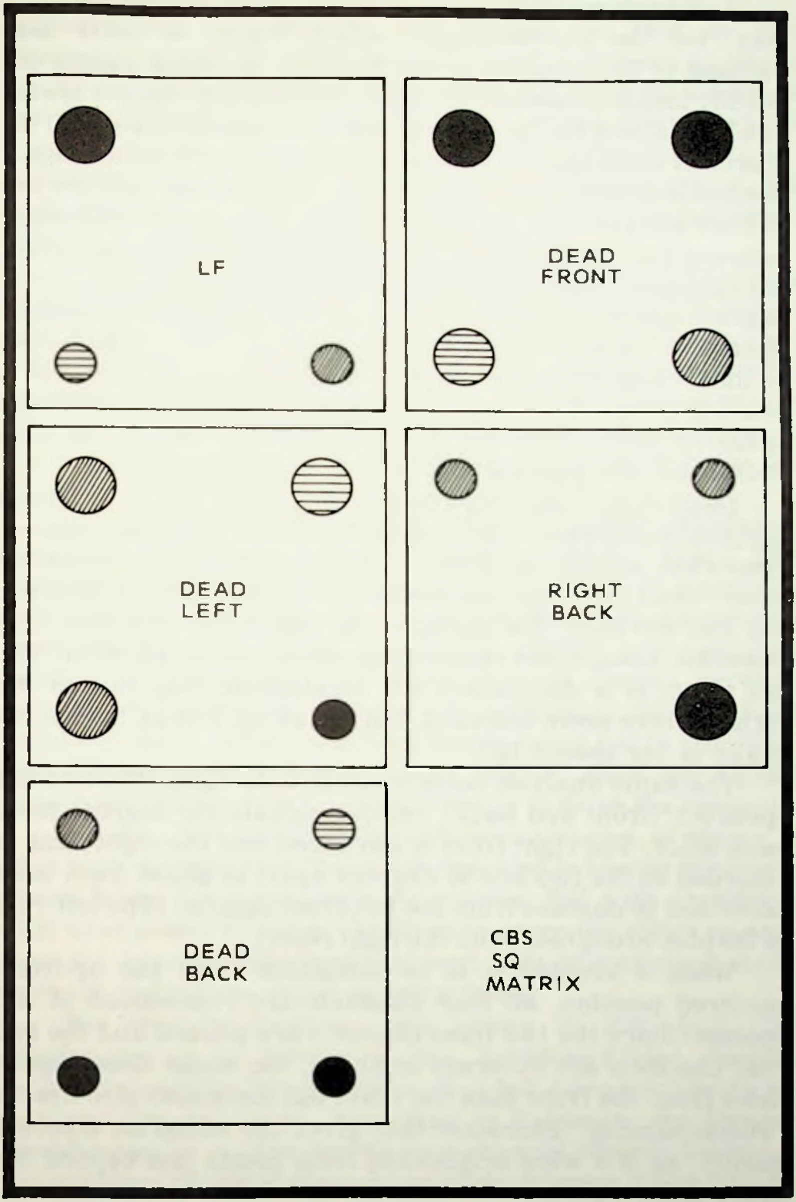


Fig. 6-10. When a corner signal dominates, the mating channel on the other side is down to such a level that separation is infinite. The two channels opposite, however, are down but 3 dB. When phantom signals are placed (front, back, or sides), however, the CBS system delivers full audio volume to all four channels.

Left back and right back channels are processed the same way, but the psychoacoustic effect begins to falter here because of the vagaries of our hearing: In-phase signals are readily localized toward the front; out-of-phase signals are not easily localized by the listener, and they tend to sound as if the source is back there somewhere." When a left-center signal predominates, the two rear channels are phased, and the two in front are reverse-phased, but there is no volume difference between the front and back speakers. In-phase signals from the rear can create a certain "source confusion" because we humans aren't accustomed to listening to phased back signals in nature. The front speakers are reverse-phased, which adds to the confusion for the same reason: We just aren't used to hearing reversed-phase front signals. If the level in the back speakers were increased, or if the front-speaker level were decreased, the psychoacoustic aim would be realized.

Dead-right and dead-left signals are also localized psychoacoustically. CBS employs a phase-shifting concept somewhat similar to Sansui's. When a source is recorded from "dead left," the playback retards the phase of the front left and advances the phase of the left back. The two right channels, though, are reversed in phase from each other. The net effect is a diminished left localization that makes the signal source seem leftward, but not as far left as the source really is (or should be).

The same analysis holds true for dead right. The two left speakers (front and back) receive signals 180 degrees from each other. The right front is advanced and the right back is retarded so the two are 90 degrees apart in phase from each other and 90 degrees from the left-front signals. (The left rear is 180 plus 90 degrees from the right rear.)

When a vocalist is to be simulated from the up-front, centered position, all four channels are reproduced at full volume. Since the two front channels are phased and the two rear channels are reversed in phase, the sound does appear more from the front than the rear; but the sound also has an "encompassing" character that gives the sound an ethereal quality, as if it were originating from points just beyond the upper front wall and the ceiling overhead.

DYNACO

The various encoding-decoding schemes have been subjected to an almost unending series of alterations, as each manufacturer incorporates this feature or that in the ceaseless struggle for rank among the ranks. Dynaco's system began as a purely passive arrangement, whereby the speakers for the rear were simply connected across the front speakers in such a way that difference signals were directed to the rear. A simple resistive element maintained the identities of left and right back channels. The circuit (or rather, its equivalent) was presented in Chapter 2.

The chief drawback with this approach, of course, was that the four channels are always derived, and were dependent upon out-of-phase components existing in ordinary two-channel stereo records. In time, the company incorporated a resistor network to cross-feed the left and right signals so that the left back and right back channels received the difference signals, as before, but they also received a portion of the appropriate up-front channels as well.

The result of cross-feeding, of course, is phase channeling of signals other than 180-degree reverse-phased types. The system works very well, and offers advantages in simplicity and economy that cannot be matched by the other competitive systems. Dynaco's claim to glory, of course, is its selected position of signal splitting. Most systems decode between preamp and amplifier, so there are a couple of decoded back channels that must be amplified before being fed to speakers. But Dynaco splits at the amplifier output, after the signals have been increased in level sufficiently to drive the speakers.

The disadvantage of postamplifier matrixing, of course, is that high-wattage amplifiers are required to handle the power levels being passed through the matrix. But with amplifiers that have power to spare, the Dynaco approach offers an easy way for an audiophile to experiment with four-channel sound without risking too great an investment.

The Dynaco diagrams are shown in Fig. 6-11. The localization is so similar to the Feldman-Fixler decoder originally used by Electro-Voice that full compatibility exists between the two types. According to Feldman, the Dynaco decoder can be used to decode four-channel records matrixed by encoders of either Dynaco or early Electro-Voice design.

The weak points of the Dynaco system show in the diagrams. For corner-front and dead-left and dead-right signals, all is well—the decoding is straightforward and effective. The rear corners and the dead-back position don't fare

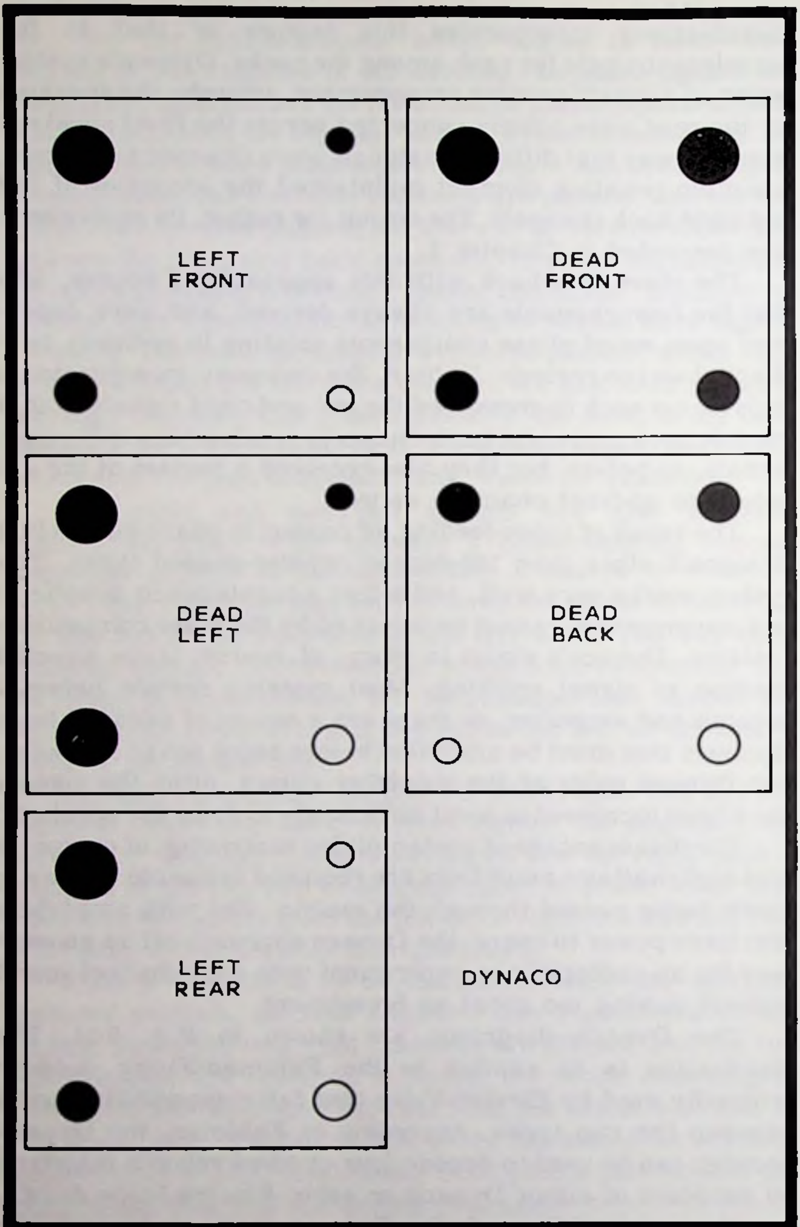


Fig. 6-11. The Dynaco decoder is the least expensive all the way down the line, but this advantage is offset by the performance with rear-dominating signals—all rear signals, whether left back, dead back, or right back, tend to sound alike.

quite so well, however. When a sound source is assigned to a corner, the level of the two rear speakers is virtually identical —certainly close enough so that it is hard to discern the level differences. What this means is that there are no substantial signal differences after decoding if the source is to the rear, because left back, right back, and dead back all sound very much alike.

Chapter 7
Technical Aspects of Matrixing

Writing in the April 1971 edition of the Journal of the Audio Engineering Society (Four Channels and Compatibility), Mr. Peter Scheiber listed a number of basic requirements to be satisfied with any matrixing scheme. The requirements enumerated by Scheiber involved economy and compatibility for the most part, but there were two significant performance parameters as well: the ability to record all sounds occurring at any point within a 360-degree field around the input transducers, and to reproduce each sound from the correct location in playback; and nondegradation of signal quality, including noise, frequency, and nonlinear distortion, as consistent with "highest standards in the state of the art."

Unfortunately, one of the drawbacks of four-channel matrixing in general is the mislocalization of information in its encoding-decoding process. This characteristic is attributable to the existence of out-of-phase sound components that cancel themselves out completely under some conditions.

There are two ways to reproduce four-channel sound by matrixing. One is a technique whereby the phase of some of the signals recorded on a conventional stereo disc is delayed. In stereo playback, the phase delays add a degree of "presence" to the recorded material; in four-channel playback, certain of the signals are directed to the rear speakers in a "synthesizing" scheme. The second method is the actual encoding of four discrete information channels onto the two stereo groove walls, and decoding the information on playback to reconstruct, as nearly as possible, those original discretes.

Crosstalk is an unavoidable consequence of any matrixing system of four-channel stereo, whereby the channels of information are matrixed or encoded into two channels, stored, and then decoded back into four channels. The question then is how to best distribute such crosstalk.

It is mathematically obvious that the maximum separation among the four channels in any matrix system (such as described above) is 3 dB down on two channels and an infinitesimal signal on the final channel. The question is, how best to exploit this mathematical ultimatum. CBS chose to use the "minus infinity" channel adjacent to the dominant one; Sansui assigns it to the diagonally opposite position.

To locate a real sound image correctly, the ideal distribution of the crosstalk is as illustrated in Fig. 7-1. In this arrangement, the 3 dB crosstalk is allowed in the two adjacent channels, X and Y, of the primary channel.

The phantom sound image formed by the crosstalk components will then be located in the primary channel. All this means that a phantom sound image resulting from crosstalk components coincides with a real sound image only if the speakers reproducing the crosstalk components are placed symmetrically on both sides of the subject speaker. The same symmetry is also required for any phantom sound image located between any pair of speakers.

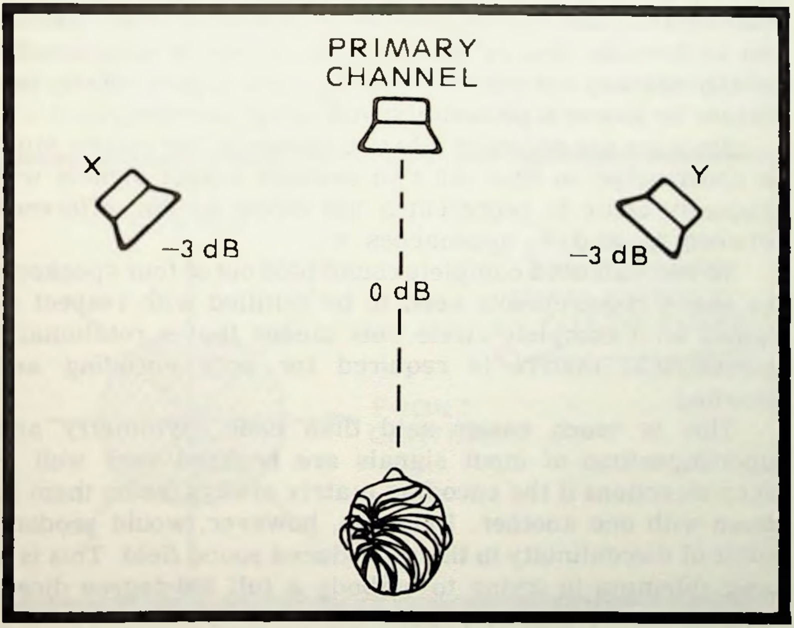


Fig. 7-1. The crosstalk of two −3 dB in-phase signals produces a phantom channel (primary channel) at a point equidistant from the two adjacent channels if the listener is positioned at an equal distance from channels X and Y.

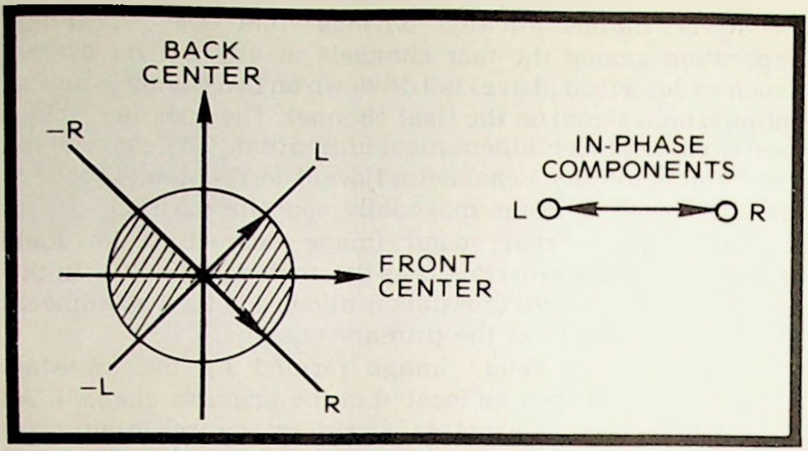


Fig. 7-2. In-phase stereo signals offer strong source localization, but limit the apparent sound field to the space between two speakers.

If identical signals were fed simultaneously to the two input terminals of the encoder which are directional by Φ_1 and Φ_2, respectively, they must be superimposed upon each other inside the encoder, without canceling each other to any extent, and be encoded into an output signal which is directionally exactly halfway between Φ_1 and Φ_2. This is particularly important to locate a phantom sound image correctly.

Since we are encoding into two channels, the matrix must be constructed so that the two encoder output signals will gradually come to represent a full circle as the difference between Φ_1 and Φ_2 approaches π.

To reconstruct a complete sound field out of four speakers, the above requirements need to be fulfilled with respect to signals on a complete circle.This means that a rotationally symmetrical matrix is required for both encoding and decoding.

This is much easier said than done. Symmetry and superimposition of input signals are realized very well in given directions if the encoding matrix always treats them in phase with one another. Doing so, however, would produce points of discontinuity in the reproduced sound field. This is a basic dilemma in trying to embody a full 360-degree directionality in two channels only.

Early stereo records were recorded with only in-phase signals, as shown in Fig. 7-2. This system helped to strongly localize the two channels but it contributed little to the apparent "live" quality of the program. As the sketch shows, there was little vertical stylus movement because both walls of the record groove tracked very closely with one another.

As recording companies began to place more emphasis on character of sound and less on "point-sourcing," they learned to stagger the signal phasing to some extent, as shown in Fig. 7-3. When the two groove walls are recorded at a phase that is not exactly equal, the stylus has a vertical motion as well as horizontal. One groove wall curves in toward the center of the record slightly ahead of the other, which causes a "pinch" near the bottom of the groove, forcing the stylus up between the walls, toward the surface of the disc. When the outer groove wall is cut slightly ahead of the inner wall, a counter-clockwise "spiraling" movement of the stylus between the groove walls results. When the inner wall is advanced, the stylus traces a clockwise vertical-horizontal spiral.

The same basic technique is used for four-channel matrix encoding, except that the phase leads and lags are precisely controlled. Fig. 7-4 shows encoded four-channel cutting using the out-of-phase area in the vector diagram to cut the information which will appear in the back channels. There are inherent weaknesses, particularly when it comes to reproducing sounds in the exact "dead back," but essentially, this concept allows recording of sound sources over a 360-

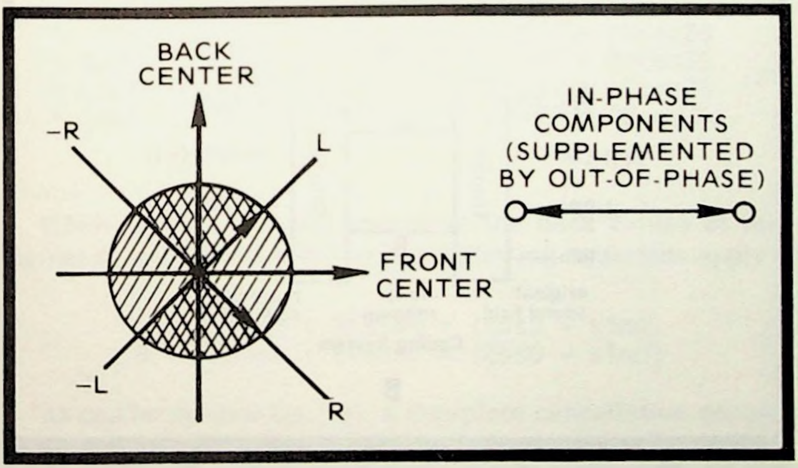


Fig. 7-3. Without sacrificing directionality unduly, some out-of-phase signals can be superimposed on the recording, as shown. This adds "presence" to the program and serves to widen the wall of sound emanating from the two speakers.

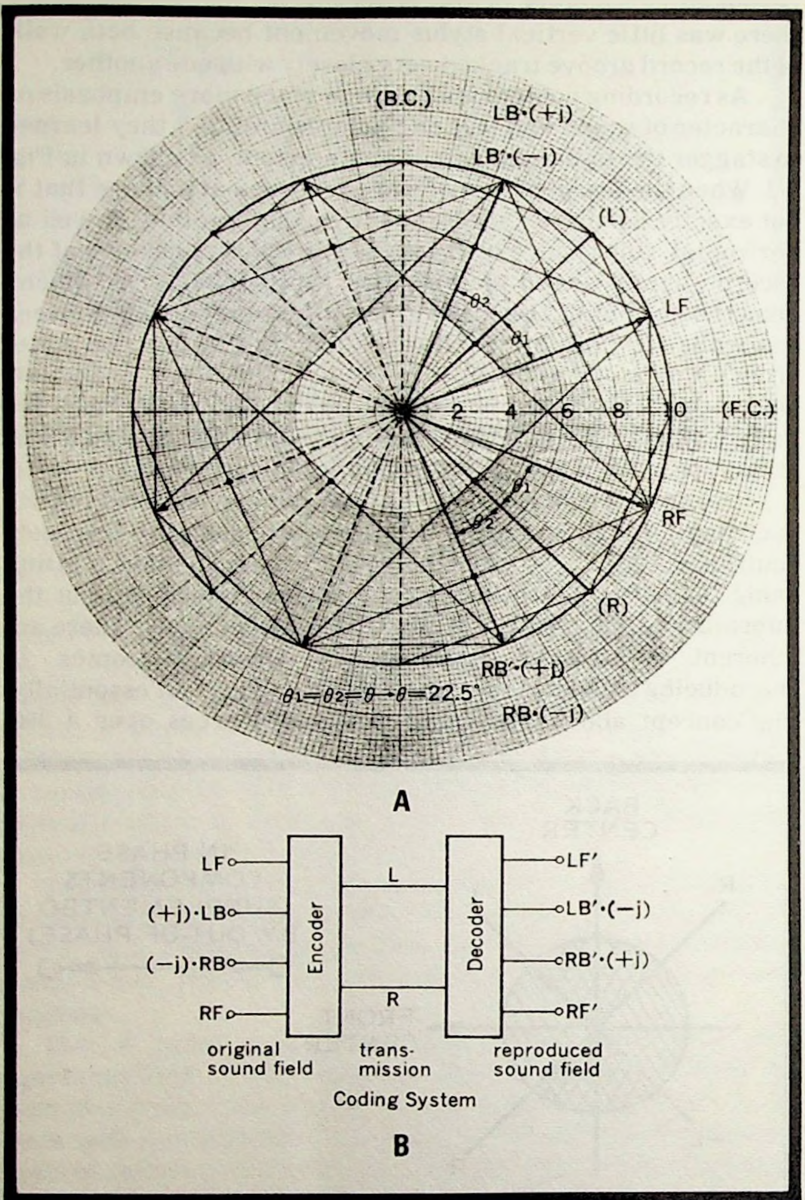


Fig. 7-4. When information is encoded onto a disc, the cutting stylus follows a controlled path horizontally and in a vertical-horizontal spiral. In sketch A, the direction and degree of stylus motion determines the out-of-phase area (rear) into which sound sources can be localized. The coding used for localization is shown in sketch B.

degree arc. Playback, in turn, re-creates the sound "field" experienced within that perimeter.

As noted previously, all basic encoding-decoding schemes are similar, though there is not necessarily a high degree of compatibility. The vector diagrams shown in Fig. 7-5 show the four basic schemes and the mathematical formulas for deriving the various conditions. The "mislocalization" of information due to out-of-phase conditions can be explained using the equation given in the figure.

The explanation for phase misplacement will be based on information supplied by Sansui. Taking the Sansui equation as our example, the encoder output will be

$$\begin{aligned} L &= (LF + LB)\cos\Theta + (RF - RB)\sin\Theta \\ R &= (RF + RB)\cos\Theta + (LF - LB)\sin\Theta \end{aligned} \qquad (1)$$

The terms LF and RF refer to the left front and right front, of course, and LB and RB refer to left back and right back. A "prime" symbol will be used to denote decoded signals. Encoded phase angles are ϕ and decoded phase angles are represented by Θ.

With encoding as given in Eq. (1), the relative decider output will be

$$\begin{aligned} LF' &= L\cos\Theta + R\sin\Theta = LF + 2RF\sin\Theta\cos\Theta + LB\cos 2\Theta \\ RF' &= R\cos\Theta + L\sin\Theta = RF + 2LF\sin\Theta\cos\Theta + RB\cos 2\Theta \\ LB' &= L\cos\Theta - R\sin\Theta = LB - 2RB\sin\Theta\cos\Theta + LF\cos 2\Theta \\ RB' &= R\cos\Theta - L\sin\Theta = RB - 2LB\sin\Theta\cos\Theta + RF\cos 2\Theta \end{aligned} \qquad (2)$$

When there is a sound source at the back center of the original sound field (LB=RB=1), the following equations apply:

$$\begin{aligned} L &= LF\cos\Theta + RF\sin\Theta + (\cos\Theta - \sin\Theta) \\ R &= RF\cos\Theta + LF\sin\Theta + (\cos\Theta - \sin\Theta) \end{aligned} \qquad (3)$$

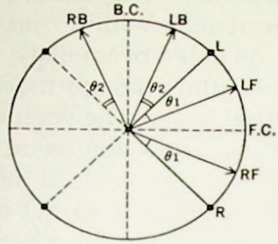
As can be seen in Eq. (3), a complete cancellation occurs in the out-of-phase components of the back channels, so that all resultant left and right channels of the encoder output are composed entirely of in-phase sounds. Thus, there does occur a loss of information (and, of course, mislocalization of sound) that is irretrievable on playback. The same premise indicates

CBS-SQ

$$\begin{cases} L=(LF+LB)\cos\theta+(RF+RB)\sin\theta \\ R=(RF-RB)\cos\theta+(LF-LB)\sin\theta \end{cases}$$

when $LF=RF=RB=LB\ (=1)$;

$$\begin{cases} L=2\cos\theta+2\sin\theta=2.60 \\ R=0 \end{cases}$$

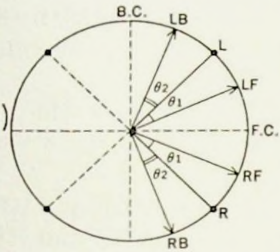


SANSUI-QS

$$\left.\begin{cases} L=(LF+LB)\cos\theta+(RF-RB)\sin\theta \\ R=(RF+RB)\cos\theta+(LF-LB)\sin\theta \end{cases}\right) \quad (1)$$

when $LF=RF=RB=LB\ (=1)$;

$$\begin{cases} L=2\cos\theta=1.84 \\ R=2\cos\theta=1.84 \end{cases}$$

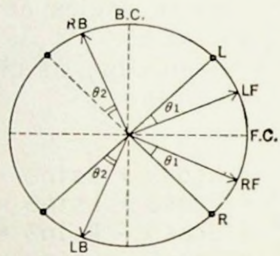


DYNACO

$$\begin{cases} L=(LF-LB)\cos\theta+(RF+RB)\sin\theta \\ R=(RF-RB)\cos\theta+(LF+LB)\sin\theta \end{cases}$$

when $LF=RF=RB=LB\ (=1)$;

$$\begin{cases} L=2\sin\theta=0.76 \\ R=2\sin\theta=0.76 \end{cases}$$

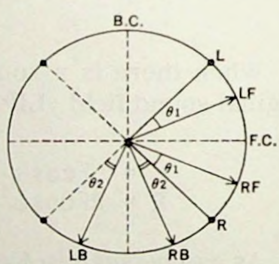


ELECTRO-VOICE

$$\begin{cases} L=(LF-LB)\cos\theta+(RF-RB)\sin\theta \\ R=(RF+RB)\cos\theta+(LF+LB)\sin\theta \end{cases}$$

when $LF=RF=RB=LB\ (=1)$;

$$\begin{cases} L=0 \\ R=2\cos\theta+2\sin\theta=2.60 \end{cases}$$



ALL DIAGRAMS REPRESENT the vector angle $\theta_1=\theta_2=\frac{\pi}{8}$

F.C.=Front Center
B.C.=Back Center

Fig. 7-5. Vector diagrams of basic matrixing techniques at a vector angle of $\Theta_1=\Theta_2=\pi/8$ equation.

that it is virtually impossible to encode four full-volume channels of identical phase simultaneously.

Since the left back and right back are reversed in phase, as shown by Eq. (2), sound sources located in the rear in a four-channel program can seem indefinite in origin. This was shown in the room diagrams of Chapter 6.

Until a recent Sansui innovation of automatically varying the matrix, back localization of sounds has been a major problem with matrixing schemes, and a great deal of the information derived from the rear channels in an actual program depended heavily on the "psychoacoustic" interpretation of the listener.

Both Sansui and CBS utilize a spurious 90-degree phase shift of the back channels to enhance the apparent spatial separation (adopted in favor of a 180-degree phase inversion). This technique puts the four encoded channels in an ideal phase relation, as illustrated in Fig. 7-6A, and prevents cancellation of 180-degree signals at the encoder.

On decoding, the back channels are reverted 90 degrees to their initial positions (Fig. 7-6B). What was out of phase in the encoding process is now in phase . In terms of vector angles in the disc's stereo groove, the encoder outputs are

$$\begin{aligned} L &= (LF + jLB)\cos\Theta + (RF + jRB)\sin\Theta \\ R &= (RF - jRB)\cos\Theta + (LF + jLB)\sin\Theta \end{aligned} \tag{4}$$

The mathematics of Eq. (3) show that dead-back information (out-of-phase signals) that are lost with 180-degree phase reversals are saved by the 90-degree phase shifting technique (j-phase).

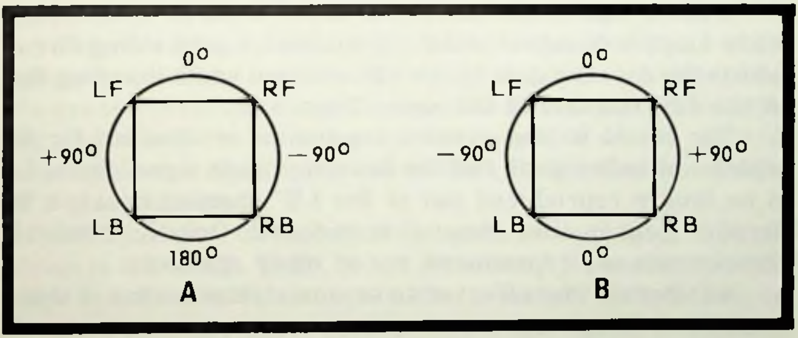


Fig. 7-6. Phase relationships between four channels. Sketch A shows the encoder; sketch B shows the decoder.

The fact remains that there is a certain unavoidable limitation on the information we can extract from the two stereo storage channels. There are two apparent courses of action to escape the seemingly inevitable fate of no more than 3 dB down on two channels with infinite separation on the last: Give some psychoacoustic treatment to the two encoded channels, or exploit the redundancy in the two stereo channels.

The CBS SQ matrixing concept places paramount emphasis on left-right separation, and it accomplishes this by placing the "infinity separation" channel adjacent to the dominant channel. The rotational symmetry discussed earlier, however, is sacrificed by this technique, and the result is degraded front-to-back separation.

As indicated in the room diagrams of the previous chapter, the audio volume level of all four channels is the same with the CBS-SQ decoder, regardless of whether the sound originates from the back center or the front center. CBS relies on the psychoacoustic properties of the human hearing mechanism to achieve front-to-back separation in this case; that is, when a vocalist (up front, centered) is to be simulated, the two front speakers receive in-phase signals and the back speakers receive out-of-phase signals of equal amplitude. The reverse is true when back-center sounds are reproduced.

Some decoders use special "gain-riding" logic circuits to keep the dominant channel dominant. This approach gives an apparent increase in separation by virtue of the decoder's relative channel gain. The chief problem with this method is the sacrifice of symmetry, which results in a distortion of the true sound field the matrix is attempting to re-create.

When a high-amplitude signal is fed to the LF channel and a low-amplitude signal to the LB channel, a gain-riding circuit boosts the decoder gain in the LF' channel while lowering that in the LB' channel at the same time.

The result is that greater separation is obtained for the high-amplitude signal, but the low-amplitude signal fed to LB is no longer reproduced out of the LB' channel because the decoder gain in that channel is reduced. Only its crosstalk components are reproduced out of other speakers.

Another serious effect of an asymmetrical matrix is that a real sound image and its phantoms resulting from crosstalk are located at different positions. The extent of this deviation increases as the logic circuit goes to work, producing a series of complicated displacements of sound images in the sound field.

Sansui has developed an interesting technological twist to skirt the law of nature. The concept involves the use of a system to control output signals symmetrically by varying the matrix itself. It is an amplitude matrix with an element of phase-matrixing added, and is possible only because the basic matrix is rotationally symmetrical in nature. (Dynaco and Electro-Voice use elongated matrixes, CBS employs a widened matrix; discrete is symmetrical, of course.)

VARIABLE MATRIX

A sound source in the direction Φ is encoded by a symmetrical encoding matrix into

$$\begin{aligned} L &= E\Phi \sin \frac{\Phi}{2} \\ R &= E\Phi \cos \frac{\Phi}{2} \end{aligned} \qquad (5)$$

On the other hand, signal $E\Theta$ which is decoded by the decoding matrix in the Θ direction is given by

$$\begin{aligned} E'\Phi &= L \sin \frac{\Theta}{2} + R \cos \frac{\Theta}{2} \\ &= E\Phi \sin \frac{\Phi}{2} \sin \frac{\Theta}{2} + E\Phi \cos \frac{\Phi}{2} \cos \frac{\Theta}{2} \\ &= E\Phi \cos \frac{\Phi - \Theta}{2} \end{aligned} \qquad (6)$$

Thus the angular difference between the direction Φ of the encoded signal and a given direction Φ in which it is decoded is always symmetrical along the encoded direction. For example, Fig. 7-7 shows the sound pressure response of the decoder outputs when a signal encoded in the LF direction is decoded in a given direction.

Under these circumstances, assume a high-amplitude signal is fed to LF. Then, as we alter the LB' decoder matrix angle as indicated by the arrow, the crosstalk of the LF signal contained in the LB' decoder output gradually decreases. When the LB' matrix angle finally coincides with the RB' matrix angle, the separation between LF' and LB' becomes

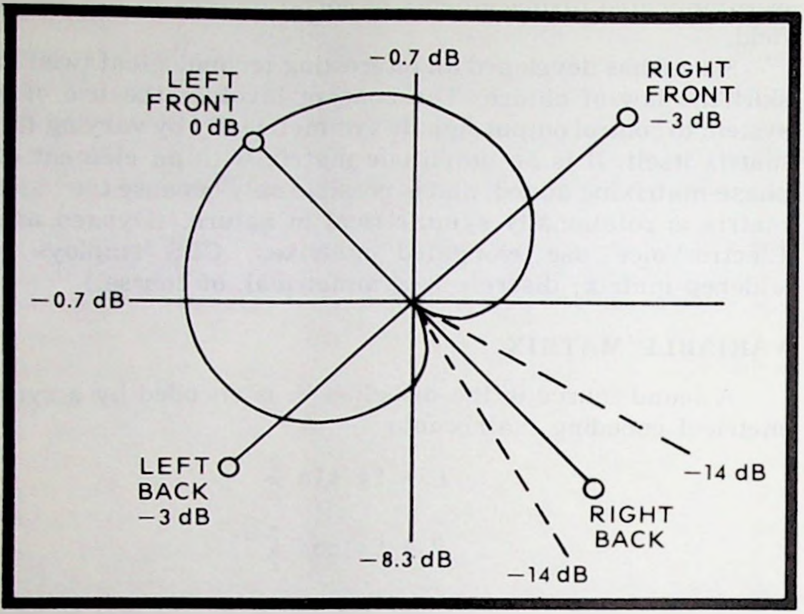


Fig. 7-7. Output sound pressure response of decoder when signal is fed to left front. The half-channel points adjacent to the dominant channel are 0.707 dB down, and the adjacent channels are 3 dB down.

infinitely large -∞ dB). Even then, whatever signal exists in LB is only attenuated by 3 dB.

Therefore, if we boost the LB' decoder gain by 3 dB simultaneously as we shift the LB' matrix angle to the RB' position, the LB signal of the original level, free of any crosstalk of the LF signal, will be delivered at the LB' output terminals.

The same holds for RF if we shift the RF' decoder matrix angle toward RB'.

The variation of the matrix as described above is performed without losing its symmetrical property. For example, the crosstalk components of the LF signals in the two adjacent channels decrease in equal proportions— which means that the phantom image produced by the crosstalk components continues to coincide with the real image as it decreases in amplitude. That being the case, the variation of the matrix as previously described does not displace sound images whatsoever. Nine representative variations of the matrix are shown in Fig. 7-8. It goes without saying that such variations can be made of signals in all 360 degrees.

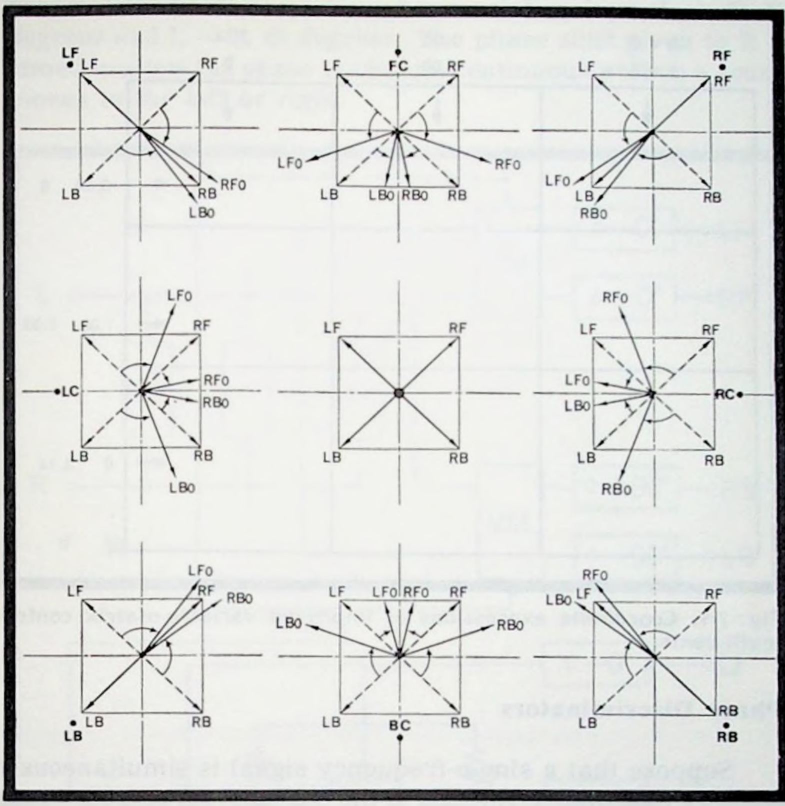


Fig. 7-8. The nine plots here are representative variations of Sansui's improved decoding matrix. Abbreviations LF, RF, LB, FC, RB, and BC stand for left front, right front, left back, front center, right back, and back center, respectively.

Control Signals

The signals to control the variable matrix are produced by phase discriminators by detecting the front-back and left-right distribution of the input signal, as shown in Fig. 7-9.

Here, the front coefficient f, the back coefficient b, the left coefficient l and the right coefficient r vary their values between 0 and 3.14. When a signal is uniformly distributed in all four directions, each coefficient assumes the value of 1.00.

The reason why the control reference value is not set at the middle of 0 and 3.14 is that the matrix shows an optimum variation for phantom sound images when it is exactly at 1.00.

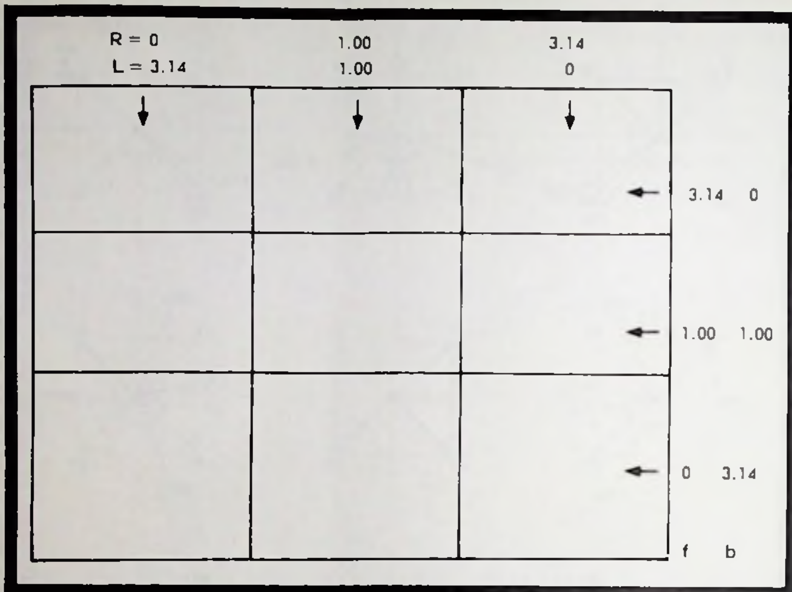


Fig. 7-9. Coordinate expressions of improved variable-matrix control coefficients.

Phase Discriminators

Suppose that a single-frequency signal is simultaneously fed to all four input terminals of the encoder. Its outputs would then be given by

$$\begin{aligned} L&=(LF+\nabla RF)\sin pt+(LB+\nabla RB)\cos pt \\ R&=(RF+\nabla LF)\sin pt-(RB+\nabla LB)\cos pt \end{aligned} \qquad (7)$$

To put them differently

$$\begin{aligned} L&=\sqrt{(LF+\nabla RF)^2+(LB+\nabla RB)^2}\sin(pt+\Theta_1) \\ R&=\sqrt{(RF+\nabla LF)^2+(RB+\nabla LB)^2}\sin(pt+\Theta_2) \\ \Theta_1&=\tan^{-1}\frac{LB+\nabla RB}{LF+\nabla RF} \qquad \Theta_2=-\tan^{-1}\frac{RB+\nabla LB}{RF+\nabla LF} \end{aligned} \qquad (8)$$

Hence, the phase Φ between L and R is given by

$$\begin{aligned} \Theta&=\Theta_1-\Theta_2 \\ &=\tan^{-1}\frac{LB+\nabla RB}{LF+\nabla RF}+\tan^{-1}\frac{RB+\nabla LB}{RF+\nabla LF} \end{aligned} \qquad (9)$$

Accordingly, the front-back proportion is transmitted by phase 0. The left-right proportion, on the other hand, is detected in the form of two phase relationships, L + R 45 degrees and L — R 45 degrees. The phase shift given to R is aimed to allow the phase to change continuously when a sound moves to the left or right.

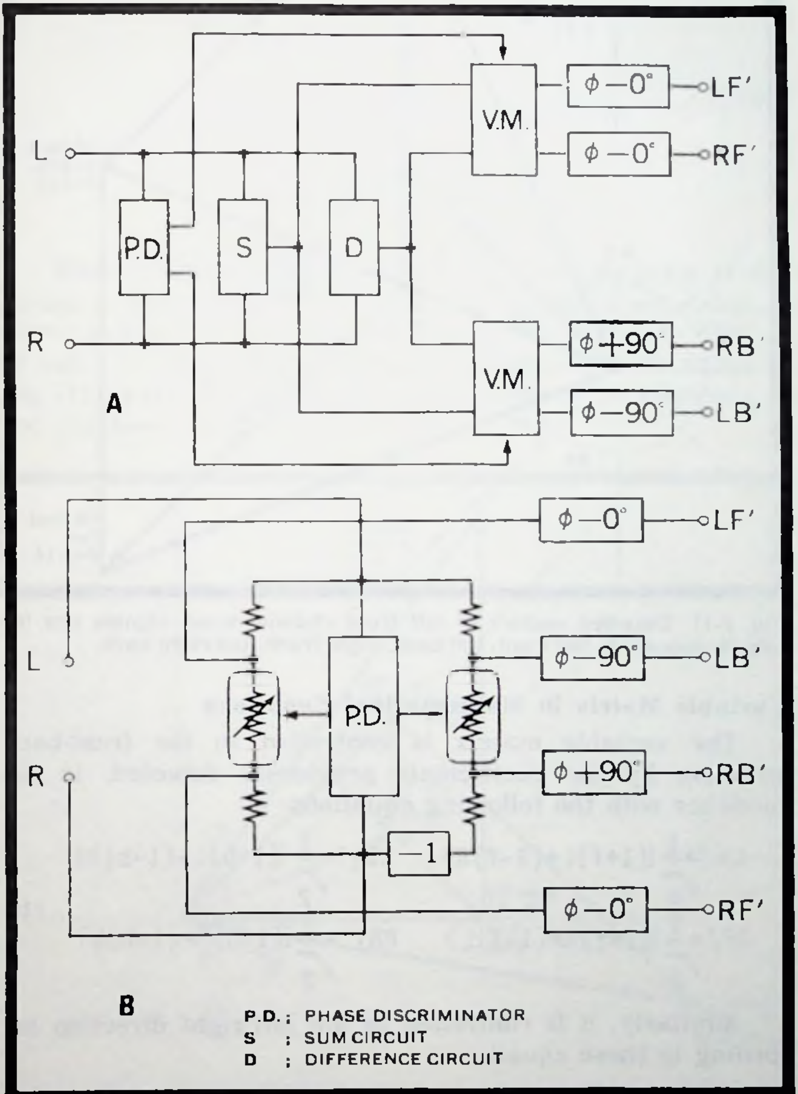


Fig. 7-10. Block diagrams of the variable-matrix four-channel decoder. Sketch A shows component layout; sketch B shows the functional operation of the circuit.

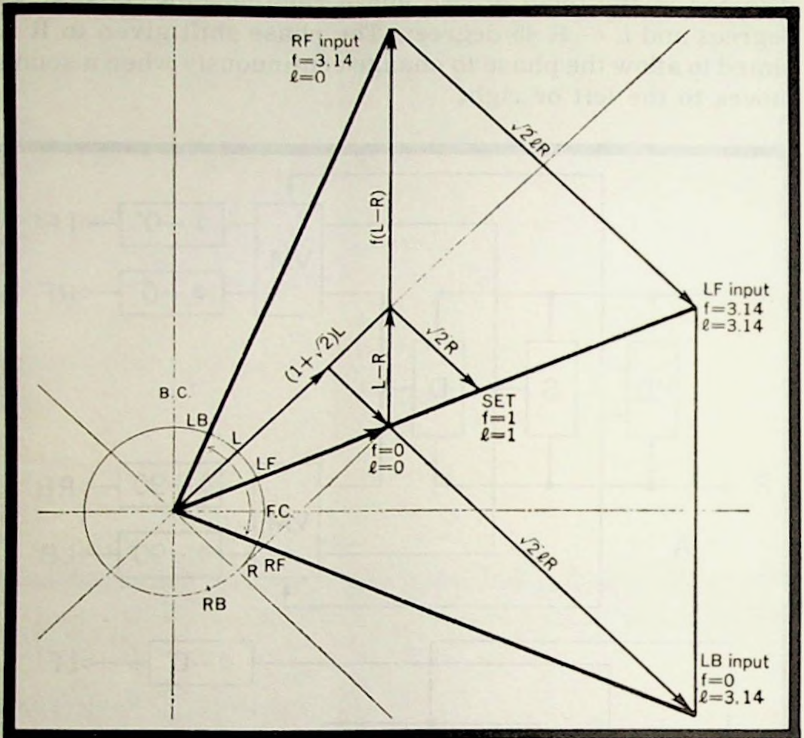


Fig. 7-11. Decoded vectors of left front channel when signals are fed simultaneously to left front, left back, right front, and right back.

Variable Matrix in Mathematical Equations

The variable matrix is controlled in the front-back direction by the coefficients previously detected, in accordance with the following equations:

$$LF_1' = \frac{1}{\sqrt{2}}\{(1+f)L+(1-f)R\} \qquad LB_1' = \frac{1}{\sqrt{2}}\{(1+b)L-(1-b)R\}$$
$$RF_1' = \frac{1}{\sqrt{2}}\{(1+f)R+(1-f)L\} \qquad RB_1' = \frac{1}{\sqrt{2}}\{(1+b)R-(1-b)L\} \tag{10}$$

Similarly, it is controlled in the left-right direction according to these equations:

$$LF_2' = L + 1R \qquad RF_2' = R + rL$$
$$LB_2' = L - 1R \qquad RB_2' = R - rL \tag{11}$$

From the above two conditions, the matrix varies according to the following equations:

$$LF'=\frac{1}{\sqrt{2}}\{(1+f+\sqrt{2})L+(1-f+\sqrt{2}l)R\}$$

$$RF'=\frac{1}{\sqrt{2}}\{(1+f+\sqrt{2})R+(1-f+\sqrt{2}r)L\}$$

$$LB'=\frac{1}{\sqrt{2}}\{(1+b+\sqrt{2})L-(1-b+\sqrt{2}l)R\} \tag{12}$$

$$RB'=\frac{1}{\sqrt{2}}\{(1+b+\sqrt{2})R-(1-b+\sqrt{2}r)L\}$$

These equations satisfy Fig. 7-8. A block diagram of the Sansui improved matrix, based on the equations presented, is shown in Fig. 7-10. The decoded vectors are plotted in Figs. 7-11 and 7-12. The vectors shown are graphic presentations of Eq. (12); each demonstrates the directions of the vibration of the playback stylus in a disc groove.

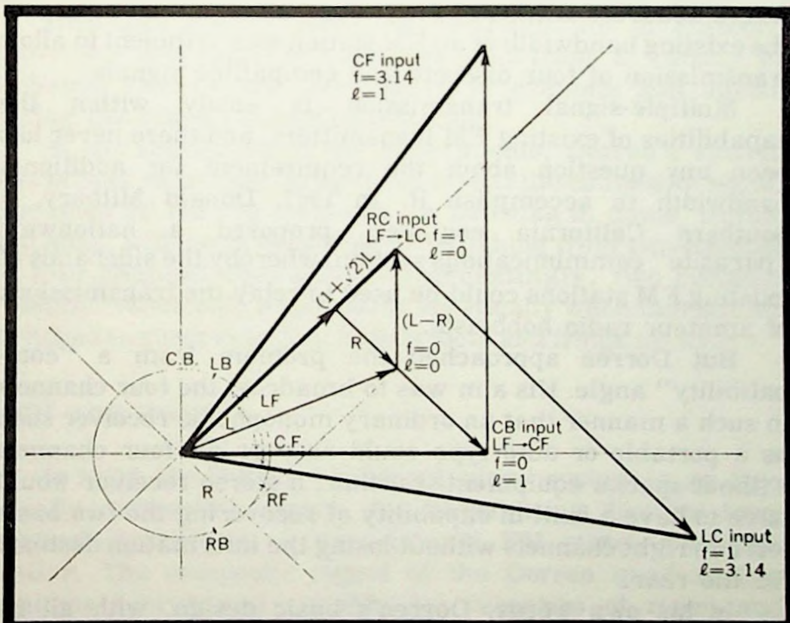


Fig.7-12. Decoded vectors of left front channel when signals are fed simultaneously to indicate front center, back center, left center, and right center.

Chapter 8
FM Quad Broadcasting

There's no big trick to broadcasting four-channel FM signals if the broadcast station limits itself to the matrix format. For live programs the requirement is simply an appropriate encoder at the transmitting station. For recorded music over a stereo multiplex station it's even simpler: just play stereo records that have been previously encoded by one of the existing matrixing methods.

But if the discrete approach to four-channel sound is ever to gain widespread favor, broadcast FM must be capable of providing a source for the four individual signals. This is where Lou Dorren entered the picture. Dorren contended that the existing bandwidth of an FM station was sufficient to allow transmission of four discrete and compatible signals.

Multiple-signal transmission is easily within the capabilities of existing FM transmitters, and there never has been any question about the requirement for additional bandwidth to accomplish it. In 1967, Donald Milbury, a Southern California engineer, proposed a nationwide "parasite" communications system, whereby the sidebands of existing FM stations could be used to relay the transmissions of amateur radio hobbyists.

But Dorren approached the problem from a "compatibility" angle. His aim was to broadcast the four channels in such a manner that an ordinary monophonic receiver such as a portable or auto type could recover all four channels without special equipment. Further, a stereo receiver would have to have a built-in capability of recovering the two basic left and right channels without losing the information destined for the rear.

In his own words, Dorren's basic design, with all its "limitations, extremes, and engineering parameters, was set down on paper as a total system—rough, ragged, and admittedly imperfect at first." But the paperwork within a matter of days was converted to hardware, which consisted of breadboard circuits ready for interconnection at a cooperating FM station.

James Gabbert, who heads Pacific FM, Inc., decided that four-channel experimentation would be a good business move for his FM station (KIOI-FM) from just about every angle. There would be plenty of publicity if his station was the first to go discrete, and the station would get the reputation of progressiveness. Gabbert filed for FCC permission to start broadcasting with Dorren's "quadraplex" apparatus. Within a surprisingly short time, the FCC granted the STA (station temporary authorization), and the tests were begun.

KIOI-FM is a 150,000-watt station in the San Francisco Bay area that enjoys the largest FM audience in its range. The station, under FCC observations, was to broadcast in discrete quad at specified times during a two-month test period beginning in January 1971. The station's management would compile "audience reaction" reports while the technical staff was to make "exhaustive tests" of performance at various locations throughout the transmitter's operational range.

Help was drafted from the computer lab at Stanford University for computer analysis and from a variety of technical experts who were asked to evaluate and advise during the course of experimentation.

At the conclusion of the experiments, Dorren's system was proved to be fully compatible in that the composite signal contained all the requirements for listening in mono, stereo, and quad, with no degradation in switching from two-channel to four-channel operation. (Interestingly, two-channel reception reportedly improved dramatically when that station switched to quadracasting, both in local and fringe areas.)

HOW IT'S DONE

In brief, the Dorren "quadraplexing" system is an electrically compatible system for transmitting discrete four-channel stereo over a conventional FM stereo multiplex station. The composite signal of the Dorren quad system contains components suitable for reception of mono and stereo, as noted above, while distributing the four-channel information so as to provide fully compatible operation in all of the three modes. Of course, what was of prime concern to the FCC was the question of bandwidth, and the Dorren system did prove capable of being transmitted within the imposed channel allocations.

In present two-channel stereo broadcasting, there are certain frequency assignments in the baseband spectrum of the FM transmitter for the various components of the two-channel transmission. These components are called out as channels which are labeled "main channel" and "sub-channel." To be more specific, the main channel is the L + R channel and the subchannel is the L−R channel. With this system the information theory is fulfilled in that two linear equations are transmitted which are the algebraic sums and differences of the two input signals.

As in the two-channel system, the Dorren quadraplex system has the independent frequency assignments in the baseband spectrum. These assignments differ from two-channel stereo transmission in that there is more information distributed in the baseband due to transmission of four independent information channels. The labels on the subchannel now become different and are named for the information that is occurring in these subchannels during four-channel transmission. The main channel now contains LF+LB+RF+RB, which is the sum of all four of the audio information channels and is the channel that is utilized by the monaural receiver for compatible reception. The subchannel is now named the quadrature subchannel. It is so named because there are two separate and distinct carriers present in this subchannel and they are in what is called quadrature modulation. The first of these carriers is modulated with the signal LF+LB−RF−RB and the second in modulated with the signal for LF−LB−RF+RB. As in two-channel stereo, this subchannel is centered at 38 kHz and both carriers are suppressed. The one last component required for transmission of discrete four-channel stereo is located at 76 kHz and is in what has been named the "quadraphonic" subchannel. This subchannel contains a carrier that is modulated by the signal LF−LB+RF−RB. As with the quadrature subchannel, the carrier in the second subchannel is likewise suppressed.

The other component present in both the standard two-channel system and in the "quadraplex" (coined by discrete-broadcast proponents) system is the 19 kHz pilot carrier. This is used for the same purpose in both systems and that is to synchronize the receiving decoders. Figure 8-1 shows the baseband frequency spectrum for the quadraplex composite signal.

Special conversion of the stereo receiver is not required to receive the quadraplex signals. The only change that need be made to the receiver is to add an output to the discriminator of the i-f amplifier. With this addition to the receiver the quadraplex decoder may be added to the high fidelity system. The only other equipment necessary is a second stereo amplifier and two additional loudspeakers.

Block diagrams of the transmission and reception systems are shown in Fig. 8-2. It is interesting to note that the outputs of the two-channel stereo receiver when receiving a quad broadcast will be the sum of left front and left rear appearing in left channel output and the right front and the right rear appearing in the right channel.

RECEIVER REQUIREMENTS

Just as matrix four-channel decoding requires circuits at the receiver, so does discrete quadracasting. The demodulation circuitry is inserted in the receiver between the discriminator and amplifier stages. Some receivers are currently being manufactured with discriminator output jacks

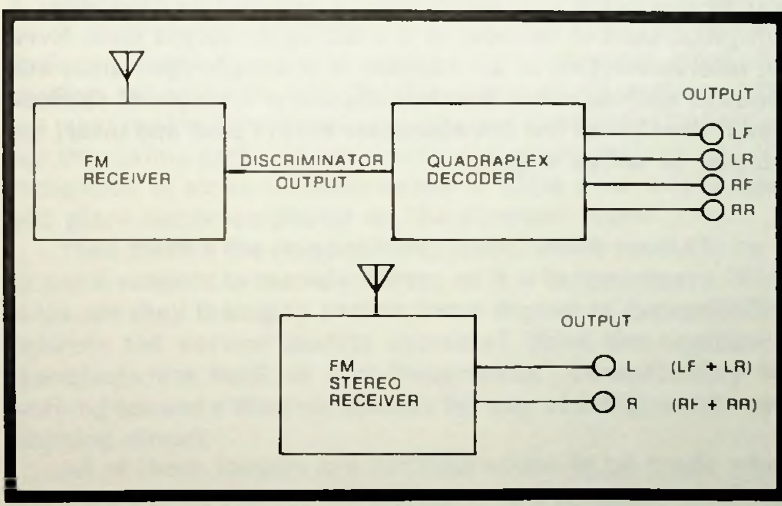


Fig. 8-1. A block diagram of the quadraplexing process is shown here with modulation level and block frequency requirements.

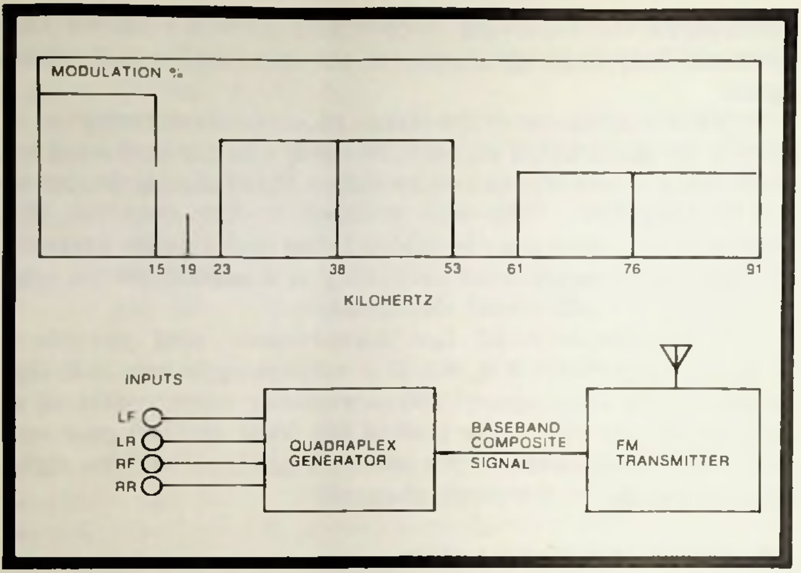
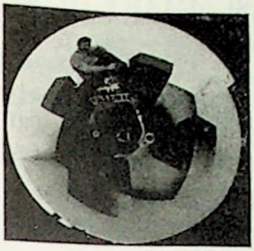


Fig. 8-2. The quadraplex (four-channel) signal is fully compatible with ordinary stereo receivers. As shown, the stereo receiver's left channel carries the entire "left" equation and the right channel carries the other.

so that an add-on demodulator can be incorporated as an adapter. Presumably, if the FCC grants permanent authorization for quadraplexing, all component-type receivers will be equipped with some form of compatibility—either a complete, built-in decoder or a quadraplex output jack. Even so, interconnection of an adapter is a simple operation, and requires nothing more than installation of a couple of shielded leads: Just break the discriminator output lead and insert the adapter in series with it.

Chapter 9
Selecting Your Quad System



There is a great deal more to getting set up with a quad system then simply paying a visit to a stereo store with a lot of money in your pocket. What system will you select? Electro-Voice, so that you will have a high degree of universality? CBS-SQ, since there are so many manufacturers producing equipment with this matrix and records from Columbia are unlikely to be scarce? Sansui, so you can make an attempt at bridging the gap between the matrix and the discrete system? Dynaco, so you can get your feet moist without immersing your whole pocketbook?

Will you settle for nothing less than a 100 percent discrete approach, or will you go for a compatible matrix system and try to incorporate discrete as time goes by?

The multiplicity of approaches to quad—and the tendency for manufacturers to make covert changes in their system designs—would be reason enough for learning what the relative advantages and disadvantages of all systems are; but what makes the educational process even more important are the claims and counter claims of the competitors and the inclination of some manufacturers to gloss over weaknesses and place undue emphasis on the stronger points.

Then there's the compatibility issue, which seems to be of as much concern to manufacturers as it is to consumers. What steps are they taking to assure some degree of compatibility between the various matrix systems? Have the equipment manufacturers built in any "upgrading" compatibility for working toward a discrete system by way of using matrix as a stepping stone?

All of these factors are considerations to be made when selecting a quad system. This chapter is a summary of design capabilities and an overview of areas of compatibility.

SUMMARY OF QUAD APPROACHES

There are two basic approaches to quadraphonic sound reproduction: matrix and discrete. The popularized matrix systems are Sansui, Electro-Voice, Dynaco, and CBS. Occasionally manufacturers do not mention the name of the company whose system they provide equipment for, but almost always they will refer to the matrix type by its own tradename. The SQ approach is Columbia Broadcasting System's. The QS approach is Sansui's. (Sansui's is also frequently referred to as the "regular" system, particularly in foreign-produced equipment.) Electro-Voice's early system is called "Stereo-4"; the later (universal) model is called "Phase II." There is no tradename to watch for with the Dynaco approach, but the key work is "passive"—there are several manufacturers producing passive quad adapters, all of which are capable of decoding records encoded with Dynaco and Electro-Voice matrices.

The big names in discrete are JVC, RCA, and Panasonic, but there are matrix decoder amplifiers that have compatibility with the basic discrete approach.

CBS and Electro-Voice have made an agreement to exchange patent rights of their systems, which was the basis for Electro-Voice's universal decoder. The "arrangement" represents a major step in achieving compatiblility in the four-channel field. According to an Electro-Voice newsletter, released soon after the CBS - E-V announcement, the move is "responsive to the growing industry feeling that the resolution of the compatibility issue is the single most important need for the full potential of quadraphonic sound to be realized."

There is probably more to the solidizing of matrix proponents than meets the eye. More than likely, the matrix people have begun to feel the pinch of RCA, with its admittedly superior discrete disc—and the "togetherness" move is a reaction that might be interpreted as a fear that the latent disorganization of matrix designers might be ammunition for the discrete advocates.

The strides recently made by RCA, JVC, Quadracast Systems, and Panasonic in the field of discrete four-channel reproduction have probably done more to align the matrix people than any other contributing factor.

When shopping for equipment, here are a few points to remember:

With CBS (SQ), the aim has been—and is now—to optimize the left-right separation, even to the extent of neglecting the front-to-back relationship. The rear channels do tend to enhance the illusion of expanse, regardless of the seemingly inequitable distribution of separation, however—and the effect is roughly that pictured in Fig. 9-1.

Electro-Voice's universal decoding system maintains a good up-front separation, but the separation across the two rear channels is compromised in favor of keeping the front well away from the back, as pictured in Fig. 9-2. The advantage of the Electro-Voice system, however, is that it does have an intrinsic compatibility with other matrix formats.

The Dynaco approach is not shown pictorially because it is quite similar in geometry to the Electro-Voice system, the

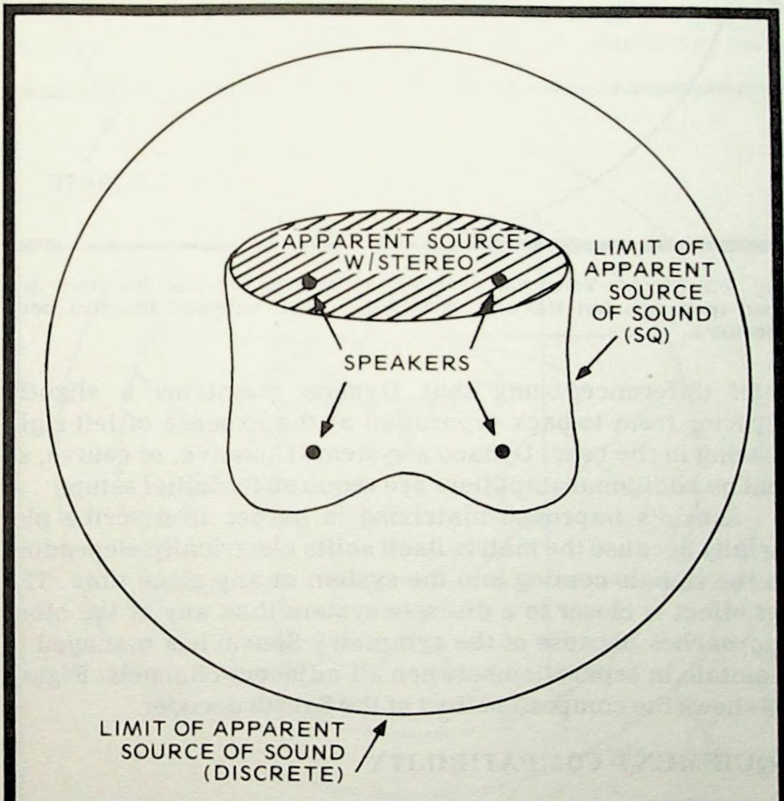


Fig. 9-1. The CBS system has very good left-right separation, but little emphasis has been placed on front-to-back performance.

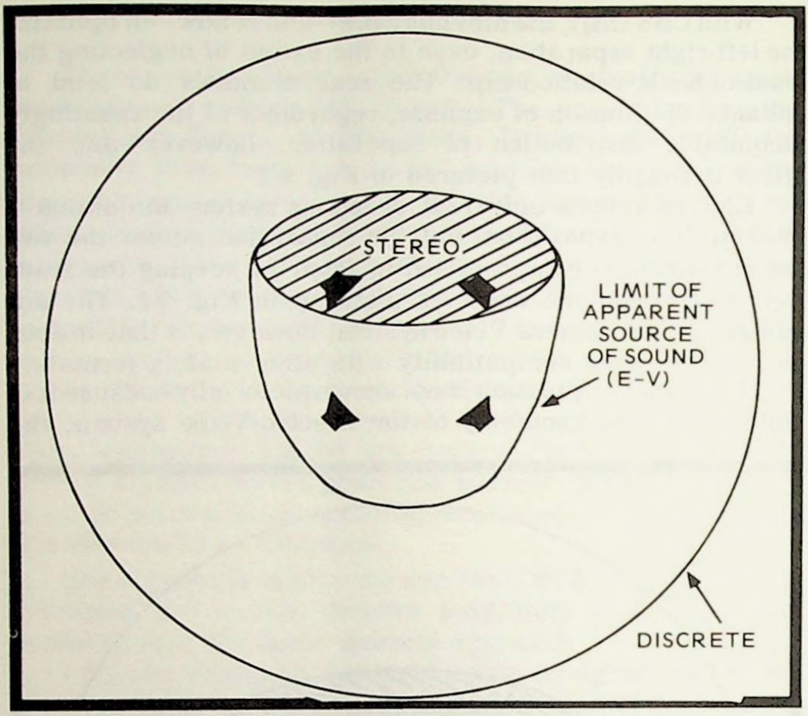


Fig. 9-2. Electro-Voice has excellent separation between the front and rear speakers, but there is practically none between the two back speakers.

chief difference being that Dynaco maintains a slightly superior front-to-back separation at the expense of left-right spacing in the rear. Dynaco's system is passive, of course, so that no additional amplifiers are required for initial setup.

Sansui's improved matrixing is harder to describe pictorially because the matrix itself shifts electrically, depending on the signals coming into the system at any given time. The net effect is closer to a discrete system than any of the other approaches because of the symmetry Sansui has managed to maintain in separation between all adjacent channels. Figure 9-3 shows the composite effect of the Sansui decoder.

EQUIPMENT COMPATIBILITY

There is an intrinsic compatibility of any multiple amplifier system with discrete, regardless of whether or not the system contains a matrix decoder. But not all quad decoders have amplifiers. When you examine the listings, bear in mind that all units that incorporate amplifiers are discrete-compatible, even though this point is not mentioned in the descriptions.

Another point to consider is that not all Dynaco-type decoders are passive. The phasing technique used in the Dynaco system can be implemented at the preamps as well as at the speakers. Units that employ this concept are marked "Syn" (for synthesis capability). Units marked "SQ" are those with CBS dematrixing circuitry. Those marked "Reg" (regular) incorporate Sansui decoding circuits.

The listings in the table below show the manufacturers of passive decoders, decoders to be used with add-on amplifiers, decoders that incorporate at least two amplifiers, four-channel amplifiers, and receivers that incorporate four-

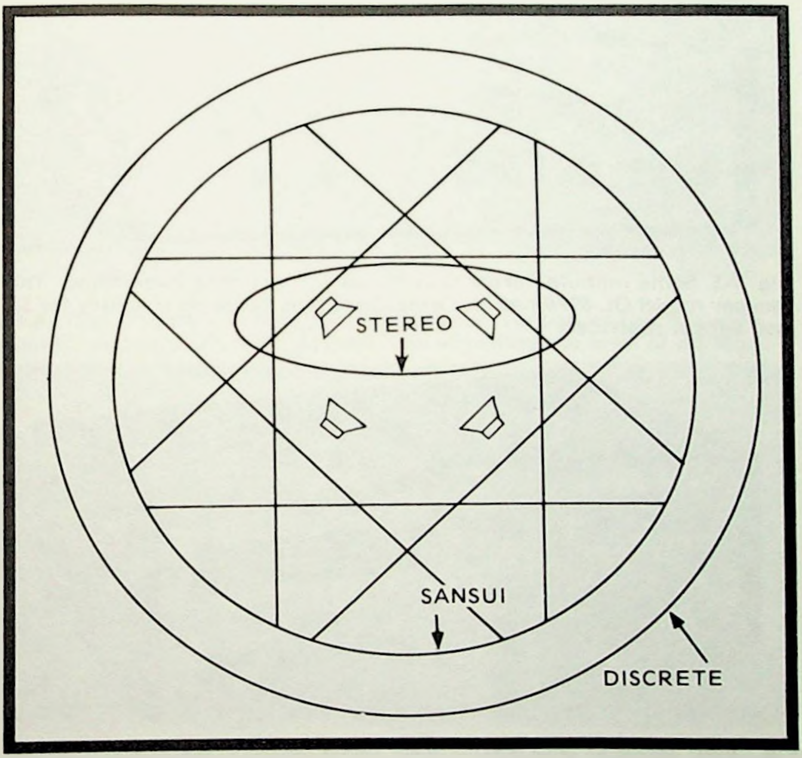


Fig. 9-3. The Sansui matrix shifts according to program content, and the result is an overall effect that resembles the discrete pattern.

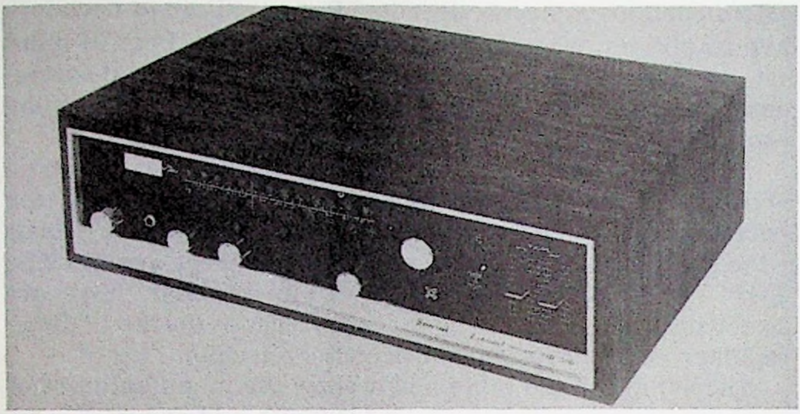


Fig. 9-4. Sansui's QR-500 four-channel receiver contains four amplifiers, a regular matrix decoder, and an "ambience reclaiming" synthesis circuit. Jacks on the rear retain full compatibility for discrete operation.

Fig. 9-5. Some manufacturers play it safe by including everything. This Pioneer model QL-600A has four amplifiers plus decoding circuitry for SQ and Sansui matrices.

Fig. 9-6. Sanyo's DCX-3300K has three matrix decoders (Sansui, synthesis, and SQ), four metered amplifiers, and a receiver with jacks for discrete operation.

Fig. 9-7. Panasonic SA-6800X includes a very sophisticated four-channel synthesizer for use with conventional stereo records as well as 300 watts of audio.

Fig. 9-8. This SA-6400X unit is a lower powered version of the 300-watt Panasonic, and contains the "joystick" for adjusting the level of all four speakers from a single control.

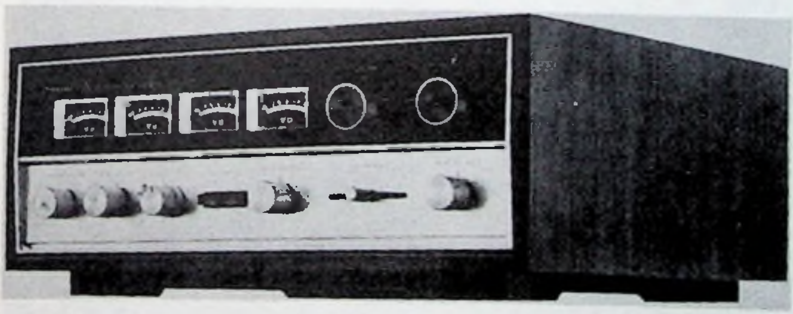


Fig. 9-9. Sansui's QS-500 contains two amplifiers and a decoder. It connects to an existing stereo system and reads out the level of all four channels on its integral meters.

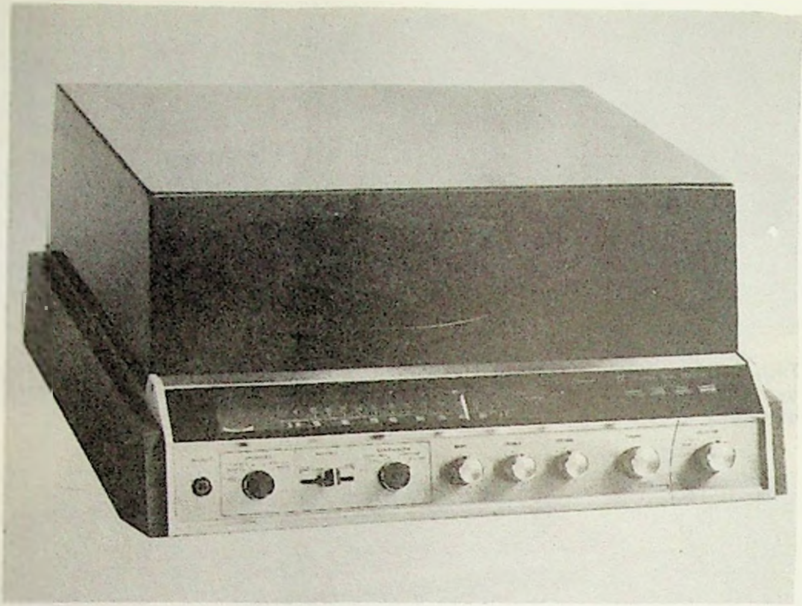


Fig. 9-10. Sansui's step away from full "componentry" is this all-in-one four channel MQ-2000 unit. It synthesizes but does not contain the company's matrix decoder.

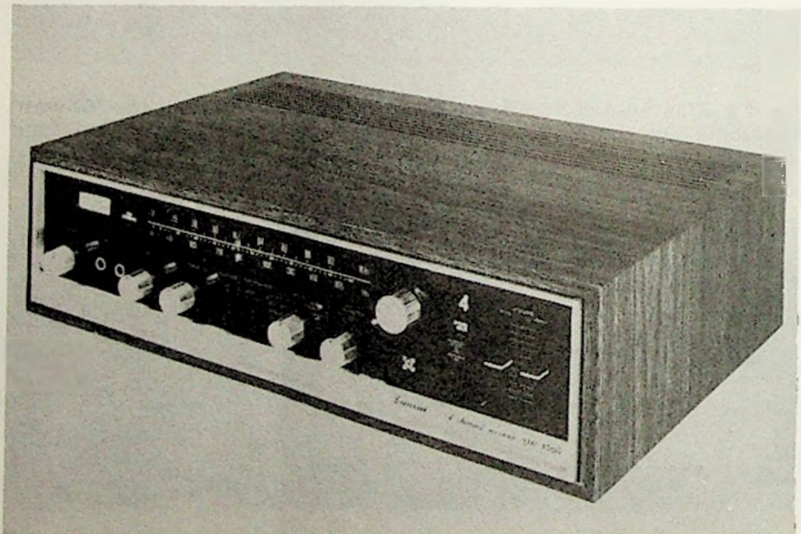


Fig. 9-11. The Sansui QR-1500 four-channel receiver synthesizes and decodes, and includes four modestly powered amplifier systems.

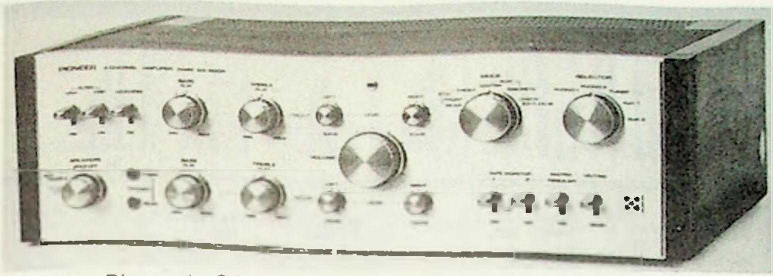


Fig. 9-12. Pioneer's QA-800A is a new approach to equipment: the unit contains four preamps and four amplifiers. It decodes SQ and Sansui matrices and synthesizes as well. It is expensive.

Fig. 9-13. Sanyo's DCX-3000K is a sensibly priced answer to quad. The unit synthesizes, decodes Sansui and SQ. Blend controls allow adjustment of spatial presence. Includes four complete preamps and amps, quality receiver, and sells for $220.

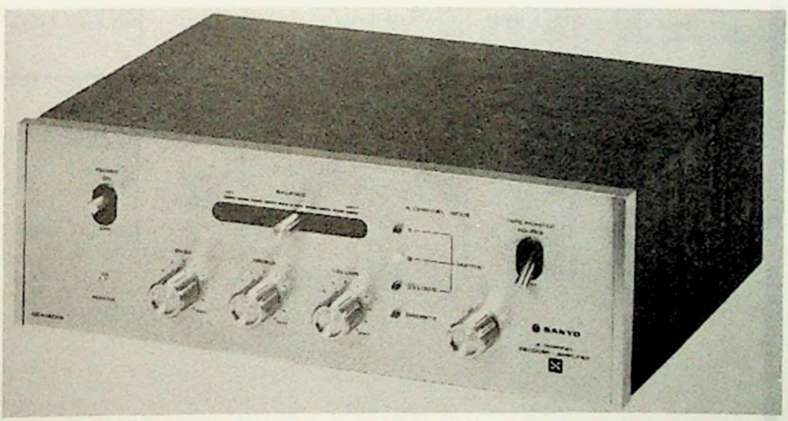


Fig. 9-14. Sanyo's DCA-1600X is a four-channel preamp and two-channel amplifier. It has decoders for Sansui and SQ and is designed for use with an existing two-channel stereo setup.

Manufacturer	Passive Decoder (Type & Price)	Decoders for use with amplifiers Type & Price	Decoders with amplifiers (Type)	Four-channel amplifiers (Type)	Receivers with decoders (Type)	Receivers with decoders and amplifiers (Type)
Akai		SS-1 (Univ. $99)		AA6100 (discrete)	AS-8100S (SQ)	
Concord		CSQ 2-4 (SQ $99)				
Dynaco	Dyn $30					
Dokorder						MR 800Q
Eico	QA-4 $30					
Electro-Voice		(Univ.) $99	1244X E-V $150			
Electro-Voice		EVX-4 (E-V) $60				EVR-4X4 (E-V)
Fisher						304,404 504 (SQ)
Harman-Kardon						Several models
Heathkit		AD-2002 (Univ.) $30		AA-2004 (Univ. Discrete)		(SQ)
JVC				Several models (discrete)		Several models (Syn & discrete)
Kenwood		KSQ-20 (SQ & REG) $109	KSQ-400 (SQ & Reg) $160			KR-6140A (SQ, Reg.)
KLH						54 (discrete w/o demodulator)
Lafayette		(SQ & Syn) $45	LA-524 (SQ) $80	LA-2525 (Syn)		LR-4000 (SQ)
Lafayette	QD-4 $25	SQ-L (SQ& Syn) $80 (Logic Type)				

Manufacturer	Passive Decoder (Type & Price)	Decoders for use with amplifiers Type & Price	Decoders with amplifiers (Type)	Four channel amplifiers (Type)	Receivers with decoders (Type)	Receivers with decoders and amplifiers (Type)
Marantz			2440 (Discrete)	4100 Syn & discrete		4415 (Syn& discrete w/o demod.)
Motorola						FH411JW (Syn & discrete)
Olson	HF 180				RA 777B (Syn. discrete)	RA 632B (SQ. E-V. discrete)
Onkyo						TS 300 (Reg. discrete)
Panasonic*		SU 3604 (Syn. discrete)	SU 3404 (Reg. Syn. discrete)			Several models complete discr reg matrix decode
Pilot				310 (SQ. Reg. Syn)		365 (SQ. Reg. Syn)
Pioneer*		QC-800A (Syn. SQ Reg. discrete)		QM 800A (discrete)	QX-4000 (SQ. Reg. Syn)	QX 8000A (SQ. Reg. Syn)
Pioneer*		QD 210 (SQ) $100	QL 600A (SQ. Syn. Reg)			
Realistic				QA 680 (Univ)	QTA 750 (E V. Syn)	
Sansui*		QS 1 (Reg. Syn) 160	Several Models (Reg. Syn)	Several models (Reg. Syn)	Several Models (Reg. Syn)	Several models (Reg. Syn)
Sanyo*			DCA 1600X (Reg. SQ) $120			DCX-3300K (Reg. SQ. E V. Syn)
Scott				499 (discrete)		554 (Syn)
Sony		SQD-1000 (SQ) $100	SQA 200 (SQ) $130			SQR 6650 (SQ. Reg.)
Sylvania	PQ $10					
Toshiba			QM SC-410 (Syn. discrete)	SB 404 (Syn. discrete)		SA 504 (Syn. Reg. SQ)

LABEL	MATRIX SYSTEM USED	NUMBER OF RELEASES
Ampes	SQ	1
A & M	Sansui	several
ABC	Sansui	less than 10
Audio Spectrum	E-V	less than 10
Audio Treasury	Sansui	several
Barnaby	SQ	several
Black Jazz	Sansui	less than 10
Capital	SQ	less than 10
Command	Sansui	about 10
Crewe	E-V	1
Dynaquad (Vanguard)	Dynaco	1
Epic	SQ	less than 10
Golden Crest	SQ	less than 10
	E-V	about 10
Monument	SQ	several
Ode	Sansui	1
Ovation	Sansui	1
	E-V	more than 10
Project 3	SQ	several
	E-V	about 10
	Sansui	more than 10
Quadradisc	RCA Discrete	more than 10
Quad Spectrum	Sansui	more than 10
Vanguard	SQ	more than 20

channel decoding matrixes. Listings marked with an asterisk are shown photographically in Fig. 9-4 through 9-14.

PROGRAM SOURCES

There are probably many who buy four-channel equipment without having the foggiest notion of where or how to obtain discs that are compatible with their units. By now, you should be aware of the fact that Columbia Records' discs are not fully compatible with Sansui's or Electro-Voice's matrix decoding systems. It really doesn't make any difference how fantastic a decoding system is if there are no program sources that can be used with it.

The record labels listed below are available with four-channel material; it is quite likely, however, that you'll have some trouble trying to find these labels in your local record distributor's outlet. If you plan to buy a four-channel stereo system, it would behoove you to pay a visit to the local record dealer and ask whether or not discs can be made available in the matrix format of interest to you. Take the list with you to show the salesman what labels he has to choose from.

Chapter 10

Setting Up Your Quad System

You might be surprised to find that the requirements for good home listening are considerably different for quad than they have traditionally been for stereo—and the differences go far beyond the simple deployment of four speakers rather than two. With quad sound, you're dealing with sound fields rather than sound sources, even though there are four sources of sound. Four-channel sound is to stereo what the mirror is to a motion picture: like the mirror, the four channels provide a sensual dimension that cannot be matched with a simple two-speaker approach to listening.

Stereo creates a good depth of field before you. But quad sound is very close to having six perfectly synchronized stereo systems. The comparison is pictured in Fig. 10-1. The correlation between channels is such that each pair of speakers, regardless of where the pair is situated, is the equivalent of a stereo system in and of itself. With discrete, of course, this effect is accentuated tremendously for there is no crosstalk between channels and no channel interrelationships that are not intentional. And the multidimensional aspects of quad, coupled with the built-in capability of a stereo quad system to duplicate the ambience conditions of a large auditorium or concert hall, are the factors that make "stereo" rooms less than ideal for setup of a quad system.

If you recall some of the early writings about stereo and its requirements, you'll remember that stereo sounds best in a room that is not too hard, not too soft—but just right. A soft room is defined as a room that is heavily draped, carpeted, and filled with "easy" furniture. A hard room is highly reflective—with few curtains or stuffed furniture, and perhaps linoleum or concrete floors and hard ceilings. Stereo's requirements called for sound bounce, but not too much of it, sound soakup (within limits), and good dispersion of the loudspeaker units, which should be placed a distance apart that is equal to two-thirds of the distance between the listeners and the loudspeaker units themselves.

That was stereo.

A good quad room is a soft room. The softer the better. With quad, you aren't interested at all in reflecting the sound around the room. You'll be depending solely upon artificial sound bounce that radiates from the two rear speakers. Any hardness in the room will tend to interfere with the "apparent" acoustics of whatever size hall you want to simulate with the indirect and reflected sounds from the back channels.

But using a soft room introduces other problems, because you can't rely on wall bounce to distribute your sound for you. It means you'll have to have plenty of power to spare in the

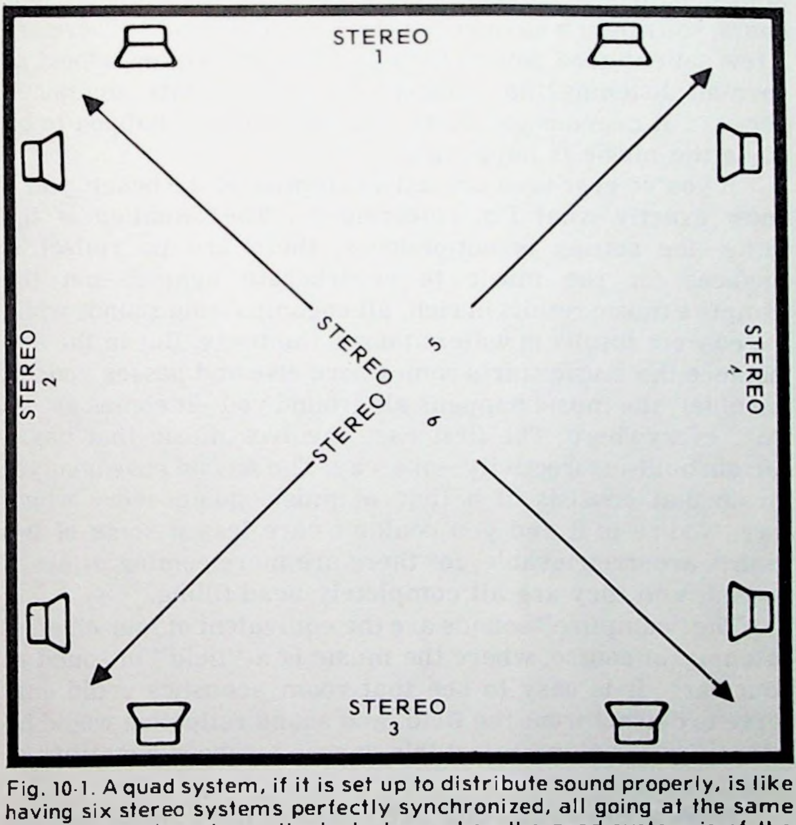


Fig. 10-1. A quad system, if it is set up to distribute sound properly, is like having six stereo systems perfectly synchronized, all going at the same time. The analogy is particularly true when the quad system is of the discrete class.

amplifier, for example, because the sound will be soaked up by the furnishings as soon as it's radiated. It means that you should be using omnidirectional speakers if possible, and there are several reasons for this other than the fact of soakup —which is sufficient reason unto itself.

Have you ever listened to sound in an anechoic chamber? It's very much like hearing a concert in the open air. The sounds that pass your ears are lost forever, for there's nothing around to keep the strains around for a while—no walls or surfaces upon which the sounds can dance to titillate your ears. The effect amounts to an emptiness or weakness, which you can directly interpret as a shallowness in the high-end region.

If listening to music in a soft room can be likened to the almost unpleasant experience of listening to a concert outdoors, you might well question the advisability of it. There are a few salient good points, though. There is another aspect of open-air listening that cancels the weak points mentioned above ... it depends on where you, the listener, happen to be while the music is happening.

If you've ever been around a campfire at the beach, you'll know exactly what I'm referring to. The situation is the same—the setting is out-of-doors, there are no reflective surfaces for the music to reverberate against—but the campfire music results in rich, all-encompassing sounds while the concert results in watered down faintness. But in the one instance the music starts somewhere else and passes you; in the other, the music happens all around you—it comes at you from everywhere. The first case involves music that has a certain built-in directivity—one way. The second case involves music that consists of a field of music going every which way—you're in it and you couldn't care less if some of the sounds are irretrievable, for there are more coming at every instant, and they are all completely head-filling.

The "campfire" sounds are the equivalent of four-channel listening, of course, where the music is a "field" of sound at your ears. It is easy to see that room acoustics could only serve to detract from the field, and sound reflection would be intensified to an uncomfortable degree by sheer quantities of sound sources.

There are two basic characteristics of quad sound, and the proponents of one often cannot understand the appeal of the other. On the one hand, there is the surround effect that puts the listener in the middle of the band—which was our argument against acoustics. On the other, there is the ambience effect that keeps the musicians up front and the listener in the audience. Would the "soft room" approach kill the effect for the listener who wants to be part of the audience rather than part of the band?

No. Here's why. Being a member of the audience in Carnegie Hall is to sit in one of the seats and hear the performance from the front, as modified by the **acoustical environment of the hall itself.** This modification consists of hearing a sound initiated from the front, followed, perhaps milliseconds later, by a montage of reverberations caused by music playing on whatever surfaces are available inside the hall, as shown in Fig. 10-2. Suppose you listen to a program that was recorded at Carnegie Hall. Do you think you will hear the subtleties of the reverberations if your listening room has four walls within ten or twenty feet of each other? Perhaps so, but the fact that your four walls can make the sounds bounce before they would in the hall will certainly augment the effect, and in some cases nullify it altogether. By eliminating the

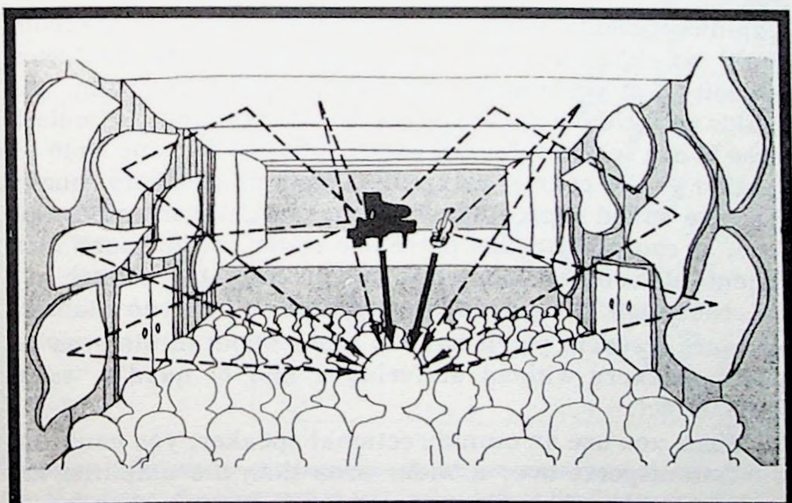


Fig. 10-2. In a concert hall, very little sound reaches a listener from the stage. Most sounds arrive indirectly after being bounced from the back of the stage or one of the walls of the auditorium. Any given sound, then, is experienced for a period of time that starts with the arrival of a direct wave and ends with either the last bounce or the masking of the reverberation by new sounds being radiated from the front.

sound bounces within your room, you can use the two front speakers to generate the on-stage sounds and the two rear speakers to generate the spurious acoustics according to the proper timing and intensity required to simulate a performance in Carnegie Hall.

The theory of this approach is good. Unfortunately, it introduces factors that complicate the problem, since soft rooms soak up high frequencies and lessen the effect of directionality. It means, then, that listening in this type of environment will require a certain bolstering of the high frequencies, and perhaps limiting of the lower frequencies. Even bolstering the high end is not the ultimate answer, though, because of the highly directional character of the high frequencies. No matter how much the high end is emphasized, the high frequencies will be lost to the listener if his ear is not in the path of sound. In normal rooms, the listener relies on bounce to reclaim high frequencies that are radiated out of the path of his hearing. But no capability exists in the soft room, as indicated in Fig. 10-3.

It boils down to a set of requirements that differ markedly from those you're accustomed to if you're a stereo fan. Ideally, you want the sound from all speakers to radiate omnidirectionally, so that no matter where you sit in the room, you'll be exposed to the same frequency range and sound intensity that you'd hear at any other spot in the room. This goal is not achievable, of course, if you're sitting particularly close to one speaker, for the sound will tend to come from the speaker you're nearest rather than from all speakers equally. Discrete sound reproduction offers a significant advantage here, of course, because there are fewer of the sound components of the other channels in any given single channel. As a generalization, the greater the separation between channels, the more freedom you'll have to move about in the area between speakers without suffering a loss of quad's "sound field" effect.

When you use an omnidirectional speaker, you cause the sound to disperse over a wider area than the amplifier has been set up to allow. To compensate for evenly distributing these sounds over the volume of the room, you'll have to increase the gain control of the high end. If your amplifier has a midrange control as well as a treble control, it is likely that you'll have to increase the setting of both. Your ear will be best judge of just how much it should be increased, however—you can't get subjective information on questions of this nature from a book.

A tone control is not a frequency control, in spite of what many think. It is a volume control for frequencies that are already being produced by the amplifier. Lowering the treble control setting does not reduce the high frequencies that are being reproduced; it merely lowers the volume of the high end without affecting the volume of the lower frequencies. The bass control adjusts the volume of the low end. When you increase the setting of either of these controls, you are not distorting the output of the musical program, as some audio "purists" would have you believe. With quad in a soft room, and particularly with quad in a soft room that involves the use of omnidirectional speakers, it is vital that the level of the high frequencies be increased sufficiently to compensate for the soakup and directional losses.

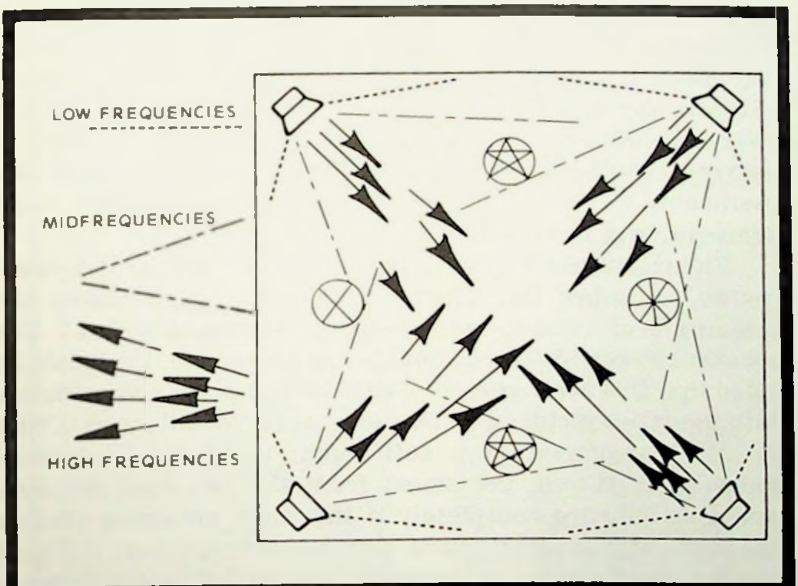


Fig. 10-3. One of the problems of soft rooms is that the listener must be in the direct path of the speakers; he can't rely on bounce to get the sound to his ears. In this sketch, a listener standing in the spot marked with the circled X will hear low frequencies but no midrange or highs. The spot marked with the circled X and cross will be within midfrequency hearing range but out of the area where highs can be heard. Realistic seating positions are those marked with the star. As indicated, listeners in these locations will hear the highs from one end but not from the other.

SPEAKER LAYOUT

In a square room, the four-corner positioning of the speakers is becoming a rapid standard; but this is not the most desirable placement, and a square room is not the best possible listening arrangement.

A series of tests conducted by one quad-equipment manufacturer determined that a grouping of speakers at one end of a room provides the best surround-sound effect, as shown in Fig. 10-4. It should be understood, of course, that there is not one "universal" solution to placement problems, and one matrix system will sound better with components positioned in the manner shown than will another. Such determining factors are degree of separation your quad system has (front-to-back, left-front-to-right-front, right-to-left), and type of program you're most interested in listening to. One factor that held throughout, however, was a preference for elevated rear speakers.

The arrangement shown in Fig. 10-4A comes closest to the "standard". But note that the two front speakers are moved in a little closer together than the two rear speakers, which will affect reproduction symmetry to some extent. According to the researchers who made the test, listeners decided that this positioning was best for mood music, rhythm and blues, vocal numbers, and records that were recorded "live."

Figure 10-4B, which was rated best for symphonies, operas, chamber music, and "big-band" jazz, shows the listening area outside the sphere of "surround sound." The speakers shown are directional in appearance, but it should be noted that this arrangement would be completely unworkable with the types pictured if the room were "dead" (soft). With the B arrangement in a soft room, using the directional speakers as shown, the sound from the two rear speakers would go unheard completely if the room softening process has been 100 percent effective. But that arrangement is a good one for small rooms or listening areas that are somewhat "hard" in character, for excellent use can be made of reflected sound, as can be seen. The front speakers radiate sounds directly into the room, as shown. If the rear speakers are directional, they radiate toward the front-speaker wall, where the sounds are diffused somewhat and reflected back to give the effect of a larger room than it actually is.

Fig. 10-4C is an excellent arrangement for soft rooms that are large enough to support the number of speakers shown, particularly if some of the speakers are omnidirectional. Ideally, the omnidirectional units would be those marked REAR (A). The advantage of this layout is that you have the choice of listening "in the music" or "in the concert hall." A simple switching arrangement on the amplifier allows either set of rear speakers to be energized (or both). Long rooms also are well suited to the arrangement

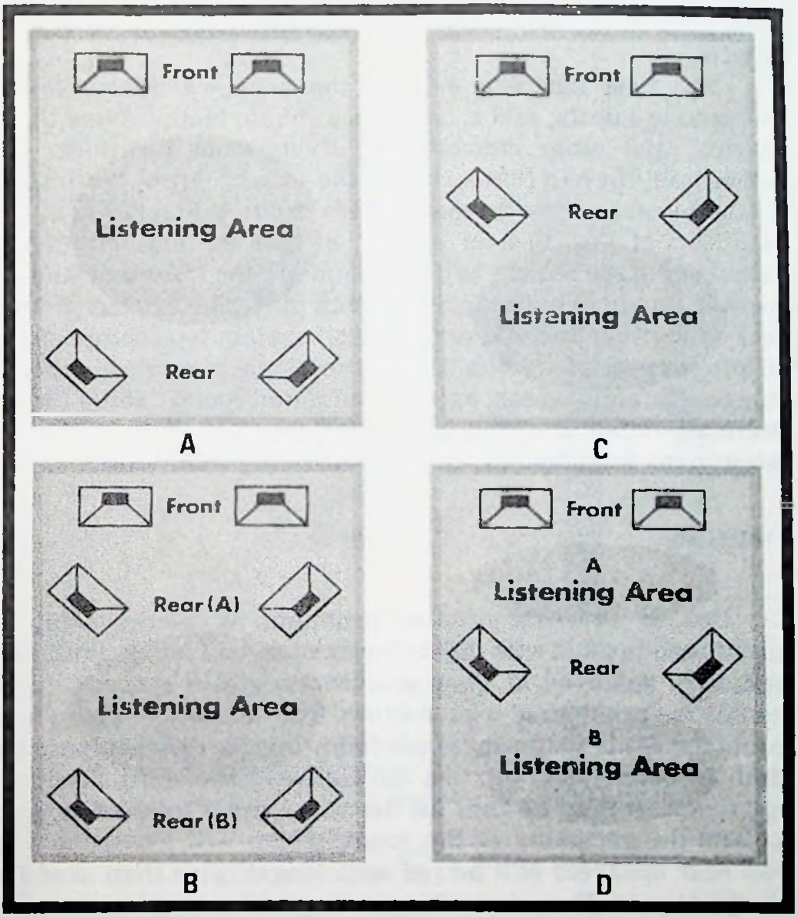


Fig. 10-4. In a series of quad listening tests conducted by one manufacturer, the positions as shown above were rated as "best" for various music types. In all cases, the rear speakers are more effective when elevated.

shown in Fig. 10-4D, but here again the speakers will be situated poorly if the room is soft unless the two rear speakers are omnidirectional.

The situation in my own listening room is pictured in Fig. 10-5. The room is 13 feet across at the widest point, and 25 feet long. Such limitations as a brick fireplace, entryways, and paneled walls keep me from making the room as soft as might be desired, but the lack is compensated for. The two front speakers are omnidirectional, and they are the two at the midway point. The other two are simply multiple speakers mounted on narrow corner-positioned floor-to-ceiling infinite baffles.

The first listening area is the area we devote to entertaining guests, and it contains a divan, chair, coffee table, lamps, and other conventional living-room furniture. The other half, the half that's "inside the sound," is our "intimate" listening area. The layout differs from that shown in the sketches of Fig. 10-4, of course, in that the omnidirectional speakers at the room's halfway point are the front pair and the corner-mounted baffles are the back pair. The arrangement is very effective, and allows our sound system to accomplish the triple purpose of serving as a conventional stereo for either of the two listening areas, a quad "surround sound" setup for our intimate listening area, and a "full dimension" four-channel stage area for listeners seated in the living-room side.

WIRING

One of the ever-present problems of stereo setup is multiplied twofold with quad: the existence of wires. With four speakers deployed at diverse corners within a room, there comes the problem of wire routing. The simplest solution is to route the leads to the speakers following the shortest possible path in each case, but this isn't always the most practical approach. It may be that all the wires must follow one path around the periphery of the room, which will mean that the two rear speakers will be fed with longer wire than used for the front pair. There is nothing wrong with using longer leads for one set of speakers, of course, as long as the speaker leads are capable of handling the amount of power to be passed through them.

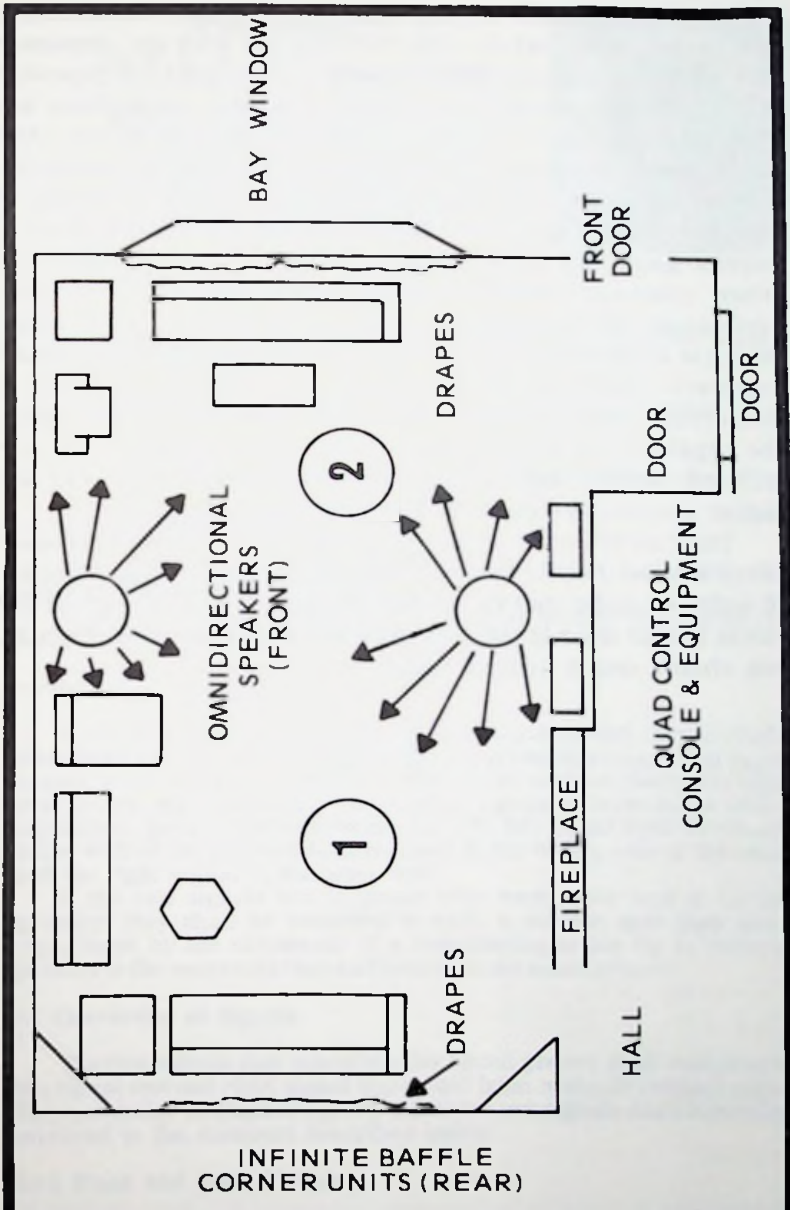
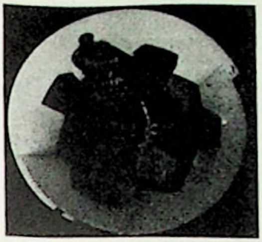


Fig. 10-5. In long rooms, such as this 12- by 25-foot example, omnidirectional speakers placed at the halfway point serve a multiple purpose, providing two listening areas so that "surround sound" and concert hall effects can be realized without switching.

It is unfortunate but true that most speaker installations are inadequately set up. The fault lies not with the consumer but with the manufacturer of wire that is sold as "speaker wire." In component type stereo systems, amplifiers are typically capable of handling large amounts of power. And large power outputs require heavy conductors to carry the power excesses. Wire has resistance, and the amount of resistance depends on the diameter of the wire. A long wire of narrow gage has a relatively high amount of resistance; and when connected to a speaker, it changes the effective resistance (or impedance, since we're dealing with audio, which is alternating current) of the load. The least that could happen is that a mismatch occurs, as far as the amplifier is concerned, and the power delivered to the load is well below the capability of the amplifier. At the other extreme, the wire will get warm under some conditions of loading, and the higher amplitude signals will be distorted at the speaker.

The rule of thumb that always seems to suffice for speaker wires is what I call "the eighteen rule": Use 18-gage wire for 18 watts of audio up to a maximum distance of 18 feet. If the run is longer than 18 feet, if the power is in excess of 18 watts, you should use a heavier gage.

Appendix I
RIAJ "Regular Matrix System Disk Record (Standard)"



REGULAR MATRIX SYSTEM DISK RECORD

Standard of The Engineering Sub-Committee, The Record Industry Association of Japan

Prepared on March 23, 1972 by the Engineering Sub-Committee of the RIAJ

(This is a translation by Sansui of the original Japanese document.)

1. Scope of Applicability

This standard shall apply to the regular matrix system disk record which is commercially marketed.

JIS regulations set forth under S. 8502 (Disk Record) shall apply to all aspects of such record not covered by this standard.

2. Recording System

The sound groove of the regular matrix system disk record shall be modulated by two—left and right—signals in two directions at 90 degrees to each other and at 45 degrees to the record surface. Such two signals shall be converted from multiple original signals in accordance with the regulations given under subsection 2.1. The left signal shall be recorded in the wall of the groove which is closer to the center axis of the record, and the right signal in the other wall.

If the two signals are in phase with each other and of identical quantity, they shall be recorded in such a manner that they can be reproduced by the movement of a reproducing stylus tip in directions parallel to the record surface and lateral to the sound groove.

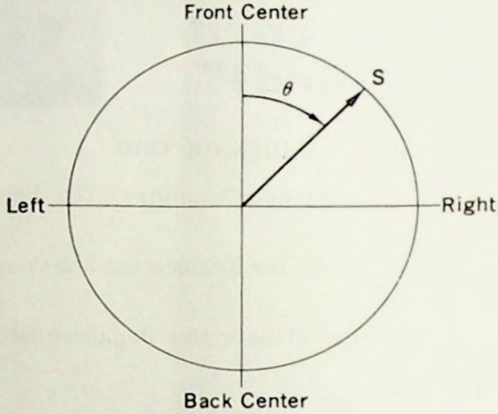
2.1 Conversion of Signals

The two signals that modulate the sound groove shall comprise one left signal and one right signal converted from multiple original signals. The conversion of original signals into these two signals shall basically be achieved in the manners described below.

2.1.1 Front and Back Signals

A signal originated at the front center shall be converted into a left and a right signal which are mutually in phase and of identical quantity. A signal originated at the back center shall be converted into a left and a right signal (which are out of phase with each other by 180 degrees but of identical quantity).

Fig. 1. Direction of Sound Source (S)
Front Center Right Back Center Left



2.1.2 Left and Right Signals

A signal originated on the left-hand (right-hand) side of the front and back centers shall be converted so that the left (right) signal is of greater quantity than the right (left) signal.

2.1.3 Center Signal

A signal originated at the center of the original sound field shall be converted so that the left and right signals are of identical quantity but the former has a relative phase lead of 90 degrees from the latter.

2.2 Relationship of Direction of Sound Groove Modulation to Sound Source Direction

The relationship of the direction of the modulation of the sound groove to the direction of the corresponding sound source in the original sound field shall, in principle, be such that the angular direction of the former is half the angular direction fo the latter (See Figures 1 and 2).

EXPOSITION

Foreword

The Engineering Subcommittee of the Record Industry Association of Japan has compared and examined the various matrix system disc records being marketed by different manufacturers to date. Results of such studies have ascertained that all of them, with the exception of the SQ matrix system, are based fundamentally on one and the same system, that they are encoded similarly, and that they posses sufficient compatibility with one another. Hence the same subcommittee hereby standardizes them as "regular matrix system disk records."

1. Scope of Applicability

This standard governs only those aspects which are peculiar to the regular matrix system disc record. All other aspects, such as its physical dimensions and quality, shall be regulated by JIS. S 8502 (Disc Record).

The regular matrix system disc record which this standard regulates, encompasses all matrix system disc records that are cut by converting the information of sound source directions into linear modulations of a spiral sound groove.

2. Recording System

So as to ensure compatibility with 2-channel stereo playback, this standard is formulated in compliance with the stereophonic recording system stipulated under JIS. S. 8502.

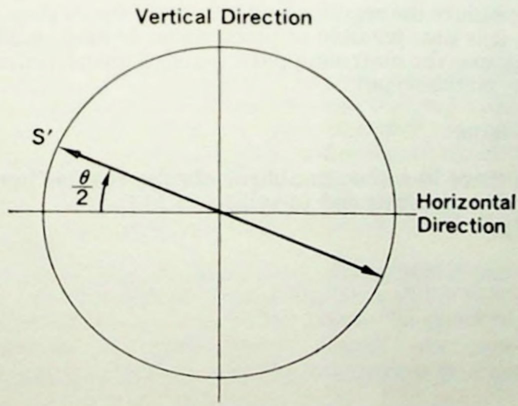
Therefore, the regular matrix system disc record manufactured to this standard, when and if reproduced by regular 2-channel stereo playback equipment, does not impair the relative sound image and sound volume balance between the left and right channels.

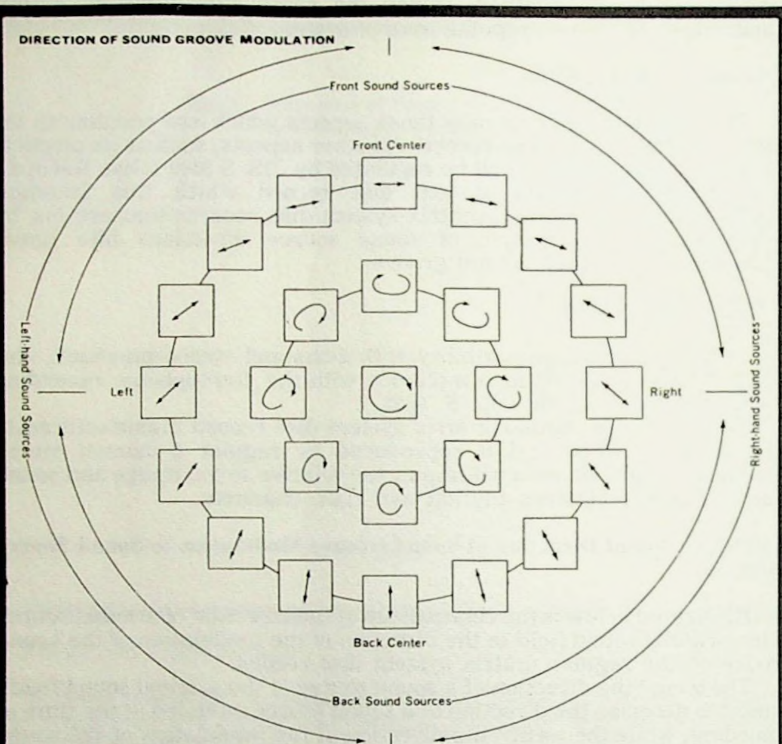
3. Relationship of Direction of Sound Groove Modulation to Sound Source Direction

Illustrated below is the relationship of the direction of a sound source in the original sound field to the direction of the modulation of the sound groove on the regular matrix system disc record.

The term "the direction of a sound source in the original sound field" is used to describe the direction of a sound source intended at the time of recording, while the term "the direction of the modulation of the sound groove" is used to describe the locus of the vibration of a cutting stylus tip.

Fig. 2 Direction of Sound Groove Modulation (S')
Vertical Direction Horizontal Direction





Note:
As the sound source moves closer to the center of the sound field, the locus approaches an ellipse.
When it is at the dead center, the locus assumes a perfect circle.

To reproduce the regular matrix system disc record in more than two channels, it is thus possible to place three or more loudspeakers freely, depending upon the matrixing parameter of the decorder used (including a speaker matrix type).

4. Abbreviation

When there is a need to abbreviate the regular matrix system disc record, it is recommended to utilize "RM."

Appendix 2

Dorren Quadraplex System

THE DORREN QUADRAPLEX SYSTEM OF FOUR-CHANNEL FM BROADCASTING

by

Louis Dorren, Director of Research
Quadrascast Systems, Incorporated

and

James J. Gabbert, President
Pacific FM, Incorporated

This paper also includes a discussion of discrete vs. matrix sound systems.
Presented at the 42nd Convention of the Audio Engineering Society, May 2-5, 1972 (AES Preprint No. 850 G-4)

1. Introduction

Popular acceptance of any basic engineering development usually dictates—from the standpoint of progress and financial return—that the initial concept be broadened to include whatever is technically and economically feasible. Thus, the recording and—or transmission of sound in single channels served only as an introductory period to the highly-acclaimed realm of audio engineering.

While this plodding, monophonic era was responsible for the outgrowth of superior equipment and techniques that culminated in true high-fidelity sound of the 1950s, it also fostered progressive experimentation which led to the first logical extension of sound from point sources. The exciting, intermediate development of two-channel or stereophonic sound distributed audio information over the length of a horizontal sound bar, providing for exact localization of sound sources between two speakers.

But acoustical technology reached its ultimate phase only when the horizontal sound bar was spread across the area of an entire plane to provide localization anywhere within that plane. We speak of it now as discrete four-channel or quadraphonic sound, an unprecedented phenomenon capable of totally immersing the listener in a sea of sound that has maintained its spatial and spectral integrity from recording to playback. Discounting various electrical gimmickry that attempt to provide a semblance of sound immersion at the prodigal expense of this collective integrity (see section 6), four-channel sound systems are exemplified by discrete tape equipment and the recently demonstrated discrete four-channel CD-4 recorded disc (a joint venture of JVC America, Panasonic, and RCA).

Prior to the inauguration of commercial broadcasting in the 1920s, recorded sound material was available only to those who could afford to purchase the necessary paraphernalia. Similarly, several years elapsed between the first public demonstrations of stereophonic sound recordings and augmentation with a broadcast system that permitted their undistorted, compatible transmission over a single FM station. Now, we have entered another unproductive limbo between the introduction of discrete quadraphonic sound material and the licensing of a commercial broadcast system for transmitting all four channels over a single FM multiplex station, a waiting period rendered intolerable because even now there exists a potential audience fully alert to the incalculable possibilities of this new medium.

2. Basic Approach To a Technical Quandary

In 1969 when the concept of the Dorren Quadraplex System was first translated into working electrical hardware, several proposals to transmit four-channel sound had been greeted with varying degrees of approbation by the industry and later rejected because of their gross inability to satisfy basic broadcast requirements. It was obvious that the outright acceptance of any four-channel medium would rest not only on superior engineering characteristics, but also on its ability to maintain the same high standards currently required by government licensing agencies throughout the world for two-channel stereo; total compatibility with existing equipment; unqualified directionality and channel separation; reproduction of the 50 Hz to 15 kHz audio frequency range for all channels; minimum distortion and channel-to-channel crosstalk; and a tenable increase in signal-to-noise ratio.

The Dorren Quadraplex System began as an idea implanted in the fertile ground of need, controversy and competitive stimulation. What no one could do had to be done, to provide continuity of this newest approach to sound propagation and to preclude the possible acceptance of a broadcast system inferior to discrete four-channel sound recording. The basic design, with all its limitations, extremes and engineering parameters, was set down on paper as a total system—rough, ragged and admittedly imperfect at first. But within days a "breadboard" apparatus existed to verify the feasibility of quadraphonic broadcasting.

Although the road from conception to maturation as a professional quality system was long and arduous, the time finally arrived for demonstration to the public in the form of comprehensive, government-regulated field trials. Pacific FM, Inc.—owner of KIOI-FM, a 150,000-watt station enjoying the largest FM audience in the San Francisco Bay Area—applied for and was granted a Station Temporary Authorization from the U. S. Federal Communications Commission to begin four-channel broadcasting with the Dorren Quadraplex System at select times during a two-month period beginning January 1971. The results of listener reaction as well as exhaustive tests upheld our expectations and corroborated the theoretical predictions of a 20,000-page computer study.

The unique difference between the Dorren Quadraplex System and all other proposed methods of four-channel broadcasting rests in the employment of a quadrature-subcarrier pair as the First Subchannel in spectrum space presently occupied by the L-R Channel. In addition, another subchannel is located at the fourth harmonic (76 kHz) of the 19-kHz Pilot Carrier. This approach enables one multiplex station to transmit four totally discrete channels of sound information in the standard FM bandwidth.

The system is fully compatible in that the composite signal transmitted by the quadraphonic station contains components for the reception of monaural, two-channel, or four-channel information, with no degradation in switching from two-channel transmission to four-channel transmission. In fact, during public demonstrations of the Dorren Quadraplex System over KIOI-FM as well as over CHFI-FM in Canada, two-channel stereophonic reception was reported to be substantially improved in both local and fringe areas.

3. Technical Description.

The undisputed value of the Dorren Quadraplex System lies in the close relationship it bears to two-channel multiplex. The present system of stereophonic broadcasting employs a Main Channel, the sum of the left and right sources (L & R), which varies the carrier to be radiated from zero to 15 kHz and produces infinite pairs of sidebands whose decreasing amplitudes are expressed in terms of Bessel Functions. The Dorren Quadraplex System also utilizes a Main Channel, the sum of all four sources, Left front, Left rear, Right front, Right rear (Lf+Lr+Rf+Rr), and it also varies the carrier to be radiated from zero to 15 kHz, with consequent generation of sideband pairs.

Both systems have a Pilot Carrier at 19 kHz plus or minus 2 Hz, for synchronization of stereo decoding at the receiver. An additional channel in each system contains an amplitude modulated signal with no carrier. Commonly denoted as a double-sideband suppressed subcarrier, it is centered at 38 kHz (twice the Pilot Carrier to facilitate re-introduction of the carrier removed at the transmitter) and extends from 23 to 53 kHz.

In standard stereo, this First Subchannel contains the difference information of the two sound sources (L-R), and the sum and the difference of both sets of frequencies satisfy requirements of the information theory, which stipulates that two separate linear equations must be transmitted to derive two separate channels of information (L+R) + (L—R(—2L and (L+R) —(L—R) —2R. The standard stereo system may also be considered as switching between the two sources at a 38-kHz sampling rate. This is time-division multiplex as opposed to the frequency-division technique we will refer to in our explanations.

Also by the information theory, in order to broadcast discrete quadraphonic sound, four separate equations, which are algebraic linear expressions of the four channels, must be transmitted by the FM station. In the Dorren Quadraplex System, the use of quadrature modulation in the First Subchannel and the addition of a Second Subchannel allows efficient use of available bandwidth space to transmit four informational channels.

The first subcarrier of the Quadrature Subchannel, which is transmitted as the cosine of 38 kHz, contains the difference of the sums of the two left and two right informational channels (Lr+L4)—(Rf+Rr). This complex difference signal will be regarded as straightforward left-minus-right information by all two-channel stereophonic receivers, providing for total compatibility.

The second subcarrier of the Quadrature Subchannel, transmitted as the cosine of 38 kHz, contains the difference of the differences of the two left and two right informational channels (Lr—Lr) —(Rf—Rf). The Second Subchannel (third subcarrier—centered at 76 kHz and extending from 61 kHz to 91 kHz—contains the sum of the differences front to rear (Lr—Lr) +(Rf—Rr). Both the First and Second Subchannels are amplitude modulated with carriers suppressed at the transmitter.

It may be considered significant to note that in substituting Rf for Rr information and Rr for Rf information in all of the equations, the monophonic and two-channel stereophonic compatibility will be maintained. However, the equations in the third and fourth subcarriers will be interchanged. It is obvious that such alteration of the equations does not change the four-channel transmission characteristics of the system and only alters its mathematical derivation.

In two-channel stereophonic broadcasting, the L—R Channel is centered at 38 kHz and extends from 23 to 53 kHz in the composite baseband signal. During modulation of the FM transmitter, sideband components created by this L—R signal and by the Main Channel signal will extend to approximately plus or minus 120 kHz when peaks in the composite baseband signal reach plus or minus 75 kHz deviation. This, according to the F.C.C. Rules and Regulations on Commercial FM Broadcasting constitutes 100 percent modulation of the FM carrier.

In the Dorren System, the Second Subchannel is centered at 76 kHz. When modulated by a 15-kHz signal, the generator outputs of 61 kHz and 91 kHz will yield, under normal conditions, a modulation index of less than 0.370. According to Bessel-Function diagrams, 91 kHz produces only one significant sideband pair, also located at plus or minus 91 kHz of the transmitted center frequency. Because these sidebands contain all the information necessary for accurate demodulation, reproduction and channel separation at the receiver, a phase-linear low-pass filter can be employed in the output of the quadraplex generator to remove all unwanted harmonics above 91 kHz.

In order to verify computer bandwidth plots showing that no excessive adjacent channel interference would be caused by the system and that only the ideal finite bandwidth signal for four-channel transmission and reception would remain, 1-mV protection-ratio measurements between adjacent channels were also performed in the laboratory (Figs. 1 through 4). The results fully bore out theoretical predictions that the overall bandwidth required by the Dorren system completely satisfies current allocations for FM broadcast stations. In fact, excitation by similar signals results in a better protection ratio for quadraphonic transmission over two-channel stereo in the same manner that the two-channel protection ratio is better than for monophonic broadcasting.

The composite signal of the Dorren System may be generated using a frequency-division or a time-division method. While both techniques yield equally fine results, the time-division circuitry is less costly to manufacture and maintain. The encoding operation actually employs analog gates to sample each of the four channels for a specified period of time (about 13 microseconds or 1/76,000 second). Four times this period constitutes one frame, which is the composite-signal time-period. It is equal to the composite frame-period of present two-channel multiplex, and thus allows the conventional stereo receiver to derive a compatible two-channel stereophonic program from the four-channel composite signal. The sample order has become Lf, Lr, Rf and Rf, providing the left-

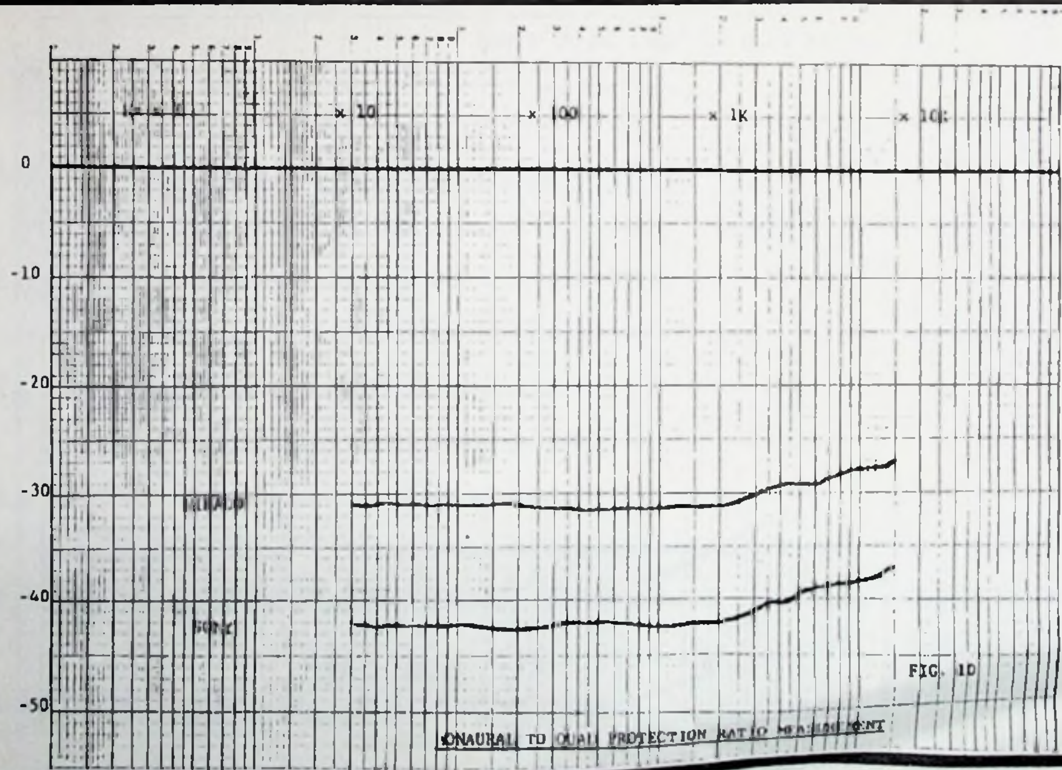
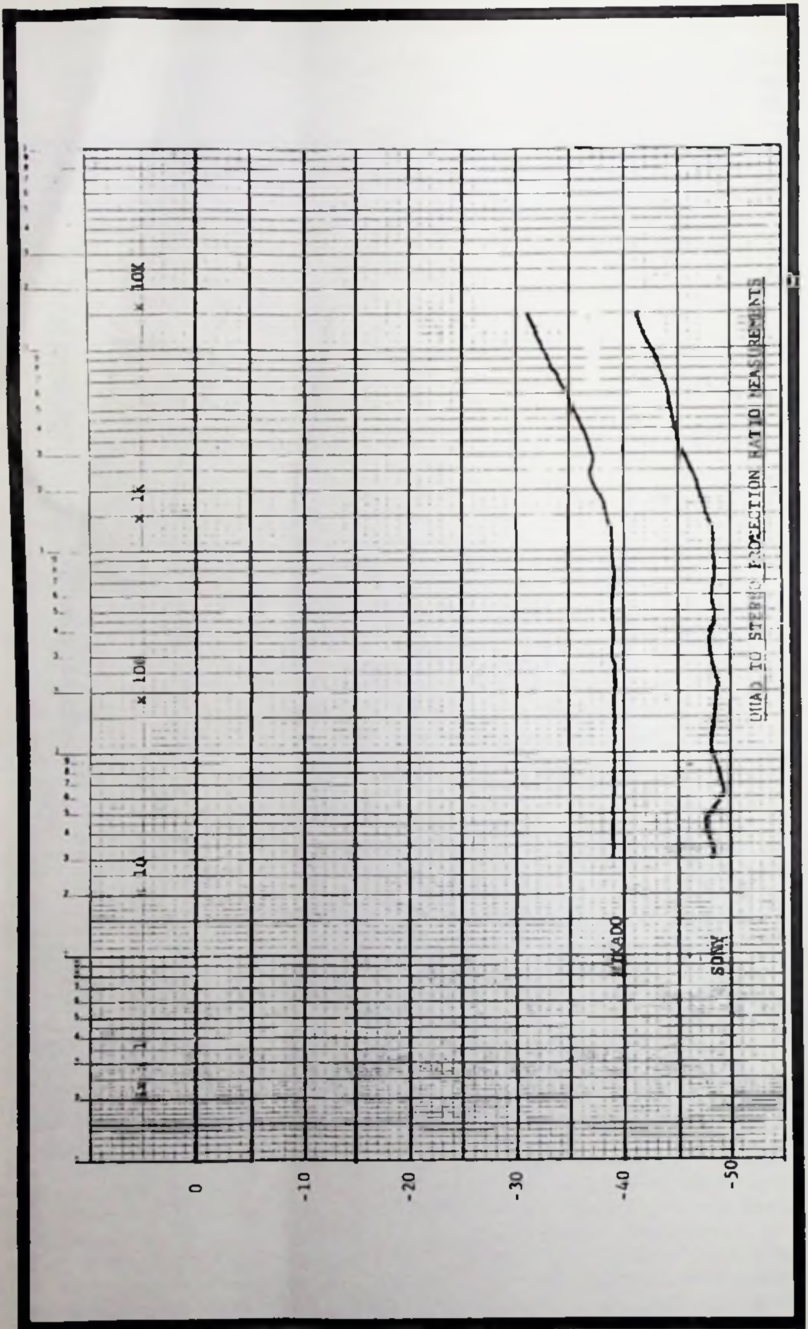
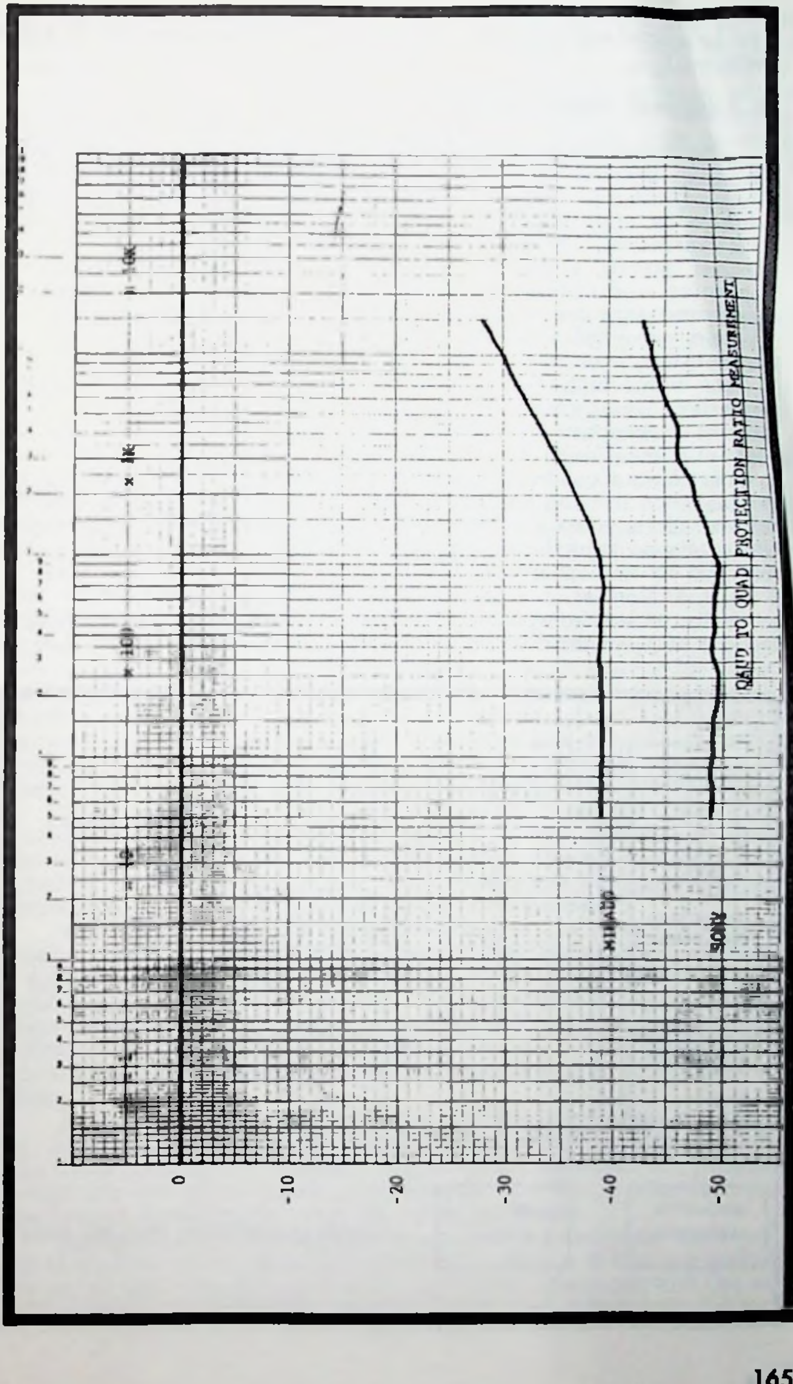


FIG. 1D

MONAURAL TO QUAD PROTECTION RATIO MEASUREMENT



QUAD TO STEREO PROJECTION RATIO MEASUREMENTS


QAUD TO QUAD PROTECTION RATIO MEASUREMENT
0
-10
-20
-30
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-50
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SONY

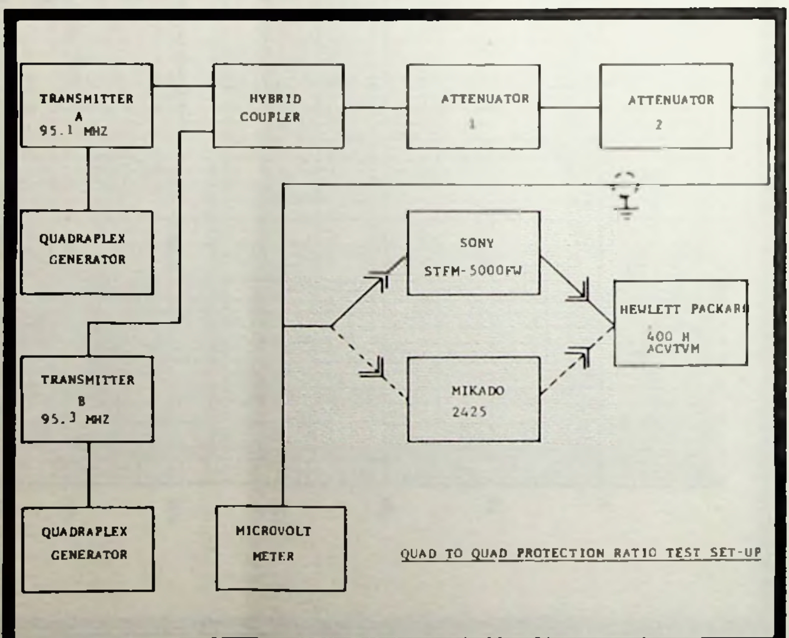
channel output of a conventional stereo receiver with the sum of the Lf and Lr sources and the right-channel output with the sum of the Rf and Rf sources.

4. Equipment Modifications

The Dorren system requires minimum modification of the equipment presently utilized by FM stations. Existing stereo generators are replaced with a four-channel generator, a light, compact unit that easily connects to, and rack-mounts above, a standard unmodified broadband FM transmitter. The quadraplex generator can be used also as a two-channel stereo generator. Specialized equipment for four-channel operation—additional microphones, mixing consoles, monitoring equipment, tape decks, a record player (when commercially available), discrete recorded tape and disc material—are also necessary concomitants to broadcast-station equipment modifications.

The same high-frequency peak-controllers and audio compresser-expanders usually installed on each channel in monophonic and two-channel stereophonic broadcasting can be incorporated in the four-channel transmission chain if higher average modulation is desired. And in contrasting a quadraphonic station to a two-channel station when both employ such limiting equipment, the audio information transmitted by the four-channel station will enjoy greater independence of amplitude, effectively boosting the apparent dynamic range of the sound information received by monophonic and two-channel stereophonic listeners.

Conversion of a standard FM receiver to four-channel sound necessitates the interposing of a Dorren Quadraplex System adapter-decoder between the discriminator output and the amplifier input as well



QUAD TO QUAD PROTECTION RATIO TEST SET-UP as the addition of another stereo amplifier and two speakers. Like its parent quadraplex generator, the adapter-decoder is a simply designed solid-state unit that can be connected to any existing multiplex receiver. Currently available, high-quality equipment is now furnished with a discriminator output jack to make interconnection speedy and costless. Older models will require a minor servicing to install the jack. However, once the Dorren Quadraplex System is approved by the regulatory agency of a particular country, the quadraplex universal adapter-decoder—in its ultimate form as a tiny integrated-circuit chip—will be made part of the receiver circuitry by all manufacturers.

5. Characteristics.

Because theoretical discussion of complex engineering principles often engenders skepticism regarding the ability of practical devices based on those principles to emulate theory, it is imperative to clearly indicate those characteristics of the Dorren Quadraplex System that have proven themselves in stringent tests as well as during on-the-air field trials at KIOI-FM, San Francisco, and CHFI-FM, Montreal.

Of primary interest to all exponents of a discrete medium is the unqualified, fully substantiated ability of the Dorren Quadraplex System to meet and exceed present two-channel stereo broadcast standards for directionality and separation. Theoretically, infinite separation is possible, but practical considerations dictate its limitation to 45 dB between channels in juxtaposition or diagonally.

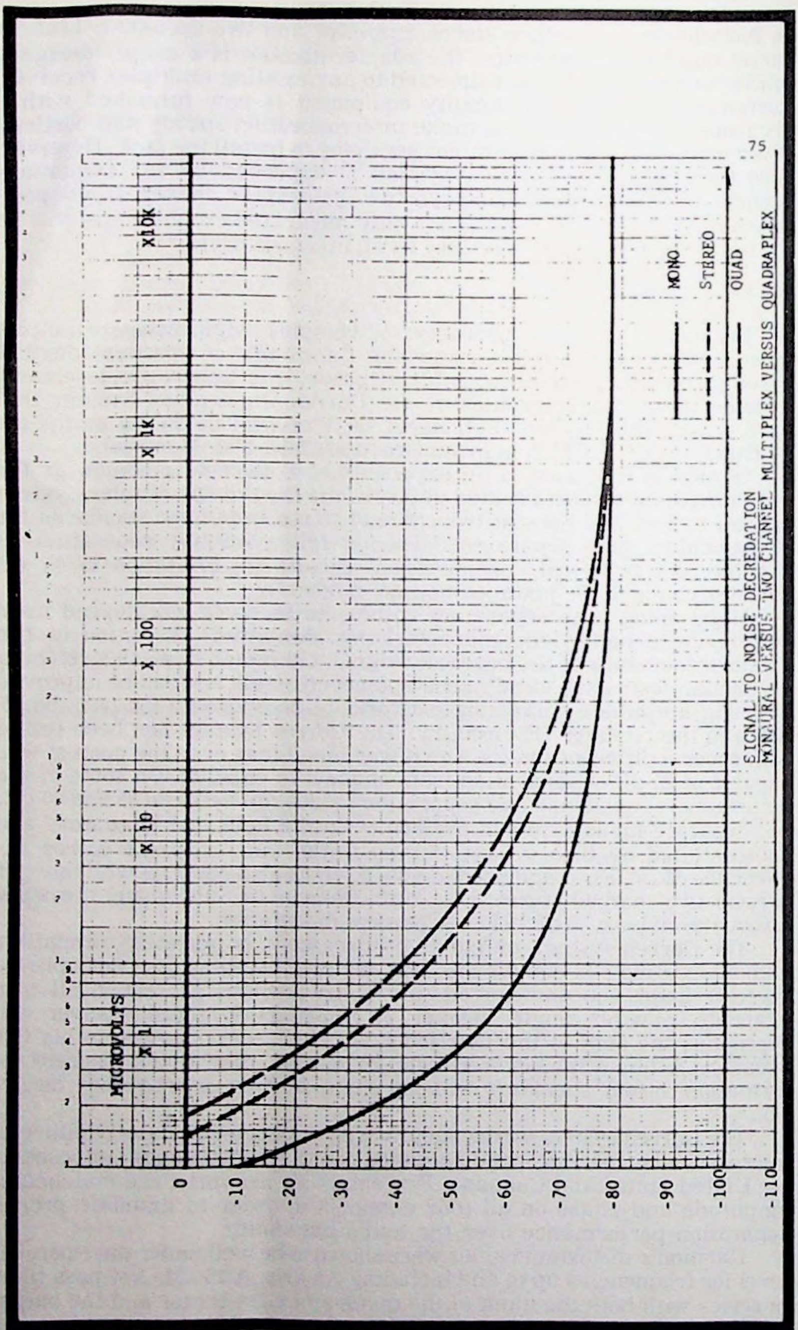
Maintaining high separation within the receiver will depend upon certain finite bandwidth characteristics. Amplitude distortion in the higher frequencies of the baseband signal will result in pulse stretching which manifests itself as adjacent channel crosstalk and can be improved by using amplitude correcting networks to compensate for poor bandwidth in the receiver IF circuitry. The Dorren system has been tested with several different tuners, and it was found that even the poorest was capable of 25 dB of separation from channel to channel. On some of the better tuners, quadraphonic separation values ran between 35 and 40 dB.

Separate measurements performed in the field and laboratory and substantiated by theory have demonstrated that crosstalk values between the Main, First and Second Subchannels also coincide with those of present two-channel stereo, i.e., Main Channel to Subchannel crosstalk attenuates at least 40 dB below 90 percent modulation.

The Dorren system, unlike its predecessors, is perfectly compatible with all existing FM equipment. An individual listening on a monophonic receiver hears the undistorted sum of frequencies present in all four channels from his single speaker. A two-channel stereo receiver will reproduce the sum of the Left front and Left rear sources on his left speaker. A quadraphonic receiver, of course, distributes discrete information to four speakers, without phase-shifting or frequency distortion.

The Dorren system works equally well with pre-emphasis of either 50 microseconds (the European standard) or 75 microseconds (imposed in the United States and Canada). Pre-emphasis networks are matched in amplitude and phase on all four channels in order to maintain proper separation performance over the audio passband.

Harmonic distortion values were shown to be well under the 1-percent level for frequencies up to and including 7.5 kHz. A 15-kHz low-pass filter in series with both the input of the quadraplex generator and the output of the adapter-decoder attenuates the harmonics of frequencies above 7.5



SIGNAL TO NOISE DEGREDATION
MONAURAL VERSUS TWO CHANNEL MULTIPLEX VERSUS QUADRAPLEX

kHz, precluding harmonic distortion measurements at higher audio frequencies. However, recorded values—measured at 100-percent modulation on all frequencies below 7.5 kHz—held true for the mono, two-channel, and four-channel conditions.

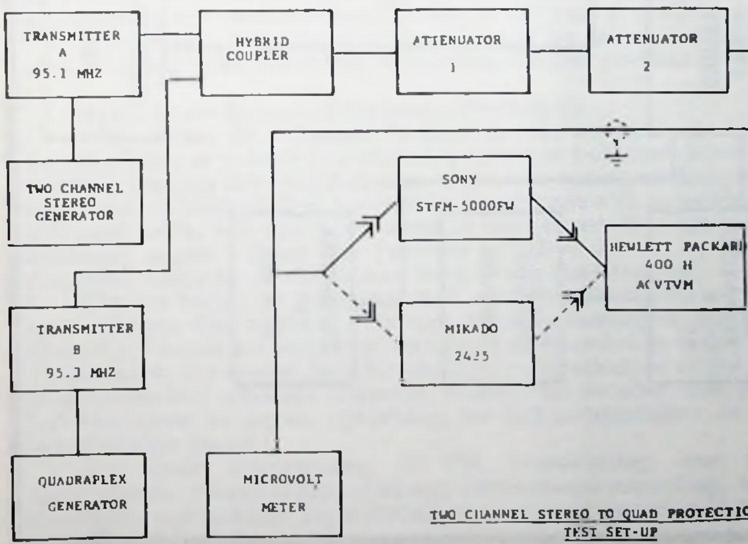
In tests conducted with a Hewlett Packard 400H AC voltmeter, a Sony STFM 5000FW stereo tuner and an A.R.F. deviation meter, audio frequency response of the overall system was flat from 50 Hz to 15 kHz plus or minus 1 dB. Measurements were consistent for monaural, two-channel, and four-channel transmission.

Signal-to-noise ratios with regard to specific receiver sensitivity were obtained at different RF levels for monophonic as well as for two-channel and four-channel stereophonic transmission and reception. In weak-signal areas, the signal-to-noise ratio for the monophonic mode was less than 1 dB and increased by 20 to 23 dB for two-channel stereo. But when four-channel transmission was substituted for the two-channel mode, the additional increase was less than 7 dB. Also, the monaural signal-to-noise degradation proved to be no greater than with two-channel stereo.

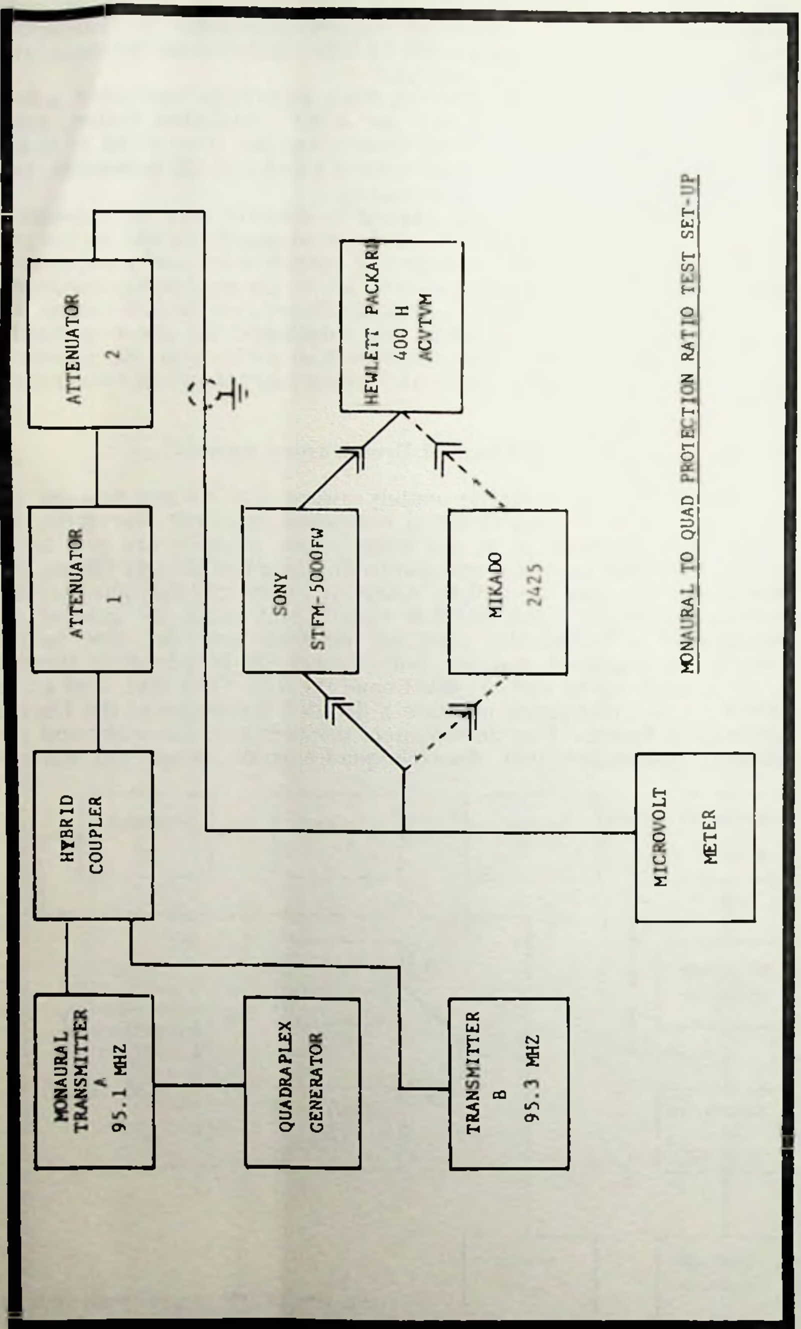
6. Why A Discrete Four-Channel Broadcasting System?

Most of us have become acutely aware that we are now on the threshold of a four-channel sound revolution, one that represents the cataclysmic upheaval of an old order whose eulogies are now being written. Although the era of quadraphonic sound has already begun, it is realistic to note that we will be catapulted fully into this glimmering, multidimensional, "third world of sound" only after the general acceptance of a technically superior, discrete, recorded disc and a government-approved, discrete, four-channel FM broadcasting system.

Of course, we've already mentioned the JVC CD-4 disc. And as its primary topic, this paper includes a detailed discussion of the Dorren Quadraplex System. One development complements the other and apparently guarantees that discrete quadraphonic sound will emerge



TWO CHANNEL STEREO TO QUAD PROTECTION RATIO TEST SET-UP



MONAURAL TO QUAD PROTECTION RATIO TEST SET-UP victorious from among a welter of less-perfect, less-honest and less-promising commercial proposals.

However, both the importance—and the urgency—of our having to select a truly discrete medium for recording and broadcasting is still being diluted by an insidious, deception-ridden, acoustical counter-revolution which insists:

1, that four amplifiers and four speakers equal four channels and, therefore, true quadraphonic sound;

2, that four channels of sound can be recorded on two record tracks and then extracted again as the same four channels of sound;

3, that separation between all but the front channels is not really important because most program material consists of sound sources in the front and only the ambience or echo in the back;

4, that the simplest electronic systems are the best—even if they don't work as indicated—because they neither rely on technical competence to engineer nor, in the case of broadcasting, do they require FCC approval for their utilization; and

5, that broadcasting acoustically "enhanced" sound does not degrade a station's signal, its monophonic or two-channel stereophonic compatibliliy, or any of the overall program information.

Such claims—either implied or boldly stated—are being made by the manufacturers and exponents of so-called "matrix" sound devices. These synthetic systems, used indiscriminately for recording and broadcast, process sound to affect an acoustical "enhancement" and an apparent feeling of sound immersion.

But by admissions of the matrix technicians themselves, these systems can provide only token separation of 3 to 4 dB between channels front-to-rear, rear-to-rear, and diagonally. This, according to the engineers of Bell Laboratories, is no separation at all and results in lack of directionality. Such blatant technical limitations seem to proclaim that matrixed sound is neither discrete nor quadraphonic, but merely a clever advertising stunt perpetrated by mercenary manufacturers to bilk an enthusiastic, four-channel-bound public of millions of dollars.

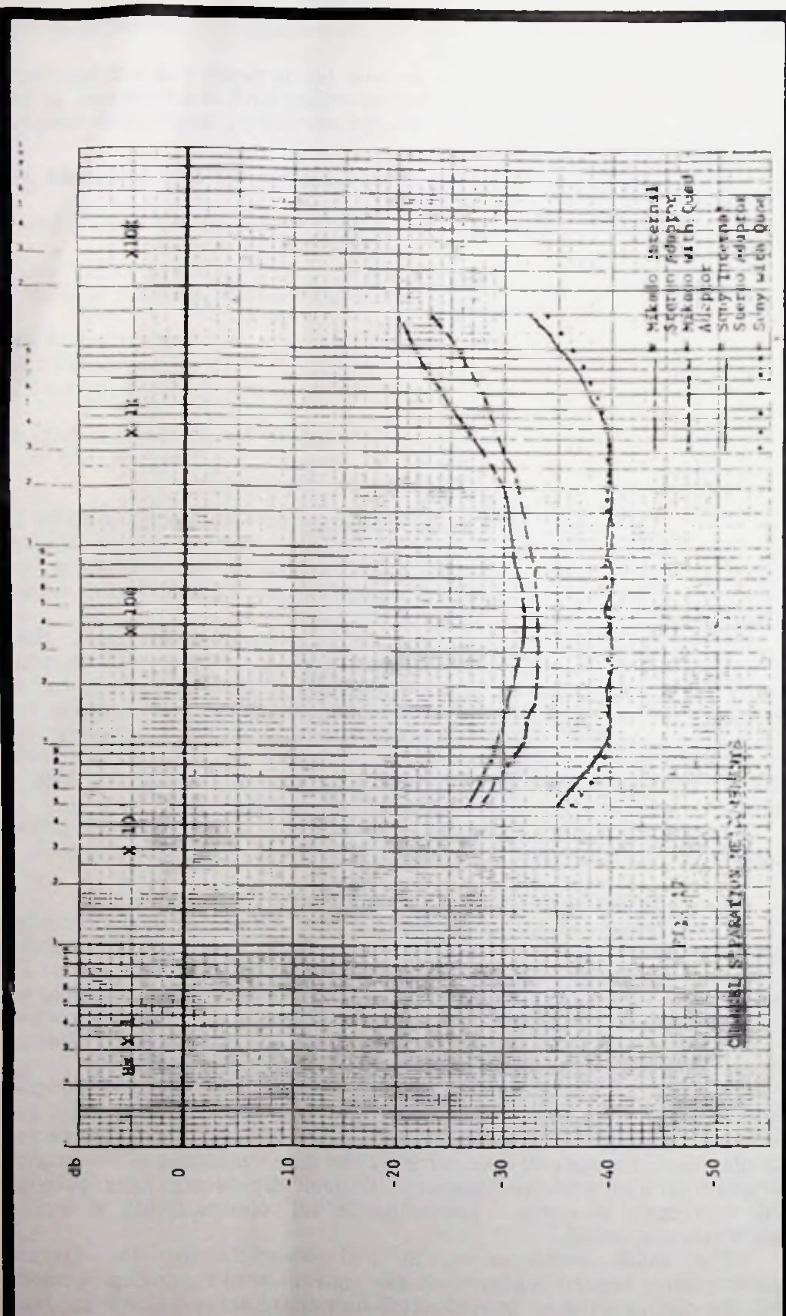
To begin with, let's examine the claims of the matrix crowd, paying close attention to the words they use to describe their products.

From a Press Release of the Sansui Electric Co.:

"(The Sansui QS 4-Channel System) is very economical; it enables the conversion of today's two-channel sources to 4-channel stereo sound, without requiring any major change of the disc cutting rechnique or FM broadcast regulations. It is also ideally compatible with monophonic and 2-channel stereo and would only need an inexpensive decoder to recover 4-channel sound." From the remarks of Clive J. Davis, President, Columbia Records, at Columbia/Sony Press Conference:

"We are happy to announce that we have developed a complete quadraphonic disc system ...Through a newly developed matrix, four channels of sound are converted into two and recorded on a disc record. On playback, the special decoder provides a reproduction of the original program on four separate channels. Without the decoder, the program will reproduce as stereo... providing for full compatibility as well as quadraphonic sound."

"For quad transmission in FM broadcasting, the stereo/quadraphonic record is played as any conventional recording. A special decoder circuit in the set provides full, four-channel reception in the home over FM Multiplex radio."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Mikado Internal
Stereo Adaptor
Mikado with Quad
Adaptor
Sony Internal
Stereo Adaptor
Sony with Quad

From a release by Benjamin B. Bauer, Vice-President, CBS Laboratories:

"CBS Laboratories has perfected a new record system which is compatible with all stereo phonographs, and yet is able to provide full quadraphonic reproduction when an appropriate player is used."

"Four-channel tapes can be broadcast over FM transmitters through the use of an SQ Encoder."

From an Allied/Radio Shack Catalog:

"True 4-Channel Sound means 4 different sound sources at the program end (records, tapes, FM) and the playback end. 'True' 4-Channel (quadraphonic) sound thus requires true 4-Channel program material and 4 Channels of reproduction (4 amplifiers, 4 speakers). The Realistic adapter yields TRUE 4-Channel sound!" (Note: this adapter is the Electro-Voice matrix decoder.)

These are some of the claims. Now, let's see how the matrix systems actually achieve the "four channel", "quadraphonic" sound they seem to imply is truly discrete. Briefly, matrix systems employ linear additive networks as well as phase-shifters to "encode" four channels into two. When encoded, the left and right outputs of the encoder will normally yield component values of each of the four signals, or three of the four signals, depending on the particular matrix system used. This accounts for the poor separation and highly-limited directionality.

According to Peter Scheiber, one of the foremost authorities on matrix encoding-decoding (Preprint No. 815 (j-5) presented at the 41st annual convention of the Audio Engineering Society, October 5-8, 1971):

"We see that four equidistant points on a plane are spaced 90 degrees apart corresponding to a 3 dB adjacent-channel separation. We may widen the spacing between a particular pair giving more than 3 dB separation, but this will narrow the spacing elsewhere, reducing the separation here to less than 3 dB. All existing 'Quadraphonic' matrixing systems embody this separation limitation: there is at least one pair of channels between which separation is no more than, and often less than, 3dB."

Please bear in mind that all matrix systems are basically the same and only rely on different addition and phase-shifting formats. Thus, matrix systems are compatible **with each other**, changing slightly the actual sound that the listener receives on his speakers.

To see how this is possible, consider the example of one popular matrix manufacturer: CBS SQ. If a signal is applied to the Left rear channel of the SQ Encoder, the relative amplitudes from the "decoder"—on a scale-factor of 1,000—equal 1.000 for the Left rear channel, .707 for the Right front channel, and .707 for the Left front channel. Because this applies in a similar manner to any of the other channels being driven independently, it appears to indicate that, when one channel of information is being transmitted, the use of "electronic logic enhancing techniques" will attenuate the gain of the other three channels, therefore "enhancing" the separation between channels. Theoretically, the resultant will work for one channel; however, with simultaneous multichannel transmission, there is no way for "logic" circuitry to separate audio information that has been summed—regardless of the phase characteristics of that audio.

As for the actual effect the listener will perceive, let's again quote Benjamin B. Bauer, before he became Vice-President of CBS Laboratories, on the subject of out-of-phase signals:

"This is a most unnatural situation which has no counterpart in the normal hearing experience. First, there is a reduction of response at low frequency. Next, there is a loss of localization and, with some observers, a feeling of 'pressure in the ears.' One can only conclude that integrity of phase relations must be carefully maintained in stereophonic sound reproduction." (IRE Transactions on Audio, Jan.-Feb., 1962; Vol. AU-10 No. 1; pages 18 to 21)

An even more convincing argument against phase-shifting matrix systems might be posed as a question: if matrix recordings are totally compatible and equal in performance to today's two-channel recordings, why is CBS keeping a dual inventory of recorded material, i.e. conventional and SQ discs?

The only answer, of course, is that matrix is no more truly compatible than it is truly discrete four-channel sound. (See Comparison Photos) In fact, the monophonic listener is seriously penalized because, according to Mr. Bauer and his CBS associates, "any sound panned to the dead back of the audience will not be received by the monophonic listener." To tell the public, or even to imply, that matrix systems are quadraphonic sound constitutes an unpardonable breach of commercial ethics. And it is time to come out before this august body of individuals and publicly denounce these matrix systems as the non-discrete, non-quadraphonic, grossly misrepresented deceptions that they really are. All of us are witnessing audio history's most successful hoax, one that has endangered those criteria or excellence that all of us have fought so hard to maintain both for the good and protection of the public we serve, as well as for the hallowed industry we represent. It is our duty as self-appointed watchdogs of the industry to unequivocally state our aspirations for excellence and likewise voice our objections towards anything that will result in a substantial diminution of our standards.

There can be no question that four-channel stereophonic sound, at its best, is the most powerful vehicle of expression an individual can experience. Shouldn't we all strive to introduce its unadulterated wonders to the greatest number of people through the economical media of discrete recorded discs and, more significantly, a discrete four-channel FM broadcasting system?

7. Conclusion

In May 1971, the results of laboratory and field tests conducted on the Dorren Quadraplex System were collated in a 418-page report for the U.S.F.C.C. Subsequently, Pacific FM, Inc, submitted a petition for rule-making that would permit four-channel broadcasting by this method. Response to such petitions usually requies six to nine months of consideration, but the proposal was acknowledged and the number RM-1847 issued within four days. Interested parties were given 30 days to respond to the petition for rule-making, and an extension of this period of comment was granted after a request by the Columbia Broadcasting System. Continued attempts have been made by Quadracast Systems, Inc., patentholder of the Dorren Quadraplex System, to satisfy all objections of the industry, the FCC and competitive systems. However, it is the contention of its outspoken supporters that the Dorren Quadraplex Systems meets and often exceeds all criteria expected of a discrete four-channel FM broadcast medium.

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