THE SOUND ENGINEERING MAGAZINE

db

APRIL 1978
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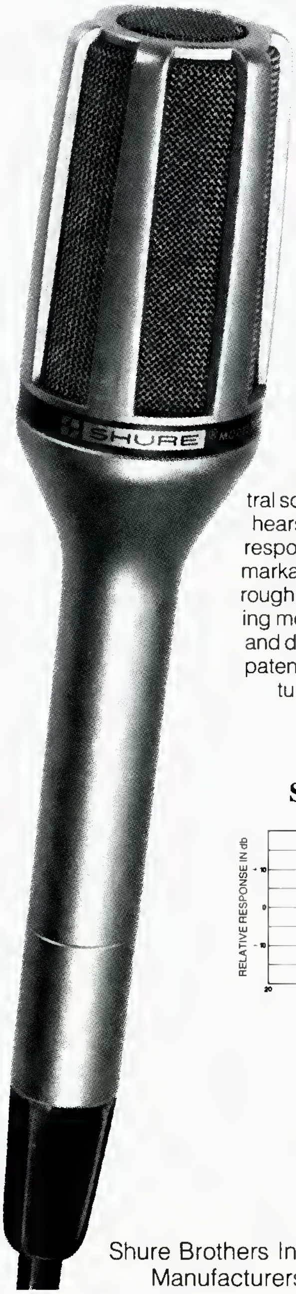

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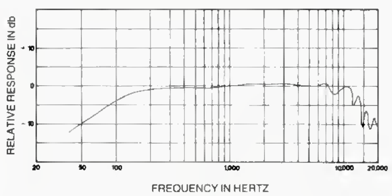

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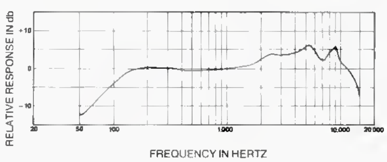
...some like a "presence" peak.



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Coming Next Month

- May's issue will contain a detailed wrapup of the convention and exhibition recently concluded in Hamburg, Germany. A lot of digital equipment was seen, and associate editor John Woram describes it all.
- Ronald Ajemian, who starts off in this issue with some basics of digital electronics goes on from the AND, ORs as he continues with his Anatomy of Digital Logic, part two.
- In The Lost Art of Recording, Arlen H. Smith details where we are at in the recording studio and where we once were. Why aren't we there now?
- It's all in the May issue of **db, The Sound Engineering Magazine.**

About The Cover

- "Girl With Violin" is the title of this photo supplied by R. Armstrong Roberts and the cover created by art director Bob Laurie.

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APRIL 1978 VOLUME 12, NUMBER 4

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db Letters

THE EDITOR:

In response to the continuing controversy in the pages of **db** over signal processing and the listeners' interests, I would like to add a few words from a different perspective. As a disc jockey, I know that different types of music are *recorded* in specific ways in order to reach certain audiences. The type and degree of limiting/compression that a station employs should reflect not only the needs of their listeners, but also the psychoacoustic and musical content of their programming.

As Fletcher & Munson pointed out some years ago, the human ear has a changing sensitivity to frequency as the sound level changes. A recording heard at 100 phons may sound flat in comparison when heard at 50 phons. The bass appears to roll off quite steeply and there will be a lack of presence at around 3,500 Hz. Any good recording engineer knows this and therefore sets his monitors at a fairly low level when recording since this is the way most records are listened to. In fact, when recorded to sound good at this low level, the record will have stronger bass when listened to at a higher level.

With this in mind, what happens to a record when broadcast on a radio station which employs limiting/compression on their signal? I think it depends on the record being played.

The newer rock, R&B, jazz-rock fusion, and disco records are often mixed in a certain way to enhance the musical content. The so called "Disco Mix" has a good deal of boost around 50 Hz to get that feeling of thump in the solar plexus which is so necessary to start feet tapping. There will also be some boost at the high end to add bite to cymbals and make those swirling arrangements of strings and synthesizers more dazzling. In part, this is what causes the dancing fever to increase: Saturday Night, as it were. For the pop and rock recordings, these techniques are used to help get that elusive "hook" which producers use to get their product recognized by the

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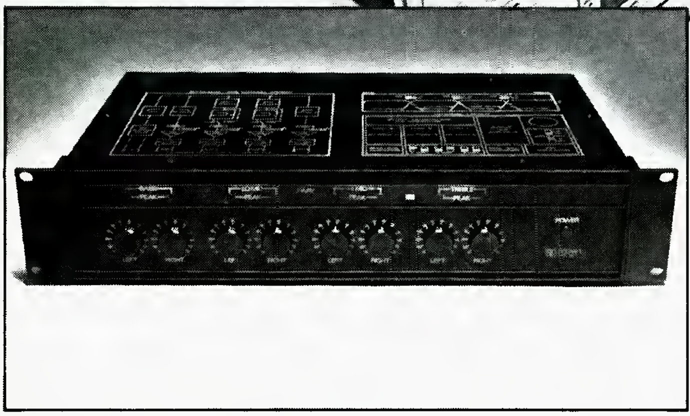

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letters (cont.)

public. As special close miking techniques, reliance on peak over vu meter readings, and special equalization in recording become more prevalent in these types of music, stations which program them must become more aware of how they affect their signals. They must also pay more attention to the type and degree of signal processing they use in order to keep the impact of the music their listeners are interested in.

Older records, mainstream jazz, and classical or easy listening music represent another situation. Often in an ensemble of orchestral musicians, or in a straight ahead jazz group, the recording engineer on a session will ask that the musicians restrain themselves somewhat so that he will be able to record them without undue limiting or compression. Close miking is rarely used and then only for isolation, and the recording is made as flat as possible. This is again in deference to the listeners' desires and use of the music. Many of these recordings are done live and overdubbing is rarely used. The effect that the producer is looking for is "depth" and realism, not a psychoacoustic "hook."

I believe one reason why Fleetwood Mac's album, *Rumors* was the biggest album of the year and is the demonstration record of choice for many hi-fi salespeople, is because of the producers' use of the mix as an integral part of the production. Even the cheapest in-house speakers sound good when played with this record. And on the radio it sounds better than most other records even when heavy signal processing is employed by the station.

In my profession as a disc jockey, I often talk to record promoters. As a condition of receiving promotional products from them I must give them written, in depth, anaylsis of each new record. This involves not only my personal opinion, but also comments on crowd response. Because of complicated surveys, the promoters know what type of clientele each club has, and based on these feedback reports they do much of their marketing strategy. I have often reported poorly on a record, not because the musicianship was lacking, but the sound just did not move my crowd. This can make or break a record and more than once I have dived for the equalizer or the gain in an attempt to strengthen a record I believe in. I have even gone so far as to re-record some records and play my tape rather than the record.

Record companies have been known to complain that promoting a record

(continued)

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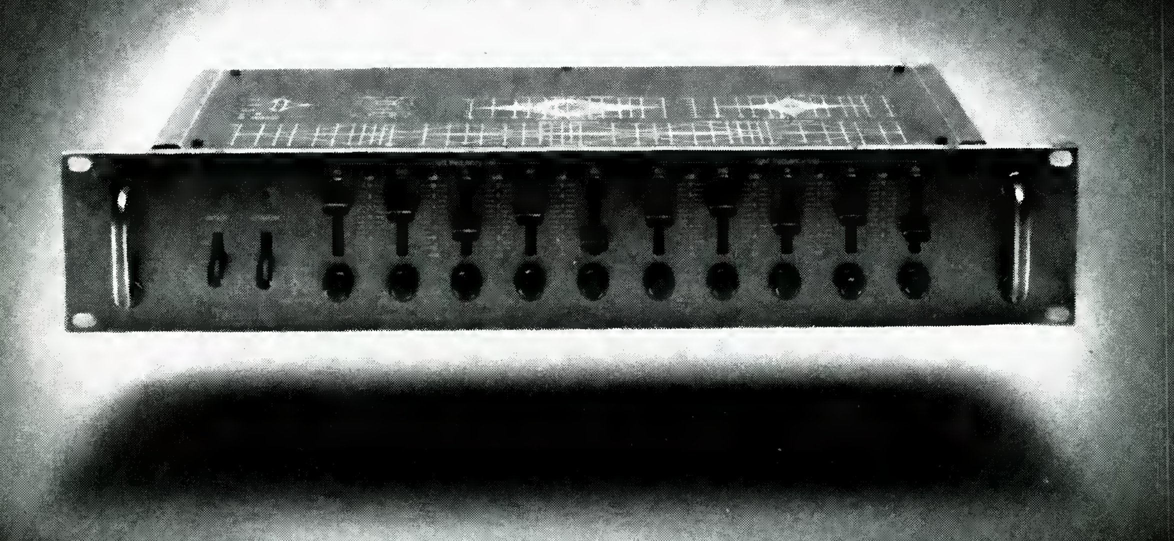

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through the discos is a waste of money. If a record is a hit in the discos, it does not follow that it will be a hit on the radio. I think the attitude of the record companies and radio managers concerning the quality of the signal is the culprit here. For many new records there is the disco version and the radio version. The disco version is typically longer, better recorded, and hotter mixed. It may have special effects missing from the radio version, such as phasing and reverb. So if a person is really turned on by something heard in a disco and later hears the same song on the radio, he or she may not even recognize it. All the dynamics will be missing due to a combination of the changed version, station's signal processing, and the reduced volume at home or in a car. In comparison, the Fleetwood Mac album is identical in every respect to the singles taken from it, and all are mixed quite hot. So it sounded great at the disco, and better than most records on the radio, either in the car or at home.

In light of this, the type and degree of signal processing employed by a radio station is obviously of interest to record companies and their promoters as well as any deejay who attempts to put together a creative show, and all listeners who have a purpose for their listening. Station managers should try to tailor their signal to the type of music they play as well as the type of listener they have targeted.

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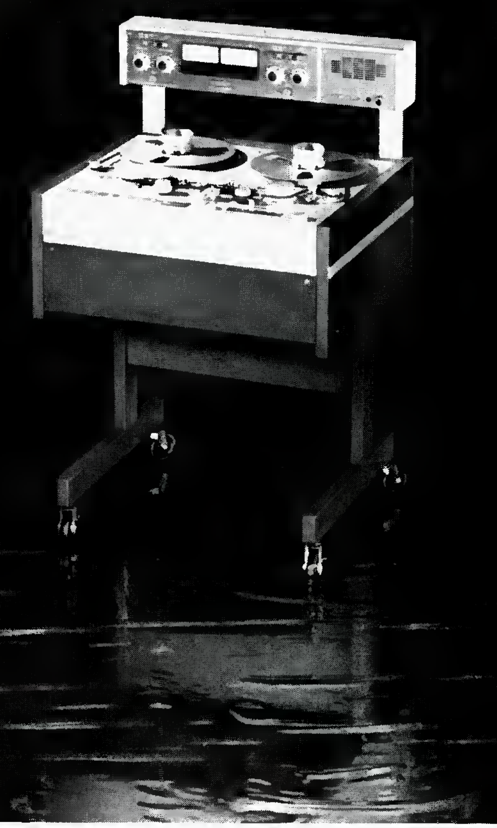

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db Theory & Practice

How Quadriphonic Fits In

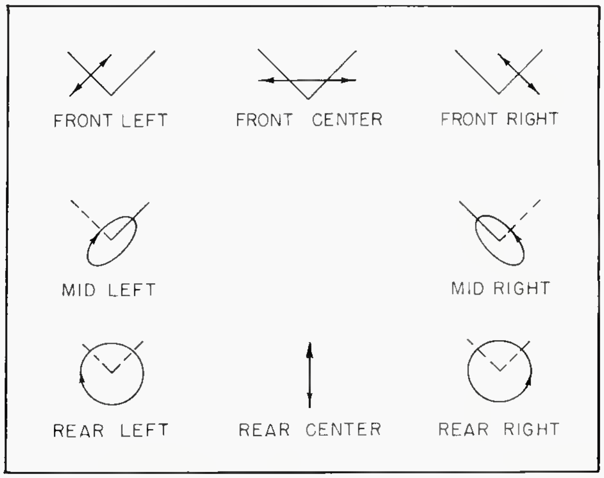
• Perhaps there is no better example of how theory and practice coincide than in the derivation of the quadriphonic system used for the SQ disc. A couple of columns back, I discussed how stereophonic really means "solid sound," so that really, however many channels may be used, if the objective is better depth perception, it is still stereo. Stereo does *not* mean "two-channel."

But using the variety of stereo that does utilize two channels, as a starting point, we can better understand how a 4-channel system can be derived from it, to produce the desired effect. And from that, with any luck, we may get into considering how to install effective quadriphonics in various kinds of environments.

Back in the days when the kind of stereo intended for the home used only two channels, a lot of work was done listening to program from two loudspeakers to see what was responsible for creating the illusion that the sound sources occupy space, instead of just coming out of a "hole in the wall," which is what any loudspeaker virtually is, from the acoustic viewpoint.

Many experiments were conducted, to determine the effect of varying frequency range, changing relative intensity, and timing, from a system consisting of two loudspeakers spaced apart, usually in front of the listener. If timing is identical—such as that used in single-channel sound, but varying the intensity fed to each unit—such changes in intensity affect the apparent

The S.Q. System



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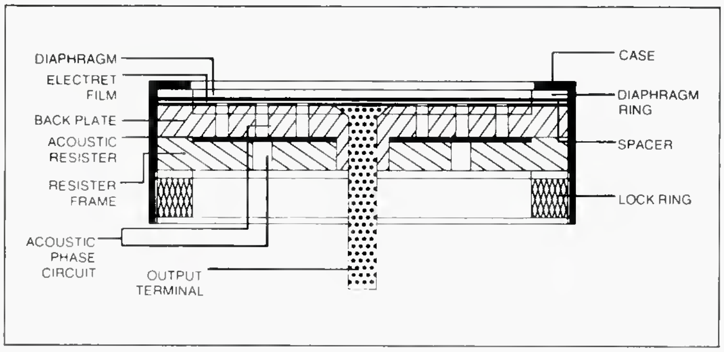

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position, to the listener, of the source.

The apparent location of the sound seems to depend on how much louder one unit is than the other. But timing also affects the apparent position of the source. In this context, the word *timing* refers to milliseconds. If the sound from one loudspeaker precedes that from the other by a few milliseconds, the one that gets to the hearer first will establish itself as the apparent source, even when the sound coming from the slower loudspeaker is quite a bit louder.

That is true mainly over the mid-range frequencies. For the low frequencies, whose wavelengths are comparable with room dimensions, there is no real sense of direction indoors. Apparent position is determined by other component frequencies that happen to belong to the same composite sound. And for the high frequencies, usually handled by the tweeter or supertweeter, the time element again breaks down, because the wave lengths are so short.

PHASING

But the effect we want to talk about here, that forms a starting point for getting into the SQ quad system, is not based on that kind of time or intensity difference. It is better related to something that caused a lot of discussion even before stereo generally made the scene on the home front. I refer to *phasing*.

The best way to illustrate phasing is to use single channel sound with two loudspeakers. Space them a distance apart and connect the two units in phase so that both diaphragms move forward together and the sound appears to come from midway between them. If you get closer to one unit than the other, the sound may appear to come from the closer unit.

But now reverse connections to one unit so the speakers are out of phase. The sound becomes discombobulated or, as we used to call it, disassociated. It no longer seems to come from those loudspeakers at all, but appears to fill the room, to completely surround you.

Before stereo program became available, one method some hi-fi nuts adopted to get a sense of space was to use two playback heads, with a few milliseconds between them, and separate amplification. They fed the output from the first pickup to both loudspeakers in phase, and that from the second pickup out-of-phase.

The effect was extraordinarily realistic, for single channel source. The first pickup came from front and center, and the second pickup simulated reverberation by seeming to fill the room, although the loudspeakers were both located in front of the listener. Then two-channel stereo came, and who needed tricks like that?

But in the course of time, some realized that two-channel stereo is limited. All of the sound comes from in front, none from behind, unless the mixing has adopted something very sophisticated, like putting in an out-of-phase delayed mix. Do you begin to see how this very naturally leads into the SQ system?

First let us take two-channel sound, coming from in front as it would be recorded on a 45/45 disc. As has been described many times before, the left channel is engraved on the left wall of the record groove and the right channel on the right wall. When an equal signal is present in both channels, in phase, both walls come into play and the groove is just like the old monophonic lateral recording: the groove wiggles horizontally, from side to side.

Signals recorded out of phase, to produce a synthetic reverberation effect, would create a vertical movement of the groove, up and down. So far, it seems we don't even need the rear speakers. But while that method works, its effectiveness depends on the acous-

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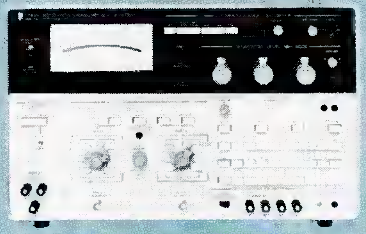

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tic environment where the program is reproduced. Let us say, for the moment, that some surroundings lead to more effective reproduction than others. So what can we do to help the tough cases?

It would help if we could have loudspeakers actually located behind the listeners. This is essentially what quadriphonic does. If, for the moment, we forget about left and right and think only about front and back, then lateral or sideways stylus movement goes to the front, and vertical stylus movement goes to the back.

LEFT/RIGHT SEPARATION

It would be relatively easy to make that separation, merely by taking the 45/45 and transcribing it as vertical/lateral, if it were not for the fact that we also want to keep our left and right separation. Let us take a hypothetical problem with such a system. Suppose we want a trumpet solo to come from right front. The component of the groove modulation representing the trumpet will be a pure 45 degree movement, modulating the right wall of the groove.

Based on a 45/45 analysis, that puts the sound on the right where it should be. But based on a vertical/lateral analysis, it contains both in equal parts, so the music would come from both front and back loudspeakers on the right, producing an illusion that the trumpeter is at the side of the listener, instead of front right.

We cannot put four independent outputs on a phonograph pickup. We can use 45/45, or vertical/lateral, not both. And we can matrix from one to the other. But essentially, the most we can have, physically, are two *independent* outputs, *either* left and right, *or* front and back. Any effort to combine them on a continuous basis results in ambiguity; more than one location of input can produce the same two-channel combination, whichever way you happen to view it.

QUADRATURE

Just a moment there. That discussion of movement refers to movements that coincide in time. Going back to the two loudspeakers, either both diaphragms move together, the same way at the same time, or they move opposite ways at the same time. Are those the only possibilities? No, they can move in quadrature too. These are two different ways in which components can be isolated—by being at right angles in the physical world, such as 45/45, or vertical—lateral; or by being in time quadrature, 90 degrees apart in their phasing.

If two loudspeaker diaphragms are moving in a 90 degree relationship, one is moving when the other is not, and vice versa. They are neither moving together, nor opposed but, in that sense, independently. This is the extra fact that the SQ quadriphonic uses.

We want to preserve the stereo illusion, across the front, so we make lateral movement, produced by in-phase left and right, representing front center, with each 45 degree movement representing its own side, left or right. Intermediate positions are represented by intermediate angles, but always in phase, just varying the relative amounts of movement.

Now, making the rotation one way, with this quadrature relationship will correspond with left rear, while the other way corresponds with right rear. Add them together and, as before if they are equal, the result is simple vertical movement, corresponding with center rear. It all adds up. What we need is a set of coding and decoding circuits, to produce these 90 degree phase shifts that will enable us to separate front and back, independently from the left-right separation.

That is precisely what the SQ system does. As the diagram shows, this enables us to put the sound anywhere we want it, around the listener. We have effectively, 4-channel resolution, but with only two information channels; we get the extra two by comparing phasing critically.

For this to be exact, we need four separate channels at the input, which must be critically processed so the composite is precisely applied with the correct phase, as well as directional relationship, at all audio frequencies. Then, on playback, or reproduction, the same code is used in reverse, to reconstruct the original four separate channels.

The system is compatible, largely because it follows a natural development, as I have shown here. If you played an old stereo, in which the synthetic reverb was achieved by delay and phase reversal, the decoder would put the reverb in center back, and program items originating at front left or front right, would come out right, too. It would not provide all the capability of a signal that had been coded from four separate channels, but at least would be compatible.

That's the theory. Does it work? CBS Labs, who developed the system, has made quite a lot of program material available, coded this way. Try it, if you have not already. ■

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3. MX-5050-QXHD Four-Channel Quarter-Inch 15 & 7½ ips • Variable speed (±7%) dc capstan servo • Other features same as two-track.

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5. Mark II-2 Two-Channel Quarter-Inch All MX-5050 features plus: • Separate transport and electronics • 15 & 7½ ips • Variable speed (±7%) dc capstan servo.

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All Otari recorders feature:
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PATRICK S. FINNEGAN

db Broadcast Sound

Crosstalk

• Many are the problems which can afflict the station's audio system, and one of these is *crosstalk*, the undesirable coupling of the audio signal in one channel into another channel. With the well controlled situation in solid state units and the excellent grades of shielded cable available today, crosstalk is not as great a problem as it once was. But that does not mean that it cannot occur. When it does happen, the cause can often be traced to carelessness in the installation of the interconnecting wiring, and when modifications are made within units. Crosstalk problems which show up in the system can sometimes be difficult to trace down, and a cure can be difficult. As with any problem, understanding and identification are the first steps in affecting a cure, so let's take a brief review of the basics.

SOME BASICS

The movement of current flowing through a wire or conductor creates a magnetic and an electric field around that conductor. The magnetic field is a complete loop around the conductor and concentric with it, while the electric field moves out from the conductor and terminates on surrounding metallic objects. These fields take on the nature of the current which is creating them. If that current is steady and in one direction (d.c.), the fields are established and steady. But if the current is fluctuating and changing polarity (a.c.), then the fields build up to their maximum magnitude, collapse and reverse, all in step with the conductor current. The magnitude of these fields is determined by the magnitude of the conductor current. Larger currents create stronger fields, and extend a greater distance from the conductor.

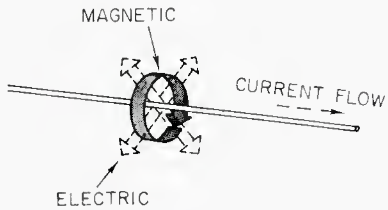

Figure 1. Fields around the current in a conductor. Larger currents produce stronger fields that also extend farther out.

When another wire or metallic object is within these fields, a current will be set up in that conductor or object, with the same nature as the original current creating the fields. The magnitude of this new current depends upon a number of factors and relationship to the field creating the current. The flux lines of the magnetic field, their density and number which "cut" the

TWO FOR THE ROAD

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The Trouper Series met the challenge of combining roadability with top performance, on the road or off, UNI-SYNC delivers sound. Designed in the same tradition, comes the MODEL 100 Professional Power Amplifier with these exclusive features:

Two Amplifiers: Not just a stereo amplifier, but actually two amplifiers in one chassis, which means accurate bass response, greater dynamics and elimination of the crosstalk distortion phenomenon.

Design: Greater efficiency due to technically superior transformer and heat sink designs.

Size: Smallest dual 100 watt professional power amplifier on the market - a 3½ inch package.

True modular construction: road tested interlocking PC board assemblies eliminate inconsistencies in performance, and serviceability problems found in hand-wired products.

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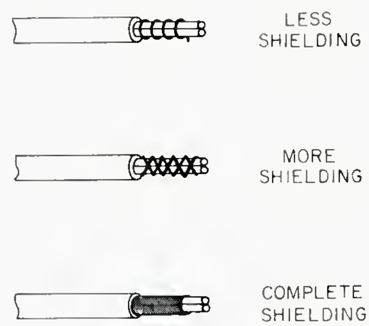


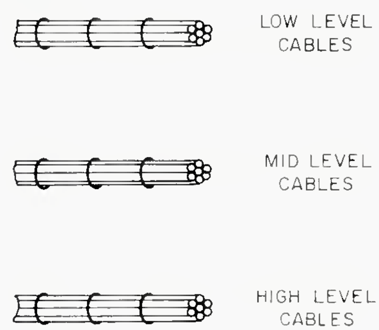
Figure 2. All shielded cable types are not the same. Some provide better shielding than others. (A) Less shielding. (B) More shielding. (C) Complete shielding.

conductor, will determine the *induced* current. The electric field induces current in a conductor by electrostatic (capacitive) coupling to that field. Either one or both fields can cause currents to flow in nearby wiring or components of another circuit and create crosstalk problems.

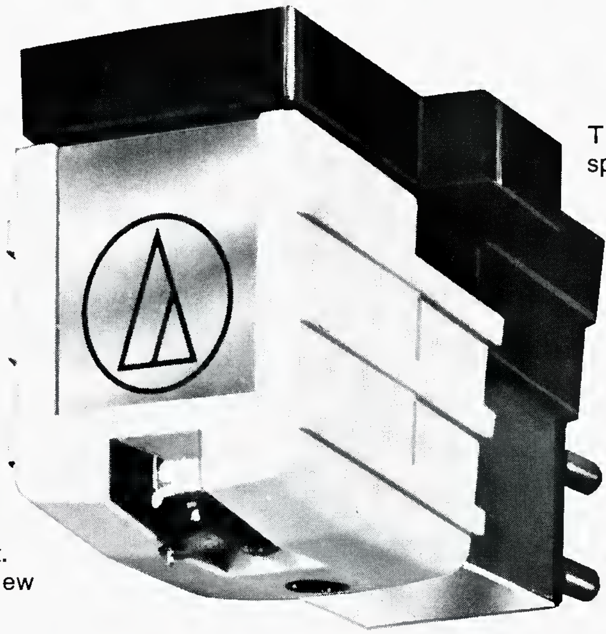
COMMON MEANING

We most generally think of crosstalk in terms of the program in one channel coupling into another channel which is carrying a different program. In this sense, the undesirable signal becomes a form of interference to the desired program audio. When the two signals mix together, the severity of the effects depends upon the ratio of signal amplitudes between the two. Usually, the undesired signal creates a low background to the desired signal and becomes a nuisance. If the coupling is very small, the crosstalk may not be heard as program but may raise the background noise level in the desired channel. In some cases the coupled level may be so high as to make only one channel usable at a time. Crosstalk in these terms is easier to define, and somewhat easier to track

Figure 3. It is helpful to separate and bundle cables according to the relative levels they carry. (A) Low level cables. (B) Mid-level cables. (C) High level cables.



Audio-Technica rewrites the book on professional phono cartridges.



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down because of the two different program audio signals involved.

OTHER WAYS

The fields around a signal-carrying conductor are no respecter of programs in *any* channel. These fields can induce currents in other parts of the *same channel* if the wiring or components are within their range. When this type of crosstalk occurs, a variety of different effects may ensue, depending upon in what areas of the channel the coupling is taking place, the phase and amplitude of the signals at that place, and so forth. A couple of examples come easily to mind: the demodulated audio from the modulation monitor coupling back into earlier stages, or the output of the console speaker monitor coupling back into earlier stages.

SOME EFFECTS

Coupling the same program audio back into earlier stages of the same channel can produce more serious effects than the mixing of two different audio programs. If the coupled signal is exactly out of phase with the ongoing signal, negative feedback occurs. This is much the same process as is created in many audio stages in a controlled manner. But in the crosstalk situation, the feedback is taking place in an uncontrolled manner.

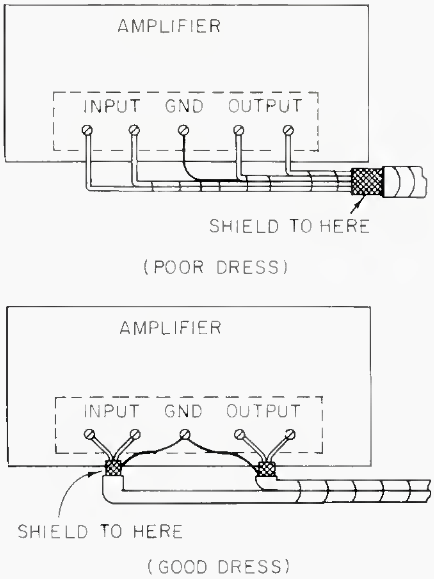

Figure 4. Poor wire dress can create crosstalk problems. (A) Poor dress. (B) Good dress.

Phase delay also enters the picture. If there is a small amount of delay, the feedback distorts the outgoing waveform. On a speaker this will often have a raspy sound in the audio as though some stage were being overloaded, or peak clipping is taking place. Greater amounts of phase delay can produce an echo effect in the program. This is about the same as when the output of a tape recorder feeds back into the console which is feeding the record section of the tape machine. If the phase of the coupled signal is such that it is in-phase with the outgoing signal, then feedback oscillation can easily occur.

STEREO

In a stereo system we have twin audio channels, the Left and the Right. A signal from the Left or the Right can couple back into its own channel and deteriorate the audio just as it will do in a single channel system. But crosstalk between the Left and the Right channels can create far more serious problems to the stereo signal. This type of crosstalk can reduce channel separation, create peculiar stereo effects such as moving singers and instruments from positions different than on the original recording, and in some serious cases can reduce the stereo signal to a monaural signal.

When a situation exists where the Left channel can couple over into the Right channel, the same conditions allow the Right to couple into the Left at the same time. Crosstalk between the audio channels can seriously degrade the stereo signal before it ever arrives at the stereo generator; that is what will be transmitted!

HOW IT HAPPENS

Unless we design and build our own audio units, we normally purchase standard commercial units and then interconnect all these together into what is our system. The units themselves are so well designed today that internal crosstalk is almost non-existent. This condition can change, however, especially when we begin to make modifications. We may desire to add an earphone jack or a switch, for example, to the front panel of a unit. Besides mounting that component, we run some "outboard" wiring to connect it to the source with which we want to use it. Unless the placement of these components and the routing of the extra wiring inside the unit is done carefully, we can easily create crosstalk problems within that unit.

When we interconnect all the audio units with external cables to form the audio system, there are many hazards which can create crosstalk situations. The average studio contains thousands of feet of interconnecting audio cables that route to many places. All this

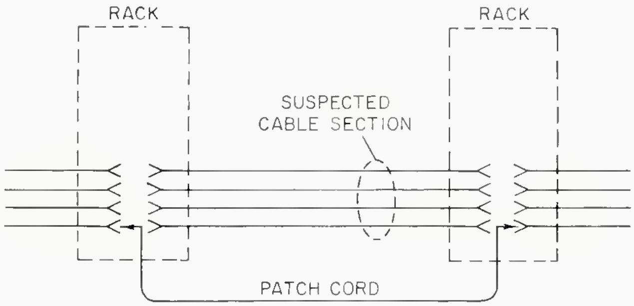

Figure 5. Use a patchcord or a temporary circuit to isolate suspected cable sections.

The first trouble spot could be the wiring itself. If we skimp and use wiring that does not have good shield coverage, we can create crosstalk problems that can't be cured without taking some very radical measures. All could easily become a "rat's-nest" unless we dress the wiring neatly into larger cables, conduits, ducts, and so forth. But when we consolidate all this wiring into a neat installation, we also bring these various signal carrying cables into close proximity to each other, opening the door for crosstalk and other problems.

shielded cables are not the same; some provide better shielding than others. The shield not only reduces or prevents external fields from entering the cable, it also contains the internal fields within the cable. Besides considering the quality of the shielding, care must be taken to see that all the shields are connected into a well controlled grounding system.

Shielding alone may not always be able to do the entire job. We can assist the effectiveness of the shielding by grouping similar signal level cables into groups of larger cables, for example, all speaker level circuits grouped together (and away from low level microphone cables).

Still other trouble spots can occur at terminal strips, terminal blocks, and jack fields in patch bays. At these locations, the protected wiring must be exposed so it can be attached to the terminals. The exposed wiring loses the protection of its shielding, while at the same time, the wire leads are brought closer together. Ordinarily, we can wire in and out of these locations without difficulty, just so long as we are careful with the wiring dress. In our desire to make a very neat installation, for example, we may place the unshielded ends of the input/output cables close together, and then tightly lace the two together!

TROUBLESHOOTING

Perhaps the best method of troubleshooting crosstalk problems is through signal-tracing methods. Make use of patchcords to isolate circuits, amplifiers, cable segments, and so forth, until the particular crossover point is located. Correction then depends upon what you discover. If the coupling is taking place in a larger cable consisting of many small cables bound together, try running some temporary interconnecting cables between the two "clean" units. If this isolates or removes the crosstalk, then run those two units over a new ciruit path which should be installed in a permanent manner. Whether or not you can still use the older cable for another purpose depends upon what will be sent over it. In severe cases, that old cable may have to be abandoned altogether.

Should the problem be occurring at a terminal block in the base of a rack, you can often cure it by moving the circuit to another pair of terminals or to another block in the same rack. We can sometimes slip up in the installation of circuits and mount a high level and a low level circuit on adjacent pairs of terminals. Crosstalk will soon remind us of the error and cause us to correct it.

RECAP

The fields around signal-carrying conductors can couple into other circuits and components that are within their range. The best cure is prevention, by the use of well-shielded interconnecting cables, a good ground system, careful wire dress at terminals and terminal blocks, and the separation of cables according to the general signal levels they carry. When making modifications within equipment units, be careful of placement of the components and new wiring. ■

db Calendar

MAY

2-6 **A.E.S. Convention,** Los Angeles Hilton. Contact: Audio Engineering Society, 60 E. 42nd St., N.Y.C. 10017. (212) 661-2355 or 8528.

10-12 **Synergetic Audio Concepts Seminar,** Los Angeles. Contact: Bidwell Sales Associates (213) 770-0300.

19-21 **International High Fidelity Show.** Georgia World Congress Center, Atlanta, Ga. Contact: Inter. High Fidelity Show, 331 Madison Ave., New York, N.Y. 10017. (212) 682-4802.

20-22 **International Light & Sound Show.** Sheraton Atlanta Hotel, Atlanta, Ga. Entertainment equipment. Contact: Multimedia International Inc., 155 Michael Dr., Syosset, N.Y. 11791. (516) 364-1912.

29-6/2 **Fundamentals of Recording.** Seminar at Banff Center for Continuing Education. Contact: Banff Centre, Box 1020, Banff, Alberta TOL OCO, Canada or Stephen F. Temmer, Gotham Audio Corp., 741 Washington St., New York City 10014. (212) 741-7411.

JUNE

9 **NRBA Sales Management Seminar.** Welsh Company, Tulsa, Okla. Contact: NRBA, Suite 500, 1705 De Sales St., N.W., Washington, D.C. 20036. (202) 466-2030.

11-14 **Consumer Electronics Show.** Chicago, Conrad Hilton Hotel, McCormick Place, McCormick Inn. Contact: CES, 2001 Eye St. N.W., Washington, D.C. 20006. (202) 457-4919.

New York Management Seminars. Contact: Heidi E. Kaplan, 14NR, N.Y. Management Center, 360 Lexington Ave., New York, N.Y. 10017. (212) 953-7262.

1, 2 **Unlocking Creativity.** NYU. Chicago.

5, 7 **Project Management for Engineers.** NYU. Houston, Texas.

8, 9 **The Federal Procurement Process.** U. of Chicago. Los Angeles.

12-14 **Management of New Technology Projects.** NYU. Chicago.

21-23 **Effective Communications for Engineers.** NYU. New York.

22-23 **Industrial Noise Control.** NYU. New York.

MARTIN DICKSTEIN

Programmers—Digital and Otherwise

• Have you noticed how complex some of the latest multi-media audio-visual shows have become? It seems as if the show producers are trying to outdo motion pictures with the rapidity of motion of slides on the many screens. Filmstrip at 24 frames a second creates the illusion of continuous movement although we know the film is being pulled through the gate one frame at a time. Slides, too, must be shown one at a time. How, then, do you speed things up? Programmers . . . of all kinds.

Until fairly recently, it was necessary to operate a projector manually. That is, someone had to push a button to get the slides to move. This is still being done, of course, in single-slide single-screen presentations, where the audio portion is a person speaking. Not that this is bad. It isn't. Depending on the circumstances, this type of show must still go on. But when it came to many images being activated, that many hands had to push that many buttons to show that many slides. And, just like in the simplest form of slide presentations, it took practice to make the show perfect. Both of these types of presentations, whether simple or complex, had to take place in "real" time (as opposed to "fake" time?). That meant that as the audio was being heard (live or from tape) the slides were to be activated at the required places in the script. The limitations are obvious. Slow, plodding, limitations of the mechanics of the projector to a maximum speed of a slide per 1½ seconds, brief but obvious black spaces between slides . . . etc. Not the best way to keep all eyes open and on the screen.

The black spaces were soon eliminated by use of a dissolve system, again, manually operated in the old days, but soon after made electrically controlled. Now that the black intervals were eliminated by using more than one projector, the speed of slide movement could also be increased with multi-projectors. The trick was to be able to control them as desired, and then to be able to record the movement so that it could be repeated as many times as required with perfect synchronization each time, after that to permit changes to be made in the programming to change effects or to vary the way the slides moved with the audio track.

This brings up the possibility of classifying the slide movement with respect to the audio. If the slides have to be shown in direct sequence with the track (and consist usually of written copy or charts) then the presentation might be labeled *linear*, as opposed to slides with abstract images, polarized slides, or random photos, which can be shown in random action. The non-linear show tries to create an effect, perhaps with emotional impact. The linear format however, has a message and requires chronological arrangement. The difference between the two has a very important effect on the programming of presentations.

PROGRAMMING

Programming in "real" time keeps the sequencing fairly simple. As the show progresses, slides are moved. Even using many projectors, the speed of the human operator with only ten fingers on two hands is restricted. The type of programmer usually used in this method of operation is of the *pulse tone* category. The simplest control used to be a device plugged into a slide projector and a remote control plugged into the "sound synchronizer" device. As a slide was advanced, a "noise" was developed which could be recorded on an audio tape. On playback, the broad band sound fed through the device and activated the slides. Most times it worked fairly well.

Another simple unit is the tape recorder (usually a cassette machine) which has a button on it which advances slides when the output cable is plugged into the carousel projector, and simultaneously puts a single frequency tone on the "second" track of the cassette. On playback, the tape tone activates an internal relay which acts as a closure and advances the slides.

The next step was a multi-button programmer which still put out audio tones for recording on tape, but now allowed different frequencies to activate more than one channel. This permitted more than one projector to be controlled, either on one screen or on several. Programming was still in real time, but now the show could be spiced up with a bit of action. Since the human doing the programming still had only two hands and ten fingers he remained limited.

As for correcting the programmed pulses on the tape, this was also getting more complicated. With one pulse at a time and one projector, the spaces between pulses allowed modification to take place fairly easily. With the multi-channel tones, a complete sequence might have to be redone because the action within the sequence did not permit editing a pulse at a time.

There were different types of devices used for programming. One was the well known paper tape unit. Holes were punched in the paper tape when an action was desired, and the desired action would take place on cue as the tape moved between tiny contacts and a roller. As the contact was made (in the hole) by the sensor with the roller, a circuit was closed and a relay initiated the required action. These programmers came with as many as 82 channels. Editing was simple enough; the wrong or undesired hole was covered over and a newer, more correct hole was punched. These could be continuously running or activated by a tape recorder pulse to run a number of holes and then stopped to await another run-cue.

Another form of programmer was a type made like a pianola. A circular drum with holes in it allowed pins to be inserted in the proper positions to activate microswitches as the drum rotated. Easy editing could be done by shifting the pins.

An amazing programming system was specially developed for use at Expo '67 in Montreal. (Remember that one?) It used a 35mm film in which the frames became a punch-hole matrix. As the film moved, light showing through the holes fell on a corresponding matrix of light sensors, each one connected to a relay which in turn controlled a device. Over 75,000 commands could be controlled with the 900 function matrix.

A similar but less complex device makes use of a slide projector using only blank slides. The blanks are punched out with a much smaller matrix, and the machine has been modified (by removal of the lens) to allow a circular "probe" to be inserted instead. The probe contains a matching matrix of light sensitive devices and these are in turn connected to an external control box or to other projectors which operate normally. Editing of the program consists simply of changing the blank slides as needed, altering the punch holes. However, there is no tone to record on tape on this one.

There are other ways to control other devices—for example, notches on film, metal foil or tape on audio

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tape, reflective foil to operate light sensitive devices, translucent tape to permit light to shine through when desired to control light sensitive devices, or even simple cams driven by clock motors to activate small switches. But all of these can't be recorded, aren't complex enough, or require delicate or time consuming editing to change the effects. Then came a breakthrough. Why not use the latest technology—digital information systems? Why not, indeed?

DIGITAL INFORMATION

In the audio and video fields, sound and picture information is being converted to digital information, stored or tampered with, then spewed out again either in its original or better-than-original form. Quality improvement and storage quantity are easily achieved. Similarly, in the programmer field, the switchover has been made to digital output devices. They have several distinct advantages and a few disadvantages, too, as compared to the earlier audio tone systems.

When depending on audio tone for programming, in order to have enough tone pulses to activate numerous devices with a limited number of buttons to handle, the tones must be available individually, as well as also in combination. There is a limit to the number of tones that can be mixed together before enough distortion to prevent proper action of the controlled devices is introduced. With digital output, there is virtually no such limit. Computers use digtial information in this quantitive way to store a great deal of material in a relatively small space. For tone pulses, a similar number of controls would require many channels.

Another advantage to digital information is that it is free of frequency distortion. Digital pulses can be read from the tape even if the speed of the tape varies quite a bit, contrasted with the use of sound pulses, when wow and flutter and speed variations cause havoc in the playback, with quite a few pulses going by without activating the desired effect.

The great advantage to digital pulses is that there need not be a real time program running at presentation speed with which the pulses have to be synchronized. All programming can be done in "unreal" time, that is, at leisure, with cues set up individually at any desired speed and stored in a memory within the programmer before actually recording. These can then be played back, watched and checked visually both on the screen and on a numerical readout on the device, and corrected in any desired way, cue-by-cue, until the effect is satisfactory. Then the entire series of pulses can be dumped from the memory onto the audio tape and kept there for future reference, or for re-entry into the memory again for further adjustment. This type of operation permits a massive amount of control information to be assembled in a short space on the tape, and allows for eye-blinding, mind boggling activity on as many screens with as many projectors or devices as anyone could wish. All it takes is practice to learn how to operate the programmers and to find out how much they can really do.

DISADVANTAGES

There are a few disadvantages, too. It does take quite some effort to learn how to manipulate the programmer until its entire capability is made available to the producer. A digital programmer is quite a bit more expensive than the tone pulse devices. When using tones they can be heard when played back through a speaker, making it possible to check them aurally as well as visually by the effect they create.

The greatest advantage to digital systems is that they can be used in producing large, complex presentations which will remain perfect for any length of time no matter how many times they are repeated. Otherwise, they are probably too expensive to even fool with when planning the general run of presentations, which tend to be simple. This also illustrates the biggest problem expensive digital systems introduce—the fever to buy equipment, produce one or two big shows, and then finding them of such limited use for all the other presentations that the devices collect dust.

There are presently over 35 manufacturers of programmer devices, each with its own features, each with its own advantages, and each probably best for certain effects and operations. Among these are such names as Arion, AVL, DuKane, MacKenzie, Tiffen, UAV, Wollensak, A-V Services, Audio-Sine, Avtek, Columbia Scientific, Clear Light, Bergen, EEG, Electrosonic, Motiva, Montage, Pavco, Spindler & Sauppe, and others. Digitals are here to stay, and will make a great advance possible in the visual effect that audio-visual multi-media presentations will have on the viewer, but they are not necessarily the overall answer to all presentations and their needs. However, digital devices have already and will continue to make their marks and to play a most important role in the look of future audio-visual presentations. ■

JOHN M. WORAM

db The Sync Track

The Truth About Digital Audio

• Every year around this time, **db Magazine** likes to offer its readers a little something extra, beyond the restrictions of its usual format.

This year, our subject is digital technology. Our research staff has recently discovered that there is actually no such thing as digital audio. Rather, it is all part of a well-planned plot to raise the price tags of conventional analog audio hardware. We have found that once the term "digital" is stencilled on the face plate of any signal processing device, it is possible to raise the list price by at least one order of magnitude (±3dB).

In the past, our ever-watchful staff has uncovered other misleading practices, such as the well-known cardioid microphone fallacy. Our research has demonstrated that the so-called cardioid polar pattern is actually kidney shaped. However, the term "cardioid" sounds more respectable, and therefore more expensive.

And so it goes. In order to eliminate any confusion over the current rash of digital expressions, we offer here a collection of the most-often-seen terms, together with their proper analog definitions. Once these terms are properly understood, it will be seen that there is really nothing at all to digital electronics. And remember, you read it here first.

A/D Converter
A device used to convert an audio signal into a series of useless pulses.

D/A Converter
A device used to convert a series of useless pulses into an audio signal.

Flip-flop
A mild heart condition, brought on by failure of a digital delay line during a live concert.

Disc Memory
Trying to remember who took the test pressings home.

(continued)

the sync track (cont.)

Fan-Out
A frequent problem with studio air conditioning systems.

Integrated Circuit
A circuit in which at least twenty per cent of the electrons are minority carriers.

Logic
A quality rarely found in digital electronics circuits.

Buffer
Someone who can explain digital electronics to an analog engineer.

Multiplexing
A digital logic circuit that confuses more than three people.

Parity Bit
Trying to con the producer into giving you equal billing with him on the album jacket.

Positive Logic
Your method of arguing with the producer.

Negative Logic
His method of ignoring you.

Ripple Counter
Keeping track of the empty wine bottles.

Random Access
A faulty record button.

Packing Density
Ability to crowd a string section into the isolation booth. Packing density decreases with fan-out.

Truth Table
Sitting around the chief engineer's office, trying to find out who erased the brass tracks.

Low Logic Level
A measure of the producer's ability to explain what he wants..

Sampling Rate
Measure of ability to check out the back up vocalists, after the session.

Bistable Multivibrator
Device used to sample kinky back up vocalists.

Floppy Disc
Leaving the test pressings next to the coffee machine.

Algorithm
A new disco step, introduced by an Algonquin Indian.

Data Block
Nervous condition, which prevents an engineer from understanding binary arithmetic.

Binary Arithmetic
Math for people who have trouble with the two-times table.

PCM
Pretty Confusing Machine. ■

Another Limiter?

So ask the cynics. That's why we made the Orban/Parasound 418A special. It's a stereo compressor/limiter/high-frequency limiter system that compresses the dynamic range of complex program material with astonishing subtlety and freedom from side-effects. It simultaneously and independently controls the high frequency energy to protect preemphasized media (like disc, cassette, and optical film) from high frequency overload distortion. It's cleaner than most linear amplifiers (THD at 1 kHz is typically 0.02% for any degree of gain reduction), and stereo tracking is locked-in for life without adjustments.

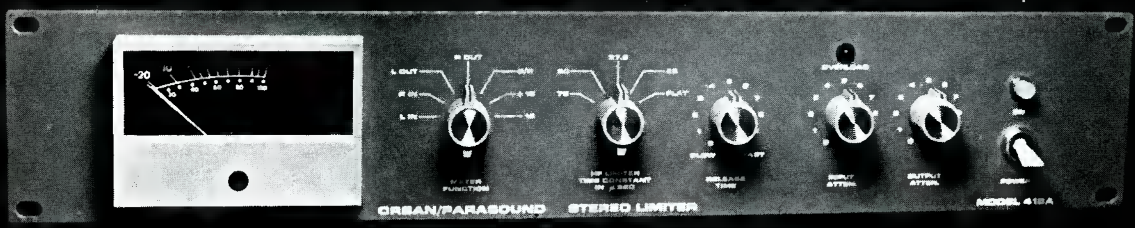
The 418A is highly "smart" and automatic. There are only three controls that affect the sound quality. This means that the 418A can speed the process for budget-conscious customers (like commercial producers) and bring them back again and again. The 418A is also ideal in the broadcast production studio ahead of the cart recorder, where it guarantees clean carts, free from overload and high frequency saturation due to excessive EQ.

The recording studio can use the 418A to generate master tapes which will transfer to disc and cassette gracefully and cleanly. The subtle, dynamic high frequency control means that high frequency equalization can be used more freely than ever before without danger of overload. The cassette duplicator and optical film recorder can condition problem masters to maximize signal-to-noise and eliminate high frequency splatter in these touchy and demanding media.

The Orban/Parasound 418A isn't "just another limiter"—it's a time-saving *system* that handles chores ordinary limiters can't touch. It's available at your Orban/Parasound dealer.

orban/parasound

680 Beach Street, San Francisco, CA 94109 (415) 673-4544



Orban/Parasound Products are manufactured by Orban Associates, San Francisco, Ca.

Circle 30 on Reader Service Card

Divide and conquer.

Conquer distortion, defeat clipping, clean up your mix.

Bi-amplification or tri-amplification with Yamaha's F-1030 frequency-dividing network can take you a long way down the road to audio perfection.

By separating high, mid and low frequencies before amplification, the F-1030 increases efficiency and headroom to the point where you need fewer amplifiers and speakers to produce the same sound level. What's more, by dividing the sound for several amplifiers and many sets of speakers, the F-1030 eliminates the cost of individual passive crossovers.

Control your own! Unlike other dividing networks, Yamaha's F-1030 offers dB-calibrated detented controls on both inputs and outputs, as well as transformer-coupled XLR and standard phone jack connectors. Twelve selectable crossover frequencies range from 250Hz to 8kHz, with your choice of 12dB/octave or 18dB/octave slopes, plus a switchable 40Hz 12dB/octave high-pass filter.

Use with confidence! Noise and distortion are virtually extinct. The Yamaha F-1030 will drive a full +24dBm (12.3 volt) output into a 600 ohm load. It will also accept input levels to +30dB.

There's just not enough room here to give you the whole story. So send this ad along with three dollars. (Please, certified check or money order only. No cash or personal checks.) We'll rush you the F-1030 operation manual. Or better yet, see your Yamaha dealer.

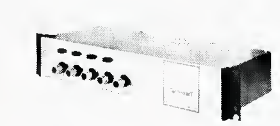
YAMAHA

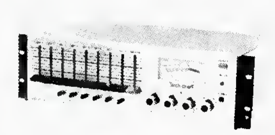
Musical Instrument Combo Division
6600 Orangethorpe Avenue,
Buena Park, CA 90620.
Write: P.O. Box 6600, Buena Park, CA 90622

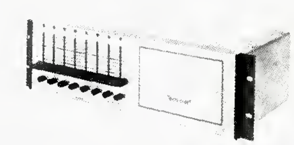
Circle 28 on Reader Service Card

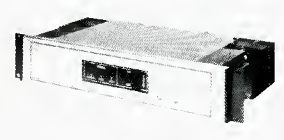
Crazy Like a Fox

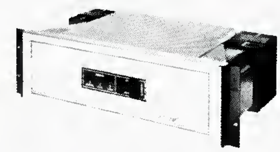

TCM-100 Mixer

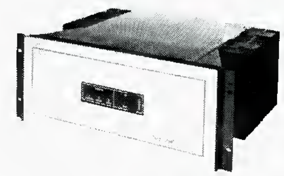

TCM-E100 Mixer-Extender

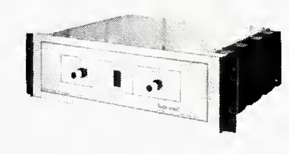

TCM-200 Mixer


TCM-E200 Mixer – Extender

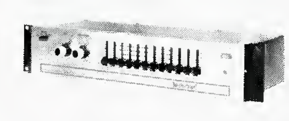

TCB-60 Power Amplifier

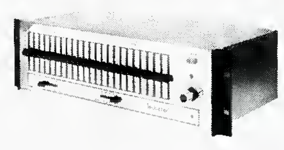

TCB-125 Power Amplifier

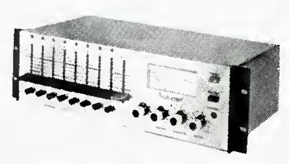

TCB-250 Power Amplifier


TCB-S160 Dual-Channel Power Ampl

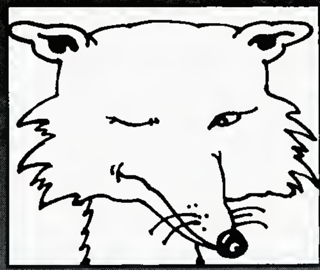

TCX-E100 Electronic Crossover


TCE-100 Equalizer


TCE-200 Equalizer


TCA-75 Mixer-Power Amplifier

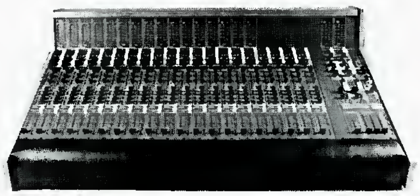

:raFT
POWER AMPLIFIER



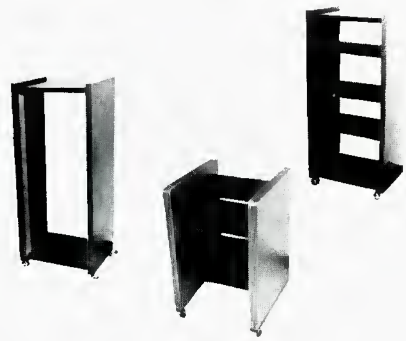
New Products & Services

MODULAR CONSOLE

● Expanding configurations from 12 x 8 up to 36 x 32 are possible with Series 1600 automated mixing console, featuring a fully modular mainframe. The console is available with the automation package installed or an automation retro-fit in two steps can be added later, providing first vca automation control cards to each input module for sub-grouping and then adding the automation processor, allowing full level and mute automation compatible with MCI's automation system. Rather than designed around a standard I/O module, the console has a separate input module and output module which interface electrically and mechanically to form one unit. Assign matrix and equalizers (3-band parametric with 20:1 frequency sweep and 4 "Q" positions per band or 3-band, peak/dip type with 4 frequencies per band) operate from separate internal subassemblies. All jacks are mounted on separate removable p.c. boards.
Mfr: Sound Workshop
Price: $10,000-$60,000.
Circle 50 on Reader Service Card



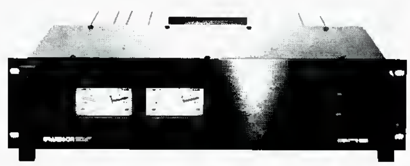
COMPONENT RACKS



● Rack-it shelves are planned to fit the special configurations of audio components. The equipment racks are 51 in. overall height and 17 in. deep. The shelving rack has four adjustable shelves and will accept equipment up to 23½ in. wide. The 19 in. deep rack contains 25 rack spaces, totalling 43¾ in. The tape console will hold any tape machine up to 19 in. wide and 21 in. long, with an additional shelf for tape and accessory storage. All units are finished in a dark walnut formica and come with 2½ in. ball-type casters.
Mfr: Midwest Sound Co.
Price: $180.00.
Circle 52 on Reader Service Card

POWER AMPLIFIER

● Complementary-symmetry circuitry are featured in Grandson amplifier, with claimed harmonic distortion at less than 0.3 per cent. Open-loop and closed-loop gain-bandwidth maintains intermodulation distortion to below a claimed 0.01 per cent. Grandson is rated at 80 watts per channel into 4 ohms, yet capable of operating at loads as low as 2 ohms, where it delivers 120 watts per channel output. Following the input capacitors, the circuitry is fully d.c.-coupled and includes a servo-control loop to maintain the d.c. offset to less than ± 25 mV. The amplifier is equipped with an integrated-circuit, bias-current regulator and thermal breakers. Grandson is available in two models, one equipped with a pair of fast-acting power meters calibrated in watts and decibels, and one without meters.
Mfr: Great American Sound Co. Inc.
Price: Metered: $349.00-Unmetered: $309.00.
Circle 51 on Reader Service Card



HEAD CLEANER

● Especially attacking "wet shed," a condition where tape binder oxide adhesive sluffs off the tape through friction and is deposited on recorder heads, Model 85-02 Pentagon Tape Duplicator, Heads & Guide Cleaner is also safe for rubber parts. In fact, the manufacturer claims that it actually conditions rubber. The cleaner is also effective on tough residue and impacted dirt clinging to capstans, pinch rollers, and heads. It leaves no film.
Mfr: Pentagon Industries
Circle 53 on Reader Service Card

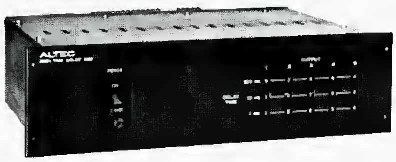
PLINTH & COVER

• For kit addicts, the BD1 plinth and cover may be combined with an arm of your choice. The plinth is made of walnut veneer, fitted with spherical, antivibration feet. The cover is acrylic, in bronze, hinged with a two-position stay. The unit stands on a totally enclosed metal base. Dimensions are 15 x 13¼ x 3 ins. Claimed rumble is —65 dB and hum level, —80 dB. BD2/A, an integrated assembly, including tone arm with automatic lift off, is also available, as well as larger units.
Mfr: Hervic Electronics
Circle 54 on Reader Service Card

RADIO COMPUTERIZER

• Expandable Series 7000 computer center offers information processing and display useful to radio automation. The control center is a Z-80 microprocessor with a computer-grade CRT. The basic sequential system also includes 16 audio source capacity (the first nine are random-selectable to 999 trays), and thousand-event memory. The system is expandable to four separate CRT channels, up to 10,000 events, and as many as 64 audio sources. Features include plain English programming, programming error detection, programming lookahead (19 program and 9 time events, with optional clock), multi-CRT capability, dual (music and voice) program busses, programmable source cards which are switch-selected to match various makes or models of audio sources, a six-source multi-cue system, and multi-level subroutine capability. The system has computer ports for billing/traffic automated systems, telephone line remote connection, VEL, a "debug" module, modem, and a load/dump cassette system.
Mfr: Cetec Broadcast Group
Circle 57 on Reader Service Card

TIME DELAY SYSTEMS



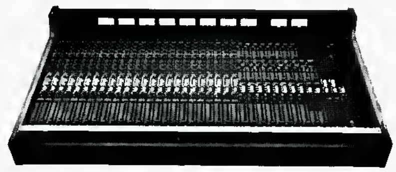
• Three time delay systems, Models 1640, 1660, and 1661, have been developed to serve situations in a sound reinforcement system when seating under a rear balcony can't be covered by the central loudspeaker cluster or when the room is very long. Model 1640 features shift-register digital circuitry, with a maximum delay time of 120 ms; six outputs are at fixed 20 ms. delay intervals. Models 1660/1661 have Random Access Memory circuitry and accept up to six memory modules for a maximum delay time of 510 ms., with five output modules.
Mfr: Altec Lansing
Circle 55 on Reader Service Card

POWER AMPLIFIER

• Two-channel power amplifier Model S500-D generates 500 watts per channel in a 3½ inch rack space. Forced cool dissipators keep the unit cool even with 2½ ohm loads. The output section, easily replaceable, is completely modular. Especially slated for p.a. use, the device is lightweight and compact.
Mfr: Audio Marketing, Ltd. (HH Electronics)
Price: Under $1,000.00.
Circle 56 on Reader Service Card

24-IN/8-OUT MIXER

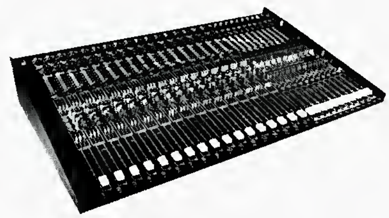
• A sophisticated echo circuit that can send the reverb signal to print, the studio, or the control room is featured on Model 15 24-in/8-out mixer. The unit contains the following features: switchable six-bank equalizer; new knob controls that allow pre- and post-fading for both cue and echo mix: two 8 x 2 sub-mixes that can be used separately or cascaded and from which either bus or tape can be monitored; 100 mm. sliding pot controls; plug-in modules. The power supply is a separate unit.



Mfr: TEAC
Price: 24/8: Under $9,000.
16/8: Under $7,000.
Circle 59 on Reader Service Card

AUDIO MIXING CONSOLE

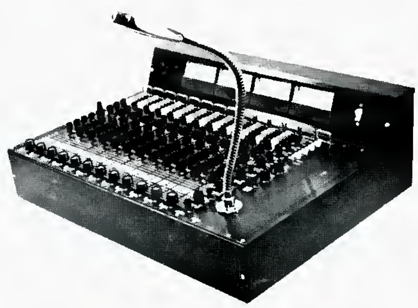
• Totally modular construction is employed for the XPC Series mixing consoles, eliminating motherboards. The unit is designed for studio or touring use, built of ⅛ in. aluminum extrusion. Included features are full stereo assignable submastering, eight selectable band point equalization, switchable channel breaks, two effects send (selectable post-preamp, post-equalization, post-fader), solo, pan, phase reverse, pad, variable gain, three-light l.e.d. meter and integral rear patchbay. Any number up to fifty input modules may be used with a single power supply. Additional modules include submasters, A/B/Solo masters, an eight-mix master module and a communications module.



Mfr: Custom Audio Electronics
Circle 60 on Reader Service Card

12-INPUT MIXER

• Twelve inputs, with four main outputs, four cue sends, and four echo sends are included in Model 12 x 4 mixer. Four four-inch lighted v.u. meters can be switched to the outputs. There are two possibilities for modules, type N for control-room mixdown and type TS for studio talkback and slating. The input modules provide gain set in 10 dB steps with two input-pad positions, pushbutton track or submix assign, monitor-only solo, line/mic input select, odd-even pan, and four cue sends which can be pre- or post-slider and one of which doubles as echo send. Echo returns to each track and to the mixdown are included. There are a number of interface options available.



Mfr: Interface Electronics
Price: Standard: $3,720
8-Input: $3,000.
Circle 61 on Reader Service Card

OTARI

From Otari for uncompromising recordists. MX5050·2SHD designed for peerless two·track quarter·inch masters.

It's an exception of compact recorders. Specially designed for critical professional applications from the ground up. It leaves nothing to be desired. 68dB signal-to-noise and greater-than-60dB crosstalk. Variable speed DC-servo capstan motor for less than 0.05% wow/flutter and ±7% pitch control. +19dBm headroom before clipping. Motion sensing control logic. Front panel edit and cue; stepless bias adjustability; built-in test and cue oscialator; all front accessible. 600 ohm, +4dBm or -10dBm fixed-level output and XLR connectors. Remote controllability for all transport functions. In short, it's a sheer professional masterpiece to produce desired 15 or 7-1/2 ips masters.

The performance and reliability have been fully proven since its original version was introduced in 1973, in more than one thousand practical applications by broadcasters, studio recordists, audio-visual professionals and musicians all over the world. For the full story of this unique and compact professional machine, ask anyone who uses it or get in contact with your nearest Otari distributor.

Please send me details on
MX5050-2SHD

Name

Company

Address

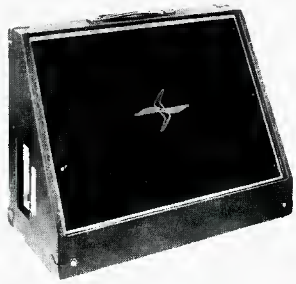
db

Japan: Otari Electric Co., Ltd., 4-29-18 Minami Ogikubo, Suginami-ku, Tokyo 167, Japan **U.S.A.:** Otari Corporation, 981 Industrial Road, San Carlos, California 94070
Canada: Noresco Manufacturing Co., Ltd., 100 Floral Parkway, Toronto, Ontario M6L 2C5

Circle 35 on Reader Service Card

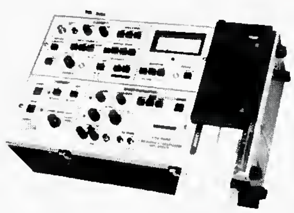
STAGE MONITOR/SPEAKER

• Meeting the dilemma faced by entertainers who want to monitor their own work while performing is the purpose of Model 702 monitor speaker. The device is designed to provide a tweeter configuration that disperses the sound in a broad pattern, allowing performers to move about freely. Frequency response has boosted midrange and controlled bass rolloff, cutting through ambient sounds on stage. The small ($15\frac{9}{16}$ x 21 x 10⅜ in.) unit may be used in either of two positions, at 60 degrees for distant throw, and at 30 degrees for close throw. The unit can be used with any amplifier capable of delivering up to 50 watts to a 16-ohm load. Included are parallel phone jacks for interconnecting 702s using only one cable from the amplifier.
Mfr: Shure Brothers Inc.
Price: $238.00.
Circle 62 on Reader Service Card



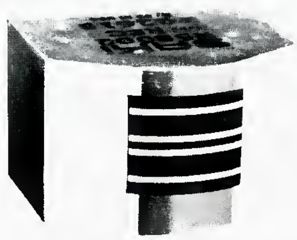
FREQUENCY RESPONSE RECORDER

• Measurement and graphic charts are achieved by Model LFR-5600 recorder for frequency response, wow & flutter, drift, voltage, and temperature parameters. The unit consists of two sections, an audio sweep oscillator and a pen recorder. The sweep oscillator may be used separately for direct frequency response readout on an oscilloscope, and the chart section can also serve as a direct current reader to 10mV/cm. Included among the features of the portable unit are automatic start circuitry, standard signal frequencies of 1 kHz and 333 Hz for reel-to-reel or cassette recorder check-outs; selectable 25 dB, 50 dB, or linear scales, 20 dB and 40 dB attenuation readings of the sweep oscillator, use of external signals for response checks, and a slow speed range of 1/10/cm which permits long-term drift measurements.
Mfr: Leader Instruments Corp.
Price: Under $3,000.
Circle 63 on Reader Service Card



REPLACEMENT HEADS

• Hot pressed glass bonded ferrite heads for Ampex AG 440 and 350 recorders are available in all track formats, compatible with existing electronics. It is claimed that heads of this construction, including glass bonded gaps, will outlast standard metal by 10 to 15 times.
Mfr: Saki Magnetics
Circle 64 on Reader Service Card



New Literature

PANEL METERS

A four-color 60 page catalog lists stock analog and digital panel meters, meter relays, controllers, and test instruments. Mfr: Simpson Electric Co., 853 Dundee Ave., Elgin, Ill. 60120.

MEASUREMENT & COMPUTATION

This leaflet includes information regarding a microprocessor, semiconductor memory, software libraries, and a programmable i.c. tester. Mfr: Hewlett-Packard, 1507 Page Mill Rd., Palo Alto, Ca. 94304.

PHONOGRAPH PREAMPLIFIERS

Reports by Tomlinson Holman, in extensive detail, of various factors involved in phonograph preamplifier design, reprinted on heavy paper in a durable booklet. Mfr: Apt Corp., Cambridge, Ma. 02139.

HARD-TO-FIND TOOLS

Another edition of this company's fat collection of tools. Mfr: Jensen Tools and Alloys, 1230 S. Priest Dr., Tempe, Az. 85281.

URETHANE FOAMS

This brochure describes industrial urethane foams. Mfr: Tenneco Chemicals, Park 80 Plaza West -1, Saddle Brook, N.J. 07662. (201) 646-3800.

NOISE ISOLATORS

Information sheets detail Kenetics Models SL and SM housed spring vibration isolators. Mfr: Peabody Noise Control Inc., P.O. Box 655, Dublin, Ohio 43017.

TEST EQUIPMENT

A 76-page catalog describes electronic test equipment from a number of well-known manufacturers. Mfr: Dept. AA 78, North American Electronics, 1468 W. 25th St., Cleveland, Ohio 44113.

VIDEO SUPPLIES

Over 500 video accessories are listed in "The Catalog of Video Supplies and Accessories"—tape, cassettes, labels, mics, cables, connectors, adaptors, printed video forms. Mfr: WIDL Video, 5325 N. Lincoln, Chicago, Ill. 60625.

Editorial

You probably don't need us to inform you that digital audio is "the new kid on the block." And from every indication, sooner or later this fast-growing kid will be taking over the whole block. Over the next few years, microphones and loudspeakers may not change much, but it's a sure bet that everything else will, thanks to digital technology. And that means we shall all have to do some serious studying, if we want to keep up with what's happening in audio.

Ronald Ajemian helps us on our way with **A Digital Logic Review for the Audio Engineer.** ANDs, ORs, NANDs and NORs are in our future, and Ajemian helps us figure out what these mysterious gadgets really do. In future issues of **db,** we shall try fitting them together, to get a better idea of their influence on the previously-analog world of audio.

What has digital audio got to do with recording studio construction? Plenty! In **Noise Level Limits in Recording Studios,** Michael Rettinger points out that quieter studios may be in order, if we wish to get the most out of digital technology. Rettinger examines the acoustician's NCs, PNCs and NRs, and finds them inadequate to the task of measuring studio noise levels. He proposes a simple alternative that may be more appropriate for studio specifications.

In reading R. Max Mayer's **Speech Privacy in the Open Office,** we discover that office privacy problems are closely related to studio separation problems. As in the studio, a variety of inter-related factors are juggled to achieve the desired result. As a further refinement, masking noise sound systems are introduced. While these may be just right for the office, could they be used with any success in the recording studio? Probably not, but as Edsel Murphy might have said, "When all else fails, . . ."

Although your favorite new microphone may be just right for acoustic guitar and bass trombone, don't try to use it to make sound level measurements, either in the studio or the open office. Sidney L. Silver guides us through the complexities of **Sound Measurement and Instrumentation Microphones,** and leads us to an all-too familiar conclusion; when making sound measurements, there is no such thing as the ideal microphone for all occasions.

And as for broadcast audio, the mail continues to arrive. For a somewhat different perspective, see Craig Barney's thoughts in this month's letters column.

RONALD AJEMIAN

Anatomy of Digital Logic

Open and closed switches initiate the logic behind the logic.

Most contemporary audio equipment employs digital logic, with many of the new audio equipment manufacturers standardizing on the use of positive logic. Positive logic uses the convention of a 1 to represent the "true," or more positive, level and a 0 to represent the "false," or less positive, level (0 volt). Therefore, we may say a 1 equals a HI state and a 0 equals a LO state.

Examples of states:

1 = HI (light bulb is on) = o→o closed switch

0 = LO (light bulb is off) = o↗o open switch

LOGIC DIAGRAMS AND TRUTH TABLES

Even the most complex digital audio system may be broken down into a series of relatively simple logic circuits, or gates, several of which are illustrated and explained below. A truth table is used with each logic diagram to show input and output conditions.

THE *AND* GATE

The first basic logic circuit is the AND gate. The AND gate can have two or more inputs and a single output. The output of the AND gate is HI only when all the inputs are HI. A LO state on any of the input leads will produce a LO state at the output. Thus the AND gate can be represented with the following circuit and truth table. (Figure 1).

The circuit diagram for this AND gate shows that a voltage (E) is delivered to light the bulb only when both switches are closed.

THE *OR* GATE

The second basic logic circuit is the OR gate. The OR gate contains two or more inputs and a single output. The output of the OR gate is HI when one or more inputs are HI. The circuit diagram for this OR gate (Figure 2) shows that a voltage (E) is delivered to light the bulb when either or both switches are closed.

THE INVERTER AMPLIFIER

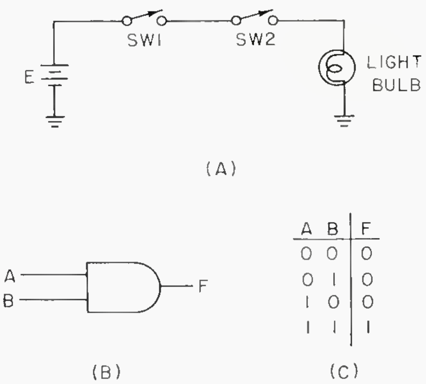
The inverter amplifier is a device with only one input and one output and hence inverts the input signal at its output. The logic diagram for the inverter amplifier can be drawn in Figure 3.

Logic diagrams make extensive use of the inverter indicator. The inverter indicator inverts the state and is drawn by a small circle at the input/output of the logic symbol. Without the small circle, the inverter would merely represent a non-inverting amplifier.

NAND and *NOR* GATES

The NAND gate (Figure 4) is simply an AND gate followed by an inverter indicator.

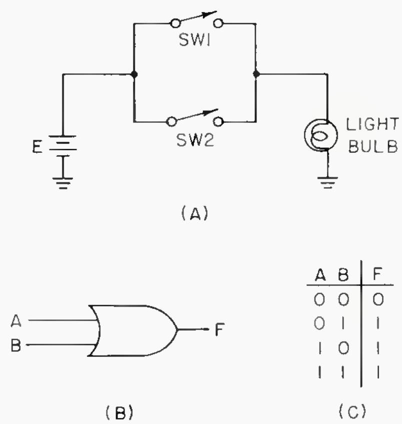
The NOR gate (Figure 5) is simply an OR gate followed by an inverter indicator. The symbol indicates this by being formed from the OR symbol plus the small circle at the output.



A	B	F
0	0	0
0	1	0
1	0	0
1	1	1

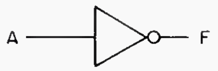
Figure 1. The AND Gate. (A) Circuit diagram. (B) AND Gate logic symbol. (C) Truth table for AND Gate.

Figure 2. The OR Gate. (A) Circuit diagram. (B) OR Gate logic symbol. (C) Truth table for OR Gate.



A	B	F
0	0	0
0	1	1
1	0	1
1	1	1

Ronald Ajemian is an engineer with the N.Y. Telephone Company.

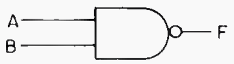


(A)

A	F
0	1
1	0

(B)

Figure 3. The inverter amplifier. (A) Inverter amplifier logic symbol. (B) Truth table for the inverter.

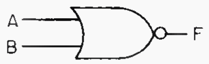


(A)

A	B	F
0	0	1
0	1	1
1	0	1
1	1	0

(B)

Figure 4. The NAND Gate. (A) NAND gate logic symbol. (B) Truth table for NAND gate.

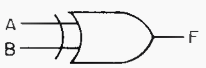


(A)

A	B	F
0	0	1
0	1	0
1	0	0
1	1	0

(B)

Figure 5. The NOR gate. (A) NOR gate logic symbol. (B) Truth table for NOR gate.



(A)

A	B	F
0	0	0
0	1	1
1	0	1
1	1	0

(B)

Figure 6. The Exclusive-OR Gate. (A) Exclusive-OR gate logic symbol. (B) Truth table for Exclusive-OR Gate.

THE EXCLUSIVE—*OR* GATE

The Exclusive-OR gate (FIGURE 6) produces a HI output when the input states are not identical. The output is LO when the inputs are identical, regardless of whether these inputs are all HI or all LO.

FLIP-FLOPS

The Flip-Flop performs the logical operation of storage and is a bistable device with one or more inputs. A Flip-Flop circuit will remain in its last state until an input causes it to change its output state. Because of this ability to "store" a bit of information, the Flip-Flop is the basic building block in digital logic. The state of the Flip-Flop is available on the output line. Almost all Flip-Flops employ a second output line on which the complement (i.e., the opposite state) of the stored function is available. The other terminals of a Flip-Flop are input terminals and may receive either level or pulse signals, depending on the particular circuit.

The basic Flip-Flop can be drawn (FIGURE 7) using two NAND gates. To understand how the Flip-Flop works, assume the two inputs R and S do not exist. When power is applied, opposite states will appear on the outputs.

For example, assume that the output of Gate X is LO. This LO will be applied to input A of Gate Y, whose output will then become HI. When this HI is applied to input B of Gate X, a LO will remain on the output of Gate X. Thus, the gates are "latched" into a stable state.

Next, connect R and assume it is HI. Thus, the output of Gate X remains LO. The state of the gates can be changed by applying a LO to input R. This LO causes the output of Gate X to go to HI, and therefore the Y

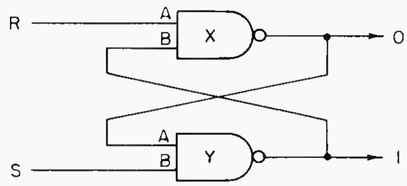


Figure 7. Basic Flip—Flop, using two NAND gates.

gate output goes LO. (NAND gate—if both or either input is LO, the output is HI.) Since input B of Gate X is now LO (from the Gate Y output), there is no way to return to the original state through the use of R. This LO input will keep the output of Gate X HI, regardless of the R input state. But now, by connecting S, the state of the Flip-Flop can be changed by applying a LO to the input (R or S) of the gate whose output is LO and the Flip-Flop is fully controlled. It is assumed that the input (S or R) to the opposite gate is HI. (If both the R and S inputs are LO, the Flip-Flop's output states will be indeterminate.) Thus, a basic Flip-Flop consists of two NAND gates with outputs cross-coupled to the inputs.

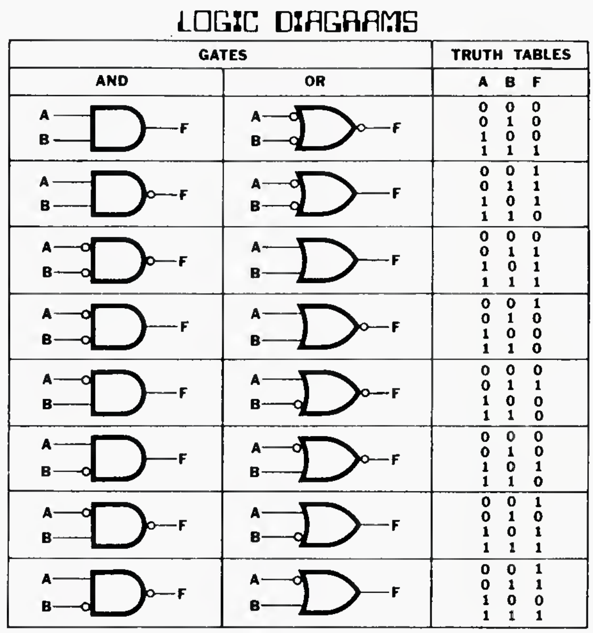


Figure 8. Logic diagrams chart.

I have provided a useful chart (FIGURE 8) on logic gates, with respective truth tables. This chart can be used for substitutions of equivalent logic gates. ■

R. MAX MAYER

Speech Privacy in the Open Office

A careful intermix of architectural and electronic masking devices keep interfering sounds out of work zones.

IN THE INTERESTS of economy, as well as flexibility, the open-office plan has become popular. Not only are portable walls or screens cheaper to construct than floor-to-ceiling walls, but it becomes possible to deploy workers in small or large groups, as needed. For example, portable walls are used in the Federal Government's New York H.U.D. offices, where there are times when the core office staff is greatly augmented by the presence of field appraisers on special assignment and larger work areas are needed temporarily.

However, along with the open-office plan, there comes the problem of conversation and other noises penetrating from one work zone to another. This is not only distracting, but interferes with privacy.

An approach to this problem has been dubbed *Speech Privacy Potential* (*SPP*). This involves the consultation of an acoustical expert, who combines his knowledge of the nature of sound and its movement with the architectural particulars of the work space and the needs of the workers. All interzone voice paths are studied until a work zone's maximum voice isolation is achieved.

A criterion for voice signals is known as *Noise Isolation Criteria Prime* (NIC). Definition of a standard NIC curve states that a minimum NIC of voice isolation must be attained between work zones through an acoustical ceiling/screen combination. Achieving acoustical privacy requires incremental adjustments of the sound-masking spectrum within individual one-third octave bandwidths.

Ideally, the final spectrum adjustment of the system should bracket the full range of each individual's voice level. However, inasmuch as some people are very loud, this cannot be realistically expected. The practical aim is to mask only conversation at the level of average male speech patterns, based on the required NIC and NC

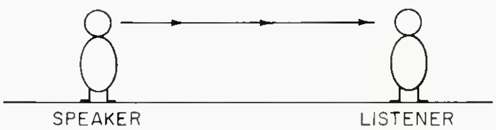


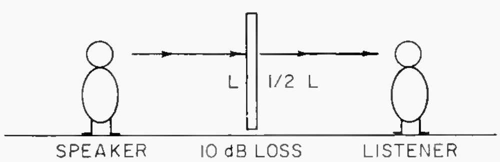
Figure 1. Sound travelling on a direct path can be easily understood by a listener 8 to 12 feet away.

(Noise Criteria) curves. These two curves are derived from years of physical and psychological testing by independent acoustic labs across the nation.

The acoustics expert juggles a number of factors in creating a viable atmosphere. He takes into consideration the nature of sound and its pathways. Then he works with architectural devices, such as acoustical tiles and separating screens, understanding their limitations as well as their value. Finally, he must consider not only the physical properties of the human ear but the psychology of hearing, which causes us to pay attention to certain sounds over others.

All of these elements must mate, complementing one another, to create a totally efficient system. None of the approaches can do the job alone.

Figure 2. A vertical barrier (screen) reduces by at least one-half loudness the transmission of sound.



R. Max Mayer is the manager of field services for the Dukane Corporation, of St. Charles, Ill.

LOCALIZATION OF SOUND (NIC CONCEPT)

Office noises, including the sounds of typewriters and telephones, as well as conversation, permeate the atmosphere, traveling over, around, through, and across obstacles, bouncing to the ceiling and the space above the ceiling, the plenum, reflecting from walls and furniture. Even an almost acoustically dead room will reveal certain noises, such as the hum of air-conditioning. However, in the typical room, air handling sounds are usually drowned out by louder noises.

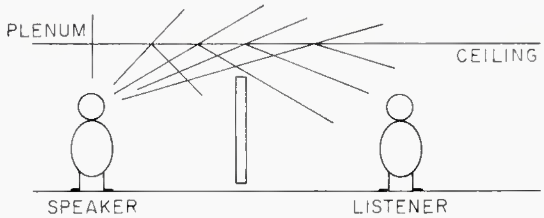


Figure 4. Voice signals pass through the plenum and at angles of 30-60 degrees; the skipping effect comes off the ceiling board surface.

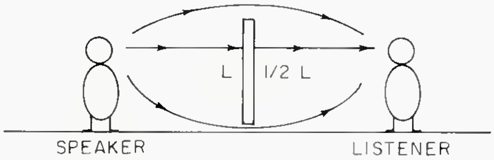


Figure 3. Sounds sneak above and below screens.

The way we hear is tied into a process known as the *localization* of sound, which isolates a particular noise. To completely understand this, imagine yourself standing on a train station platform, engaged in conversation with another person. The voice of your companion is perfectly clear—until the train roars into the station. Then all you hear is the sound of the train.

The train noise has been localized, replacing the sound of your companion's voice; your ears are paying attention to the loudest and most immediate sound present. What we are interested in doing in a work area is localizing desirable sounds, keeping communication between workers clear, and at the same time soft-pedalling distracting sounds.

According to acoustic principles, the loudest noise heard is that which travels in a straight line. The sound loses approximately 6 dB of power every time the distance is doubled. Sound travelling on a direct path from a speaker will be loud enough, on the average, to be easily understood by a listener eight to twelve feet away—typical of office spacing. Therefore, what we are interested in doing is containing the direct-path communication between workers in an enclosed area while blanking out, as much as possible, sounds coming from outside the work area.

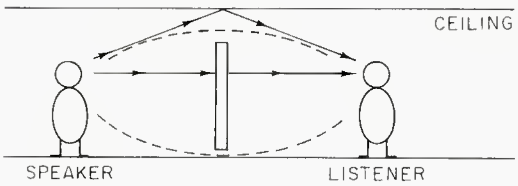


Figure 5. After the ear localizes upon the loudest signal, the other waveforms fill in the missing frequencies and makes the conversation in the adjacent work zone understandable.

SCREENS

The first obvious devices used in an open-office situation are screens, providing the portable walls needed to set off various work areas. At the same time, screens impede the direct-line passage of sound, reducing by at least a one-half loudness factor the transmission of sound between the source and the listener. Normally, a screen is padded, septumed, and braced to reduce the amount of signal bouncing off its surfaces. Often of attractive design or providing space for decorations, screens also add to the total human environment.

The screen's design should minimize any refraction (bending) of sound over its surface or reflection underneath its lowest edge, where the floor's direct reflective field is a serious consideration. Therefore, a sound seal at the floor junction is a necessity to stop noise which could filter underneath the screen. The refractive properties of wood or metal should mitigate against the use of screens with this type of edge material.

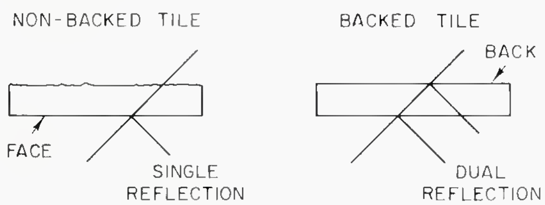
Carpeting is of some help in eliminating the reflective/refractive properties of screens and partitions. However, although carpeting deadens the room acoustics for impact sound sources, it does not affect voice levels. Speech frequency levels are not attenuated to any degree by the carpeting on an eight-foot wide interzone path.

CEILING REFLECTION

We have said that sound travels in a straight line and that screens frustrate this path. But where do the sound waves go when they are deflected by a vertical surface? Up to the ceiling, of course, at a 90 degree angle. If the ceiling is made of a hard, untreated surface, they bounce right off and drop into the next work area. The plenum (the area above the ceiling) also can reflect voice signals down into other work zones, providing another localization path. If the offices are quiet, people in adjacent cubicles will distinctly localize and recognize voice signals.

When acoustical tiles are used, they "grab" the sounds, permitting them to seep through into the plenum area. The sound waves skip along the plenum chamber and drop down through the tiles at a diffused 30 to 60 degrees, which creates a vague mumble instead of distinct conversation. The inquisitive ears of the listeners, unable to make sense out of the mumble, lose interest and psychologically turn off the sound; it is no longer localized.

Figure 6. Backed tile diffuses reflection.



The efficiency of ceiling tiles varies. At times, compromises must be made because of cost. For example, one-inch backed ceiling board will transmit less signal into the plenum space and therefore more reflected signal into an adjacent work zone than thicker tile. But one-inch backed ceiling board may be considered desirable because it is economical. This is the type of tradeoff which often takes place in the extensive consideration of an open-office planning technique.

MASKING SOUND

Although screens and ceiling tiles, as well as the careful arrangement of furniture, installation of carpeting and draperies, etc. do a good deal to mitigate distracting noise, there is still some sound remaining. A new ploy to overcome this is the masking sound, electronically produced, which localizes in the listeners' ears and dims in the listeners' consciousness the noise of the office.

The masking sound, it must be understood, is used in conjunction with architectural devices. The variants of acoustical modification must work together to produce the desired effect.

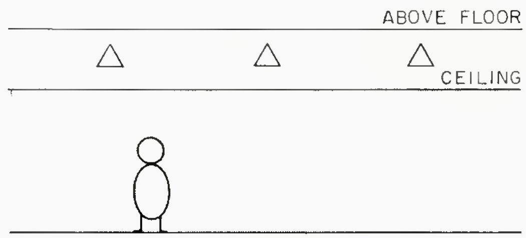


Figure 7. An induced masking signal showers down from the plenum and permeates the atmosphere.

The electronic sound loudspeakers direct the sound through the plenum area, where it skips along just as the conversational signals do, and comes down in a diffused manner into the atmosphere. Care is taken to obtain a volume which will be sufficiently loud to overcome other sounds and yet not intrusive. The type of sound introduced is pleasant and soothing, like the soughing of the sea upon the shore. The wideband sound easily masks out the leftover interzone voice levels. Combined with low screens, equal coverage results can be achieved throughout the entire floor plan.

Typical electronic devices used to produce voice masking include one-third octave filter sets, and sound-masking speaker baffles. The filter set should provide accurate, individual one-third octave adjustments of all frequency bands. Therefore, the sound spectrum in a room may be adjusted to fit the optimum SPL curve within the specific open-office environment.

The masking baffle causes the electronic sound to "shower" through the ceiling board from the plenum area, providing a uniform unobtrusive sound.

SUMMARY

Privacy and sound isolation in an open-office arrangement are achieved through a combination of architectural modifications and masking sound. Efficiently planned screens, carpeting, furniture arrangement, and ceiling tiles mute the sound sufficiently so that masking may be introduced without being intrusive. To complete the circle, electronic masking sound permits the creation of a pleasant human environment in an office plan free of inflexible, expensive floor-to-ceiling walls between work zones. ■

MICHAEL RETTINGER

Noise Level Limits in Recording Studios

Digital recording requires a reconsideration of studio noise levels.

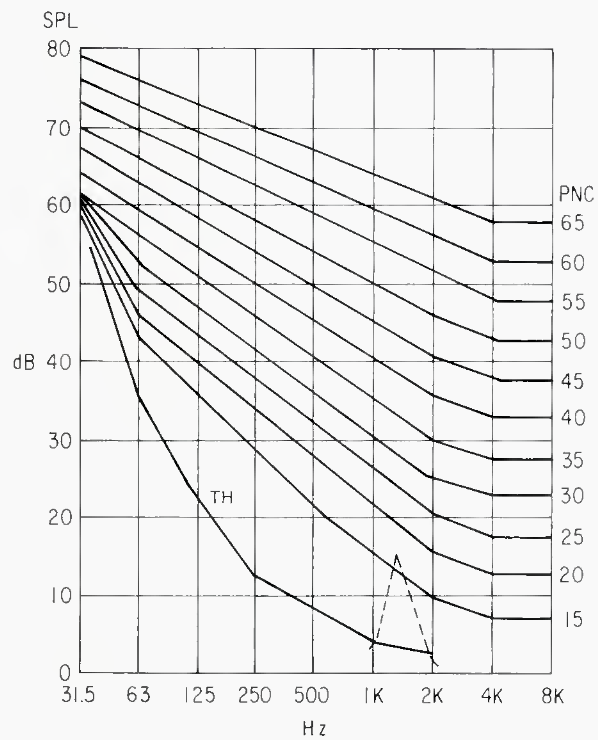


Figure 1. Preferred Noise Criteria (PNC) curves. Each curve is a code for specifying the noise level characteristic, or spectrum, of the noise in rooms. TH is the threshold of hearing for continuous noise fields. SPL is sound pressure level in decibels relative to 0.0002 microbars.

WITH THE ADVENT of digital recording on magnetic tape, new vistas in dynamic range and signal-to-noise ratios are opening up for both verbal and musical entertainment programs. In the recording studio, these improvements lead us toward a reconsideration of those maximum acceptable noise levels created by disturbances such as air-conditioning system rumble, transmission of automobile or aircraft din through the walls and roof of a building, or the electronic hiss generated in microphones and in the first stage of their associated preamplifiers.

Two questions immediately present themselves: how should such limits be expressed and what are the minimum signal levels that are to be recorded?

NOISE CRITERIA

In the past, acceptable noise level limits in enclosures were specified either as a broadband single-figure value, like 30, 25, 20, or 15 dBA, or as a Noise Criterion rating, like the NC (Noise Criterion) first proposed by Leo L. Beranek in 1967; PNC (Preferred Noise Criteria), developed by Beranek in 1971; or NR (Noise Rating, employed in Europe). None of these noise criteria pertain specifically to recording studios, but apply either to offices or to homes.

FIGURE 1 shows the series of PNC curves, together with the threshold of hearing for normal (young) people. Each curve is a code for specifying the noise level characteristic, or spectrum of the maximum acceptable noise in the room under consideration.

There is a point to be made about such ratings which is sometimes ignored. Sketched in on FIGURE 1 in dashed lines is the graphic representation of a noise which would not be found acceptable if a PNC-15 rating had been specified for a room because it penetrates the PNC-15 curve. On the other hand, a noise characteristic which almost duplicates the PNC-15 graph—only 0.5 dB below it—would be found acceptable by this system of assessing room noise. Yet it should be obvious that the former noise is more tolerable than the latter, if only because its broadband A-weighted sound level is 10 dB lower.

The table below gives the A-weighted sound levels for various PNC, NC and NR ratings. Note the discrepancies between the various rating systems. For example, a sound level of 70 dB SPL corresponds to a PNC or NR rating of 65 and an NC rating of 60. Also note that a rating of 45 corresponds to three different SPLs, depending on the rating system used.

A-WEIGHTED SOUND LEVELS

PNC	A-Level	NC	A-Level	NR	A-Level
65	70	65	75	65	70
60	65.5	60	70	60	66
55	61	55	65	55	62
50	57	50	60	50	58
45	52.5	45	55	45	54
40	48	40	50	40	50
35	43.5	35	46	35	46
30	39	30	42	30	42
25	35	25	38	25	38
20	30.5	20	33	20	34
15	26	15	28	15	30

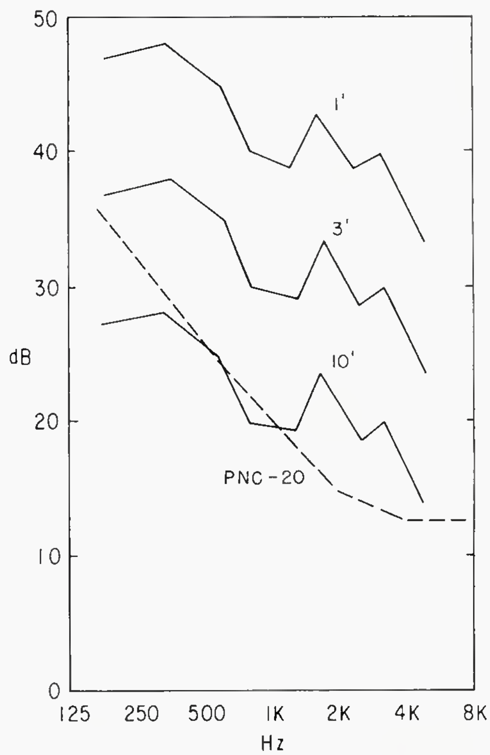


Figure 3. Decile speech peak pressure characteristics exceeded 90% of the time (composite of the voices of five women) in one-eighth-second intervals, at 1', 3', and 10' from the mouth of the speaker (calculated from information given in "Speech and Hearing in Communication" by Harvey Fletcher).

SPEECH LEVELS

Before one can answer the question of how to specify noise levels in a recording studio, he must consider the minimum signal levels to be recorded, and the characteristics of English speech. FIGURE 2 shows that the dynamic range for speech is on the order of 30 dB and that the mean level is 12 dB below the peak level, with minimum speech levels at 18 dB below the mean level.

Further information about speech can be found in Harvey Fletcher's book *Speech and Hearing in Communication.* In it he published the decile (one of the values of a variable which divides the distribution of the variable into ten groups having equal frequencies) speech levels for men and women in one-eighth-second intervals for both peak and mean levels. In acoustics, a decile sound level refers to all the values about the stated figure, so that, for instance, a decile sound level of L_{90} is that wideband or octave-band level which is exceeded 90 per cent of the test period. While Fletcher listed the results of his tests in sound levels which employed 1 dyne/cm^2 as the zero dB reference level, here we will employ the more common reference level of 0.0002 microbars or micropascals (f = 0.0002 dynes/cm^2). Thus a constant sound level of 0 dB referred to 1 dyne/square centimeter, would be defined as a sound level of 74 dB when referred to 0.0002 microbars.

Figure 2. Characteristics of English speech.

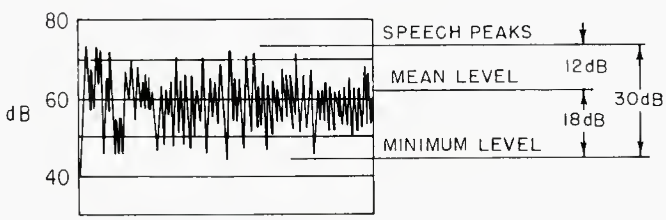


FIGURE 3 shows the decile speech peak pressure level characteristics exceeded 90 per cent of the time (composite of the voices of five women) at 1 foot, 3 feet, and 10 feet from the mouth of the speaker. Also shown is the PNC-20 rating sometimes called for in building specifications for recording studios. Note that the curves pertain to speech maxima, which are 30 dB above speech minima.

In FIGURE 4 we again see the L_{90} levels for speech peaks at 1 foot and at 3 feet, labelled P-1' and P-3' respectively (top of figure); below it the NC-15 and PNC-15 graphs and below them, the L_{90} levels for speech minima at 1 foot and at 3 feet from the mouth of the speaker. Also shown on the figure is the threshold of hearing curve.

It is seen that neither the PNC-15 nor NC-15 rating is adequate to specify the noise in a room intended for

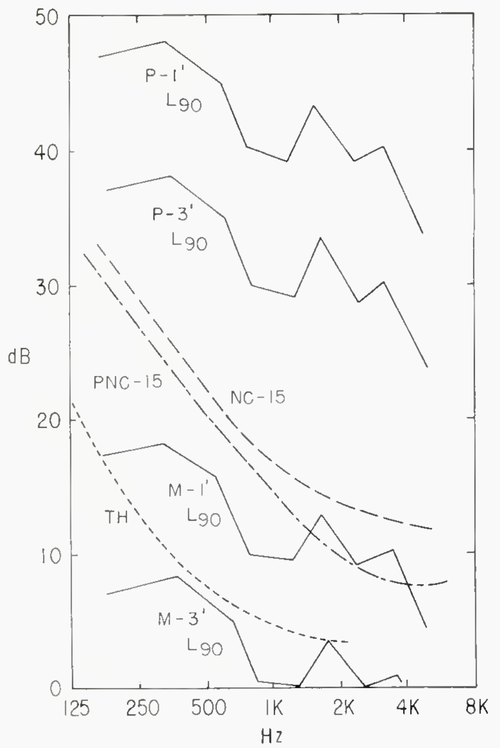


Figure 4. Speech peak level characteristics exceeded 90% of the test period at 1 foot and at 3 feet from the speaker, identified as P-1' and P-3' respectively; speech minimum level characteristics exceeded 90% of the time, at 1 foot and at 3 feet, labelled M-1' and M-3' respectively; NC-15 and PNC-15 graphs, and threshold of hearing curve, marked TH

sound recording, because the speech minima at both 1 foot and at 3 feet from the speaker fall below these two specified criterion curves.

Nor should A-weighted sound levels be employed to specify maximum noise levels for recording studios, since low frequencies are attenuated too much by the A network in the common sound level meter (16 dB down at 125 hertz).

No ratings below PNC-15 and NC-15 are available in the literature, and even if they were available, would be subject to the drawback outlined previously. Therefore, it is suggested that a B-weighted sound level, like 23 dB-B, be adopted for specifying the noise level in a new studio. In the B network, 125 hertz is attenuated only 4 dB, and noise measuring equipment of this sensitivity is readily available. (General Radio's Sound Level Meter 1551-C, Bruhl & Kjaer's 2203, etc.)

In respect to microphone noise, Dr. Harry F. Olson states that the principal sources are thermal agitation disturbances generated in the conductors and the polarizing and biasing resistors, if any, contained in the transducer. He reports that, for a high-quality microphone, the thermal agitation noise in the audio frequency range is of the order of an equivalent sound pressure level of 10 dB, which is 10 dB below the ambient noise of a "very quiet" studio.

In respect to preamplifier noise level limits, a figure of —120 dBm is often quoted for a high-quality unit, which is equivalent to 0.77 microvolts across 600 ohms.

DIRECT-TO-DISC

Digital recording on magnetic tape is not the only system which can benefit by a very low noise level in the studio. Direct-to-disc systems also derive advantages in an improved signal-to-noise ratio. There is experimentation going on to achieve an optical digital track on 35mm film by means of a light valve which opens and closes rapidly and which can also accommodate four tracks on a one-eighth-inch wide track, because crosstalk between adjoining digital tracks is very low for the hi-fi enthusiast.

CONCLUSION

Digital recording technology will offer the possibility of recording very wide dynamic range programs. To take full advantage of the potential of digital recording, acceptable noise level criteria for studio construction may have to be revised. A suitable measurement system will take into account broadband and low-frequency noise levels, both of which may become more apparent due to the improved performance characteristics of digital tape recording. B-weighted sound level measurements may be more appropriate than the various noise criterion rating systems now in use. ■

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Rettinger, M. *Acoustic Design and Noise Control.* Chemical Publishing Co. 1977.

SIDNEY L. SILVER

Sound Measurement and Instrumentation Microphones

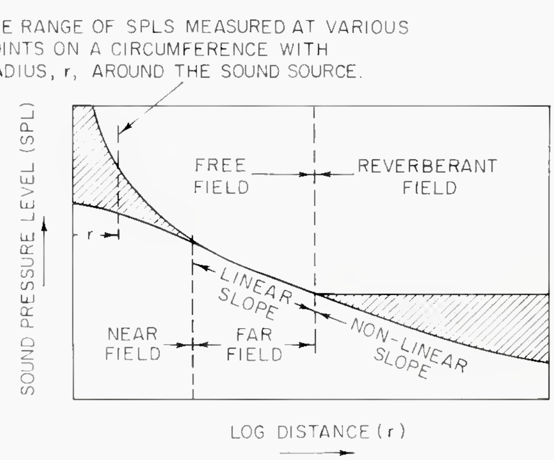
Microphones used for measurement purposes must be selected according to field conditions and sensitivity requirements.

Basically, there are two acoustical environments that may be applied in sound measurements—the acoustical free field and the acoustical diffuse field. In the free-field environment, all the sound waves arriving at the point of measurement are radiated directly from the source, and the effects of reflecting obstacles on the sound propagation are insignificant. A simple acoustical environment that closely approximates actual operating conditions for many types of sound sources is attainable by placing the source on a hard, smooth surface with no disturbing sound-reflecting objects in the vicinity. This is not always possible to achieve in practice, but nearly ideal free-field conditions can be created in an anechoic chamber, where the boundaries absorb practically all of the incident sound energy. Such an environment can also be provided by flat, open, outdoor areas devoid of large reflecting surfaces, such as buildings or trees.

In the diffuse-field environment the sound source is mounted in a large reverberation chamber, producing a sound field in which the time-averaged flow of sound energy is essentially uniform throughout the field. Here reflected sound waves from all possible directions are just as likely to arrive at the microphone with the same intensity, and the phase of the sound waves random at the microphone position. Such a field may also be approximated by a room with highly irregular walls and reflecting objects of various sizes and shapes within it to break up the standing wave patterns. It should be pointed out that diffusion does not diminish the total sound energy in the enclosure, but rather tends to increase the number of reflections occurring per unit time, hence lessening the intensity level of *individual* reflections.

As previously stated, both free-field and diffuse-field conditions relate to the properties of the surrounding environment. But it is also important to consider the behavior of the sound source radiation as a function of distance. Figure 1 illustrates the variation of sound pressure

Figure 1. Variations in sound pressure level (SPL) as a function of distance (r) between a microphone and a sound source.



Sidney L. Silver is on the supervisory staff of the Telecommunications Section of the United Nations, where he is in charge of sound and recording.

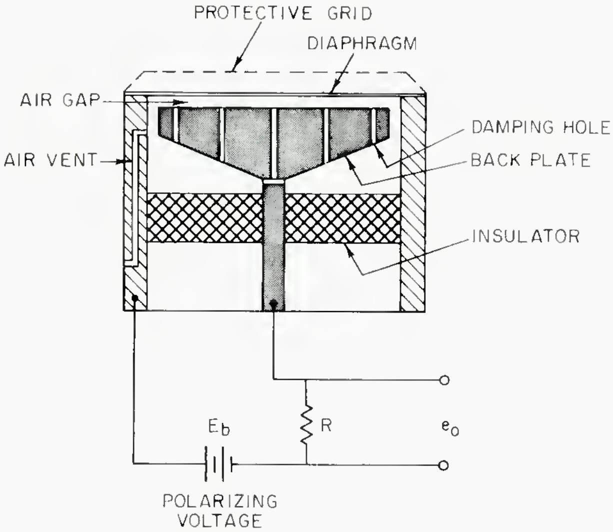


Figure 2. Schematic view of an air-condenser microphone.

between a typical source and a microphone. The region designated as the near field (within the limits shown by the shaded area) is characterized by an appreciable change in sound pressure when the microphone is moved to another nearby point at the same distance from the source. These variations are caused by the fact that the microphone responds not only to the predominant generating source, but also to phase cancellations and additions from other radiating sources in the vicinity. Probe microphones are sometimes used at close proximity in the near field to isolate sources of noise, but most sound measurements are made in the far field.

FAR-FIELD

In the far-field region (between the near field and the reverberant field), the same measurement is obtained anywhere around the path of sound propagation; the sound energy is distributed over a larger space and the air-particle velocity is in phase with the sound pressure. Under these conditions, the sound pressure level decreases at the rate of 6 dB for each doubling of the distance from the source. This relationship holds true if the source is in free space and the absorption of the room is sufficiently great, so that the reverberant field has not yet been reached. Stated another way, it is possible to satisfy free-field conditions if the absorption is not so small that the near field and reverberant field merge. In general, the near-field/far-field border is reached when the microphone distance is at least three to four diameters greater than the largest dimensions of the radiating source.

Referring to FIGURE 1, the shaded area in the reverberent field covers the limits of sound pressure fluctuations measured with small shifts in microphone position. At the far-field/reverberent-field border, the reduction of sound pressure with distance becomes nonlinear because the reflected sound energy predominates over the direct sound energy originating from the source. The microphone picks up these sounds as separate sources, thereby affecting the measurement data received. Because of these errors, the most suitable transducer must be selected and oriented in the sound field to minimize these effects, especially when making highly precise measurements.

SELECTING THE PROPER MICROPHONE

The technical literature on high-quality microphones for recording speech and music is fairly extensive, but very little information is available on microphones specifically designed for acoustic measuring purposes. However, accurate and precise sound measurements require a working knowledge of measuring microphones, if sound-level meters and other associated equipment are to be applied effectively. In order to interpret measurement results correctly, it is essential to understand the capabilities and limitations of the particular microphone utilized.

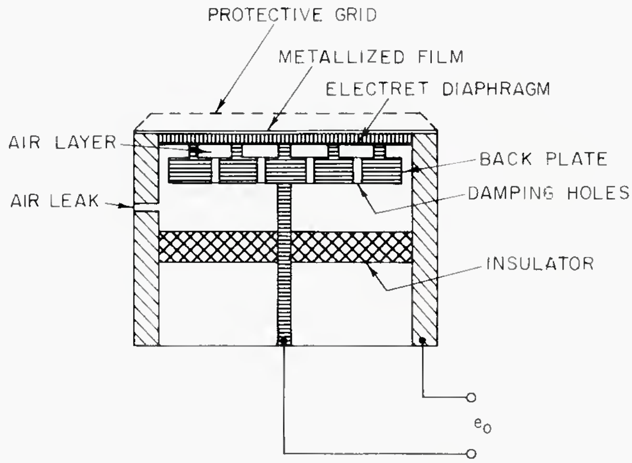
Ideally, the basic requirements of an instrumentation microphone are that it have an absolutely stable sensitivity with well-defined operating characteristics. Its structural dimensions should be small compared with the shortest wavelength of interest, so that interference and diffraction effects are negligible. Its frequency response should be flat within the desired audio frequency range without introducing phase shifts between the applied sound pressure and the electrical output. Moreover, its performance should be independent of environmental conditions, so that neither extremes, nor changes in temperature, humidity, and vibration will have a detrimental effect on the microphone or its calibration.

In practice, no single microphone can satisfy all these requirements, and certain compromises have to be made in the selection of a suitable transducer for a precision measurement problem. With regard to frequency response, directional characteristics, and dynamic range, there are conflicting aspects to be considered which relate to the geometrical dimensions of the microphone. For example, directional effects increase with frequency, (that is, with diminishing wavelength) so that a very small transducer is necessary for high-frequency measurements. But, the smaller the physical size of the microphone, the lower is its sensitivity, so only high-intensity sound can be measured accurately. Where sensitivity is of primary importance, it may be optimized at the expense of long-term stability in order to measure very low sound levels.

CONDENSER VS CERAMIC MICROPHONES

Of the various types of measuring microphones in common use today, those supplied with current models of sound-level meters are the air-condenser, electret-condenser, and ceramic types. Others, such as the dynamic types, are of limited use in precision measurement work because they are relatively less stable and lack the frequency-response control necessary to obtain accurate results. Generally, the air-condenser microphone is capable of the highest performance with regard to long-term

Figure 3. Schematic representation of an electret-condenser microphone.



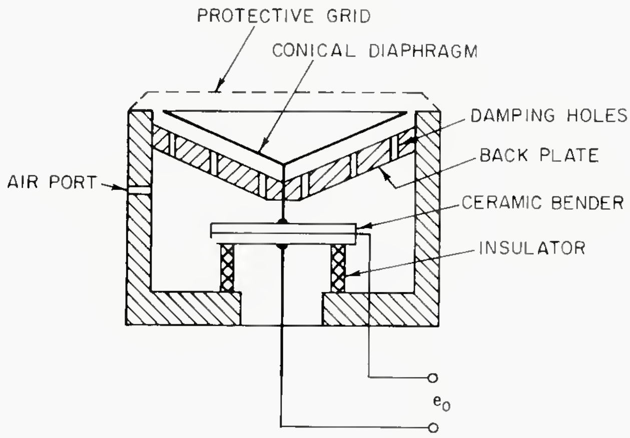


Figure 4. Schematic construction of a ceramic microphone.

stability, flat response over a broad frequency range, and wide availability in various sizes.

Basically, an air-condenser microphone consists of a tightly-stretched metal diaphragm, and a rigid back plate electrically insulated from the diaphragm, to form the electrodes of a capacitance displacement transducer. As shown in FIGURE 2, the back plate is perforated with properly designed damping holes to control the resonant peak of the diaphragm response. The peak frequency is determined by the diaphragm mass, while the damping effect is a function of the air flow between the electrodes. Motion of the diaphragm by the applied sound pressure produces an air stream through the damping holes with resulting friction and energy dissipation, the system behaving essentially like a simple mass-spring dashpot for frequencies below its natural resonance.

Because changes in ambient pressure as a function of time are considerably greater than the small pressure variations due to sound waves, a tiny capillary hole provides an air leakage path through the housing to the atmosphere. The air leak is small enough to allow only long-term pressure equalization, so that the static pressure within the microphone capsule always equals the ambient pressure. By maintaining the same acoustical loading inside and outside the microphone, a well-defined limit will be imposed on the low-frequency response. If, for example, the air vent has an acoustical time constant of, say, 0.03 secs, this will correspond to a 3 dB cut-off at approximately 5 Hz. However, for the measurement of extremely low frequencies in the infrasonic range, the air vent may be completely sealed to bring the low limiting frequency down to a fraction of one hertz.

In operation, displacement of the diaphragm caused by impinging sound waves changes the spacing of the air dielectric, and hence the effective capacitance. By charging the capacitance with a d.c. polarizing voltage (on the order of 200V), the sound pressure variations are transformed into corresponding voltage signals. Here the polarizing voltage not only acts as circuit excitation, but also determines the zero-pressure diaphragm position due to the electrostatic force between the capacitance plates. The variable capacitance of the microphone is connected in a simple series circuit with a high resistance and the polarizing source, the charging time constant being long enough to maintain a constant charge on the microphone capacitance. For stationary pressure differences across the diaphragm, there is no current flow and the output will be zero, but for dynamic pressure differences, a current will flow through the resistor and an output signal will be developed.

ELECTRET

Another transducer of the condenser type is the electret microphone, which utilizes a flexible plastic diaphragm conductively-coated on the outside surface. The electret material is actually a self-polarizing polymer film which has a permanent charge embedded within it, so that an external biasing source is not required. As shown in FIGURE 3, the nonmetallized surface is placed next to a stationary back plate, to form an electret capacitance. Across the surface of the back plate are a large number of protrusions which make contact with the electret material, leaving a narrow air gap between the raised surfaces. These protrusions correspond to sensory cells which effectively operate in parallel to generate the composite output voltage of the microphone. The perforated back plate enables the air layer to communicate with the larger cavity within the microphone, increasing the sound pressure amplitude, and hence, the acoustic sensitivity of the system.

An important advantage of the electret microphone is that a strong bond exists between the charged particles and the molecules of the elecret material. Since there is no free electrostatic charge at the surface of the diaphragm, the microphone is relatively insensitive to noise in humid environments. Also, because the plated surface of the diaphragm is electrically insulated from the back plate by the electret material, the diaphragm can be rigidly supported by the backplate with proper tensioning, thus giving the microphone a low vibration sensitivity rating.

Ceramic microphones depend for operation on the fact that an electric voltage is generated when certain ceramic substances having piezo-electric properties are subjected to mechanical forces or stresses. The ceramic material serves as a sensing element (FIGURE 4), which is mechanically coupled to a conically-shaped diaphragm fabricated of a thin metallic foil. Here the diaphragm responds to sound pressure variations by transmitting corresponding compressive forces to the ceramic element, thereby generating an electrical potential proportional to the applied sound pressure. The perforated backplate mounted in close proximity to the diaphragm acts as an absorbing device (together with the shallow air layer in between) to dampen the natural resonances of the system.

Although ceramic microphones are mechanically rugged and relatively insensitive to temperature and humidity environments, they are characterized by low acoustic sensitivity. Therefore, special precautions must be taken to avoid the generation of spurious electrical signals caused by vibrations. Also the resonant frequency of the diaphragm system is lower than that of the air-condenser microphone, giving this unit a comparatively limited high-frequency response.

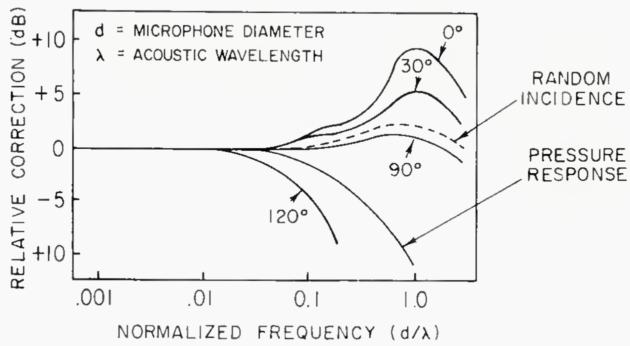


Figure 5. Typical family of free-field correction curves to be applied to a pressure curve.

FREQUENCY-RESPONSE CHARACTERISTICS

For acoustic measurements, microphones are designed to have either a flat pressure characteristic, a flat free-field characteristic, or a flat random-incidence (as in a rev. field) characteristic.

Pressure response is the term used to express the ratio of the open-circuit voltage of a microphone to the applied sound pressure, when sound is incident only on the surface of the microphone diaphragm. It can easly be measured by a number of accepted methods, including the use of an electrostatic actuator to simulate a sound pressure equally distributed across the diaphragm, or an acoustic coupler of such shape and dimensions as to make it possible to establish a uniform sound field at the diaphragm.

What is usually desired, however, is the *free-field response,* i.e., the relationship between the microphone's output voltage and the sound pressure that existed at the microphone's measuring position in the sound field *before* the microphone was placed there. The output voltage is produced when the microphone is introduced in the sound field at a specific orientation with respect to the angle of incidence. Sound waves impinging on the diaphragm will give rise to complex reflection and diffraction effects, causing an increase in the effective sound pressure. This rise in amplitude depends upon the wavelength of the sound, the microphone size and shape, and the direction of sound propagation. The reason that the presence of the microphone disturbs the sound field is that its acoustic impedance is substantially different from that of the air medium surrounding it. At low frequencies, the sound wavelength is very large compared with the mechanical dimensions of the microphone. The effects of reflection and diffraction are negligible, free-field response is the same as pressure response. But at high frequencies, where the acoustic wavelength is comparable to the microphone dimensions, the sound pressure at the diaphragm for frontal waves is considerably higher than it would be if the microphone were absent.

For a given microphone, there is a useful relationship between the pressure response and free-field response known as the *free-field corrections.* These corrections represent the increase of sound pressure caused by high-frequency interference, and have been determined for a particular type of microphone by previous measurements. FIGURE 5 shows a family of free-field correction curves for a typical one-inch microphone measured at various angles of incidence. For the sake of comparison, the pressure response is also shown. The frequency axis is normalized to show the ratio of the frontal diameter of the microphone to the sound wavelength ($f = d/\lambda$). Note that maximum correction occurs when the value of this ratio is unity. Since the wavelength in inches of sound waves in air at normal temperatures is roughly 13,000/f inches, where f is the frequency in hertz, this maximum correction is reached at about 13000 Hz for a one-inch microphone. The random-incidence curve, shown by the dashed line, represents the small correction integrated across all angles of incidence and averaged. In order to derive the free-field response of the microphone, it is only necessary to add the measured pressure response to the appropriate

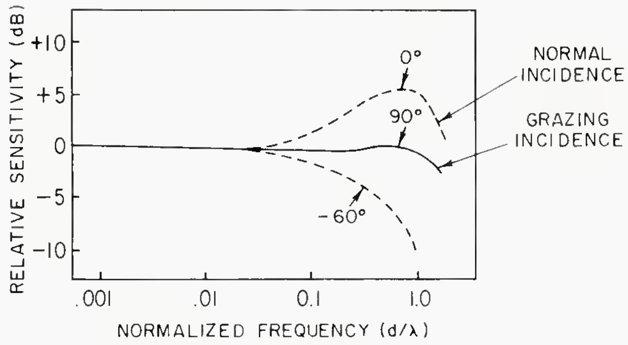


Figure 6. Parallel (ninety degree) incidence and off-axis frequency response characteristics of a typical pressure microphone.

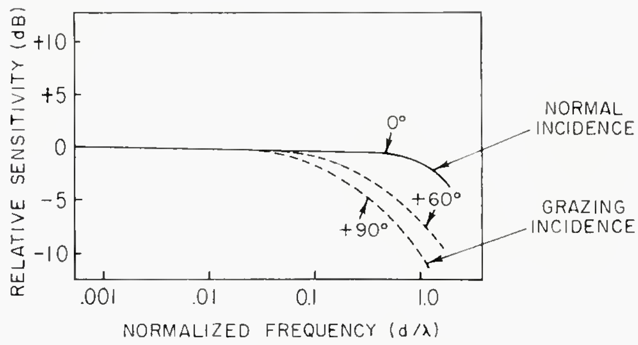


Figure 7. Perpendicular (zero degree) incidence and off-axis frequency response characteristics of a typical free-field microphone.

free-field correction curve. The resultant curve selected is based on the angle of incidence which yields the flattest frequency response.

It should be noted that it is also possible to determine the free-field response of a microphone directly by means of absolute free-field calibration techniques. But this procedure is rather cumbersome and time consuming, involving an anechoic chamber, mechanical devices, and a stabilized sound source. It is therefore common practice to add the diffraction corrections to the pressure calibration to achieve more accurate results.

PRESSURE VS FREE-FIELD MICROPHONES

In describing instrumentation microphones, a transducer whose high-frequency response is flat under constant pressure, or close-coupled conditions, such as the near field of a sound source, is referred to as a pressure-response microphone, or simply as a pressure microphone. When a pressure microphone is employed in a free field, it must be oriented so that its diaphragm is parallel to the sound source (or 90 degrees incident to the center-line axis), in order to obtain a relatively flat response over the entire audio frequency range. However, if reflective sources are present in the environment, the microphone's frequency response will be different for each source, depending upon the angle of incidence. Substantial high-frequency errors will result, for example, if the incidence is predominantly perpendicular, because the microphone will amplify sound waves at the higher frequencies and its basic response will be unpredictable. The smaller (less than one-inch diameter) pressure microphones can be used for random-incidence measurements, since the free-field corrections are very small at the high frequencies and less dependent upon the angles of incidence. Typical microphone response characteristics are shown in Figure 6 for a condenser microphone of the pressure type under three conditions of incident sound waves.

Pressure microphones find wide application as a reference transducer as artificial ears to calibrate earphones and audiometers under well-defined conditions. They are often used as a reference to regulate the sound source when calibrating close-talking microphones for communications headsets, and as probing sensors to measure noise in the near field. Application also includes measurements of sound sources that move in one plane, since the 90 degree grazing incidence is independent of source position.

Free-field response microphones, on the other hand, are designed with an overdamped diaphragm. Thus when the free-field correction for perpendicular orientation (or 0 degrees incident to the center-line axis), is added to the pressure response rolloff, the resulting free-field response is independent of frequency. This means that sound measurements must be made in a free field with the microphone pointed directly at the predominant sound source, in order to achieve a flat, wideband response. Since the measuring axis of a free-field microphone is also its most sensitive axis, sound waves incident from any other angle of incidence will have their high-frequency components attenuated. The curves shown in Figure 7 indicate that when the axis of flat response is aimed at the source, the microphone can be oriented for a maximum reading and thus ensure the recording of accurate data.

In situations involving the measurement of noise from machinery, air conditioners, electric motors, etc., the microphone must measure only the directly radiated sound from the source. Accordingly, the free-field microphone will best serve to avoid directional errors where the incidence of sound is not well-defined. Thus where there are many undesirable sound sources caused by reflections, these errors will be less significant using the smallest diameter microphone available.

In order to accurately measure acoustical environments where there is more than one source, such as in an auditorium, the microphone should have the same frequency response for each source. This requires that the microphone be omnidirectional over the dominant frequency spectrum. Ideally, true omnidirectivity implies that the microphone responds to sounds equally from all directions at all frequencies, irrespective of angle of incidence. But in practice, this condition is seldom met, and some directivity is apparent at the very high frequencies in the audio spectrum. Random-incidence microphones have a reasonably flat response in a diffuse field, where the sound waves at various angles of incidence impinge on the diaphragm simultaneously, the directional errors averaging out to near-zero. To obtain such a nondirectional response, a very small diameter microphone, say, 1/4-inch, may be used because it is far less dependent upon the angle of incidence and will not interfere with the acoustical field. But, as stated earlier, unfortunately the smaller the microphone, the less is its acoustic sensitivity. As an alternate procedure, a one-inch free-field microphone can be used for random-incidence measurements, when fitted with an adaptive device to eliminate directional errors. These so-called random-incidence correctors increase the linear range and omnidirectional properties of the microphone when applied to diffuse-field measurements. They accomplish this effect by increasing the apparent sensitive area of the diaphragm and make the transducer appear smaller than it actually is in the acoustical field. ■

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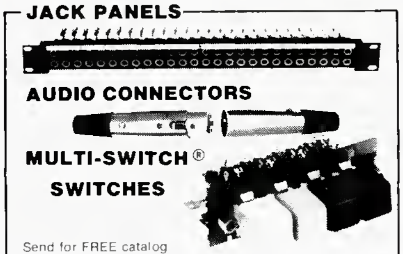

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MUSICIAN-ENGINEER seeks position involving work in both fields. B.S.E., M. Mus. degrees. Music, teaching, location recording experience. **P.O. Box 220, St. Clair, Mich. 48079.**

ESTABLISHED PROFESSIONAL sound equipment company in Dallas, Texas seeks sales applicant capable of system design and sales. Minimum three years' experience required. Excellent opportunity. All replies held strictly confidential. Send resume and reerences to: **Manager, P.O. Box 8205, Dallas, Texas 75205.**

EXPERIENCED MUSIC MIXER
Major N.Y.C. studio. New automated 24-track. Send resume to **Dept. 83, db Magazine, 1120 Old Country Rd., Plainview, N.Y. 11803.**

db People/Places/Happenings

• The new post of senior electro-acoustic systems engineer at the **Koss Corporation** of Milwaukee, Wis. has been filled by **John R. Bruss.** Mr. Bruss had been with the Heath Company.

• **Capitol Stage Lighting Co.,** Inc. of New York City has announced the appointment of **Tom Fay** as their national sales manager. Mr. Fay, an expert in show lighting, comes to Capitol from **Berkey Colortran.**

• Culminating involvement in the manufacturing process at **Teledyne Acoustic Research (AR)** of Norwood, Mass. since 1965, **Ercilio Costa** has been promoted to the position of vice president of manufacturing. Mr. Costa's particular responsibility will be the maintenance of production schedules.

• As a promotion from the post of marketing manager, microwave and scientific products, **Roy K. Durnwirth** has been named general marketing manager at **Amperex Electronic Corporation** of Hicksville, N.Y. Before joining Amperex in 1975, Mr. Durnwirth had been with the **Raytheon Corp.** and **RCA.**

• Coming from **Electro-Voice, William S. Sutherland** has joined **GC Electronics,** of Rockford, Ill. as advertising and sales promotion manager. GC manufactures electronic chemicals, tools, and replacement components, as well as the Matrecs line.

• **John Spiker** has joined the broadcast sales group of **Omega Video, Inc.** of Lawndale, Ca. He will be responsible for video system and equipment sales on the west coast. Mr. Spiker comes to Omega from the **Grass Valley Group.**

• **EICO Electronic Instruments Co.** Inc. has moved its headquarters. Their new address is 108 New South Rd., Hicksville, N.Y. 11801. The telephone number is (516) 681-9300.

• Coming from **Brookshire Electronics, Jim Brawley** has been appointed applications engineer for the professional division of **James B. Lansing Sound, Inc.** of Northridge, Ca. Mr. Brawley will assist JBL sound contractors in backup system design and will provide technical information to the public.

• A call for papers to be presented at the *SMPTE Technical Conference,* scheduled for October 29-November 2 at the Americana Hotel in New York City, has been made. Persons interested in presenting papers should contact **Lynne Robinson,** SMPTE Conference Programs Secretary, 862 Scarsdale Ave., Scarsdale, N.Y. 10583. (914) 472-6606.

• The reassignment of two executives at the **Cetec Corporation** has moved **Robert M. Ward** from the position of president and general manager to corporate staff vice president. **Edward W. Watts,** who had served as a corporate vice president, has assumed management of the broadcast group.

• Acoustical engineering inventor and writer **Abraham B. Cohen** died on March 21 in New York City, at the age of 67. Mr. Cohen was associated wtih **Instrument Systems Corp.** of Huntington, N.Y. Holder of patents in the field of loudspeakers, he had been chief engineer for **University Loudspeakers** in White Plains, N.Y. and Oklahoma City. A book he wrote on loudspeakers was translated into Spanish and Japanese for widespread use.

• One of the deans of the audio industry, **G. A. Briggs,** died on January 11 at the age of 87 at his home in Yorkshire, England. Founder of the **Wharfedale Wireless Works,** now **Rank Wharfedale, Ltd.,** Mr. Briggs was a prime mover in the promotion of high fidelity, organizing sell-out concerts in Carnegie Hall in 1955 and 1956 to demonstrate live versus recorded music. He was also well known as a writer of wit and entertaining style. Best known of his ten books about high fidelity, audio, and music was *Loudspeakers, the How and Why of High Fidelity,* published in 1948.

• **Curtis Pickelle** has been appointed to the position of marketing communications manager at **Altec Lansing International,** of Anaheim, Ca. Mr. Pickelle will be responsible for advertising, public relations, trade show, and other communications operations.

• The appointment of **Wayne Freeman** as sales manager at **Uni-Sync, Inc.** of Westlake Village, Ca. has been announced. Mr. Freeman comes from **BGW .**

• Entitled "Standard Tape Manual," a data reference source of magnetic recording equipment for sophisticated users, is available from **R. K. Morrison Illustrative Materials,** 819 Coventry Rd., Kensington, Ca. 94707.

• Ownership of the Mellotron trademark has been obtained by **Sound Sales, Inc.** of Sherman, Conn., who have commenced U.S. manufacturing of the instrument. There will be versions with single and double keyboards. A 20-in/quad/stereo mixing console will also be produced under the name Mellotron.

• **RCA Broadcast Systems** of Camden, N.J. has agreed to serve as a marketing channel for the **Sony Corporation's** one-inch non-segmented helical scan video tape recorders. The recorders are intended for television and professional teleproduction applications. The new one-inch products will conform to the recording format which is being established by **SMPTE.**

• A coalition of manufacturers has formed **CAMEO,** Creative Audio and Music Electronics Organization to promote and standardize electronic music equipment. Companies participating include, **AKG, BGW, dbx, MXR, Phase Linear, Tapco, and TEAC.** Headquarters of CAMEO are at Suite 3501 LaSalle Plaza, 180 N. LaSalle St., Chicago, Ill. 60601. **David Schulman** is the director.

• Audio tape manufacturer **Memorex Corporation** has expanded its audio tape plant in Santa Clara, California. The new operation, including a 52 ft. long cassette assembly line, will expand production considerably.

• Operating from a Miami, Florida base, **Art Nobo** is the newly appointed **RCA** manager of sales for the Caribbean and Central America. Mr. Nobo has been with RCA since 1968.