ISSN0011-7145

THE SOUND ENGINEERING MAGAZINE

db

JANUARY 1980
$1.95

Coming Next Month

• With the 80s only just launched, some are already proclaiming these next ten years as "the digital decade." That being so, our February issue is full of information on what to expect. Naturally, we'll have something to say about digital tape recorders. And that brings up the subject of electronic digital editing. Of course, we must not forget computerized recording studios either. Why, even the test bench will be micro-processor-based, as you'll discover in next month's issue of **db—The Sound Engineering Magazine.**

About The Cover

• A close-up view of the control panel of the new Ampex ATR-124 recorder—Ampex's last word in analog technology. For more detailed information on the design and operation features of the ATR-124, see this month's "db Special Report."

db

THE SOUND ENGINEERING MAGAZINE

JANUARY 1980 VOLUME 14, NUMBER 1

db is listed in **Current Contents: Engineering and Technology**

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db Calendar

FEBRUARY

1-2 **The 14th Annual Television Conference of the Society of Motion Picture and Television Engineers (SMPTE).** Toronto, Canada. Sheraton Centre Hotel. For more information contact: SMPTE TV Conference, 862 Scarsdale Avenue, Scarsdale, NY 10583.

Syn-Aud-Con Sound Engineering Seminar

19 Day of Basics

20-22 Three-day seminar. Dana Point Marina Inn, Dana Point, CA. For more information on the "Day of Basics" and the three-day seminar contact: Syn-Aud-Con, P.O. Box 1134, Tustin, CA 92680, (714) 838-2288.

25-28 **AES 65th Convention** (London). London Hilton and Park Lane Hotels. For more information contact: Audio Engineering Society, Inc., 60 East 42nd St., New York, NY 10017.

25-29 **The 13th International Instruments, Electronics and Automation Exhibition (IEA).** National Exhibition Centre, Birmingham, England. For more information contact: Industrial and Trade Fairs Ltd.,Radcliffe House, Blenheim Court, Solihull, West Midlands B91, 2BG England, Telephone: 021 705 6707.

25-29 **B&K Measurement Seminar—Industrial Noise Control I.** B&K Instruments, Inc., 5111 W. 164th St., Cleveland, Ohio 44142. Telephone: (216) 267-4800.

26-28 **"Sound 80"** Cunard Hotel. Hammersmith, London.

MARCH

Syn-Aud-Con Sound Engineering Seminar

11 Day of Basics

12-14 Three-day Seminar. Dana Point Marina Inn, Dana Point, CA. For more information on the "Day of Basics" and the three-day seminar contact: Syn-Aud-Con, P.O. Box 1134, Tustin, CA 92680, (714) 838-2288.

18-21 **B&K Measurement Seminar—Quiet Product Design.** B&K Instruments, Inc., 5111 W. 164th St., Cleveland, Ohio 44142. Telephone: (216) 267-4800.

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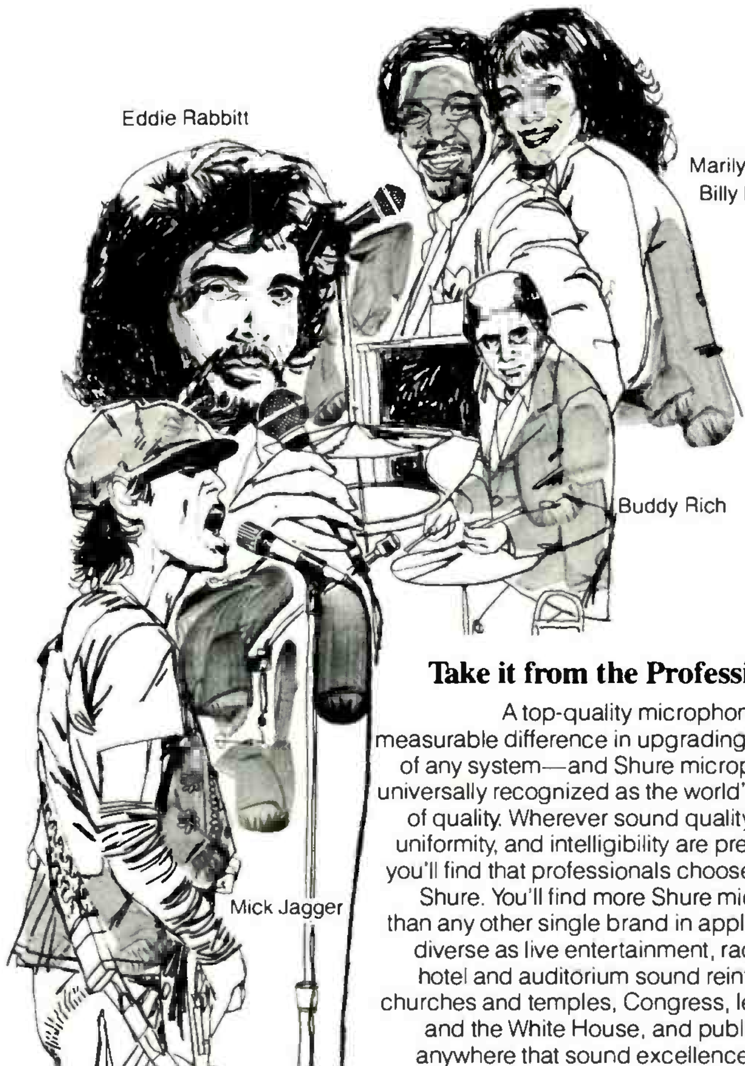

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calendar (cont.)

APRIL

Syn-Aud-Con Sound Engineering Seminar

15 Day of Basics

16-18 Three-day Seminar. Dana Point Marina Inn. Dana Point, CA. For more information on the "Day of Basics" and the three-day seminar contact: Syn-Aud-Con, P.O. Box 1134, Tustin, CA 92680, (714) 838-2288.

28-5 1 **NOISEXPO '80.** The National Noise and Vibraton Control Conference and Exhibition. Hyatt Regency O'Hare, Chicago, IL. Registration information is available from: Noisexpo, 27101 East Oviatt Road, Bay Village, OH 44140, (216) 835-0101.

15-18 **Communications '80.** Communications Equipment and Systems Exhibition. National Exhibition Centre, Brighton, England. For more information contact: British Information Services, 845 Third Avenue, New York, NY 10022, (212) 752-8400.

21-25 **B&K Measurement Seminar—Industrial Noise Control I.** B&K Instruments, Inc., 5111 W. 164th St., Cleveland, Ohio 44142. Telephone: (216) 267-4800.

28-5 1 **Audio-Visual '80 Exhibition & Conference.** Wembley Conference Centre, London, England. For more information contact: British Information Services, 845 Third avenue, New York, NY 10022, (212) 752-8400.

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6-7 **B&K Measurement Seminar—Audiometer Calibration.** B&K Instruments, Inc., 5111 W. 164th St., Cleveland, Ohio 44142. Telephone: (216) 267-4800.

27-30 **B&K Measurement Seminar—Quiet Product Design.** B&K Instruments, Inc., 5111 W. 164th St., Cleveland, Ohio 44142. Telephone: (216) 267-4800.

JUNE

19-20 **APRS '80** International Exhibition of Professional Recording Equipment. Connaught Rooms, London, England. For more information contact: British Information Services, 845 Third Avenue, New York, NY 10022, (212) 752-8400.

23-27 **B&K Measurement Seminar—Industrial Noise Control I.** B&K Instruments, Inc., 5111 W. 164th St., Cleveland, Ohio 44142. Telephone: (216) 267-4800.

db Letters

TO THE EDITOR:

Please note that in the September 1979 issue of **db.** on page 53 of the "Convention Report: APRS 1979," our valuable trademark HARMONIZER was used in small case type, and this can be damaging to our trademark rights.

At your convenience, therefore, we would appreciate it if you would publish a correction statement.

ORVILLE N. GREENE
President,
Eventide Clockworks, Inc.

db Replies:

The trademark "Harmonizer" should have been capitalized since we are aware that "Harmonizer" is an important trademark for certain products of Eventide Clockworks, Inc. and is so recognized by the industry.

TO THE EDITOR:

A belated note of thanks for the marvelous piece about WNCN in the September issue of **db**. We have received so much positive feedback from the article, particularly at the New York Hi-Fi Show.

As you probably know, technical work of the sort done by WNCN is never finished, and I think we will be able to tell you about additional progress in the near future.

ROBERT E. RICHER
General Manager
WNCN-FM, NY

db Replies:

Your comments are greatly appreciated; and your letter has been forwarded to Mr. Len Feldman, the author of the article.

TO THE EDITOR:

I just read your piece on "Education & Audio" and I began to wonder how I had ever learned what I did learn so long ago. I think it is a matter of wanting to learn, in the first place, and then working at it until you finally know whatever you want to know. That is, of course, if what you want to know is available knowledge —I have been trying to find out how the human brain is organized, for many years, and still know next to nothing.

I managed to learn a good deal about wave phenomena when I was a mere child because that was what all the kids were interested in at that time. We all, I think, wanted to hear the peculiar sounds you could hear if you built a crystal receiver.

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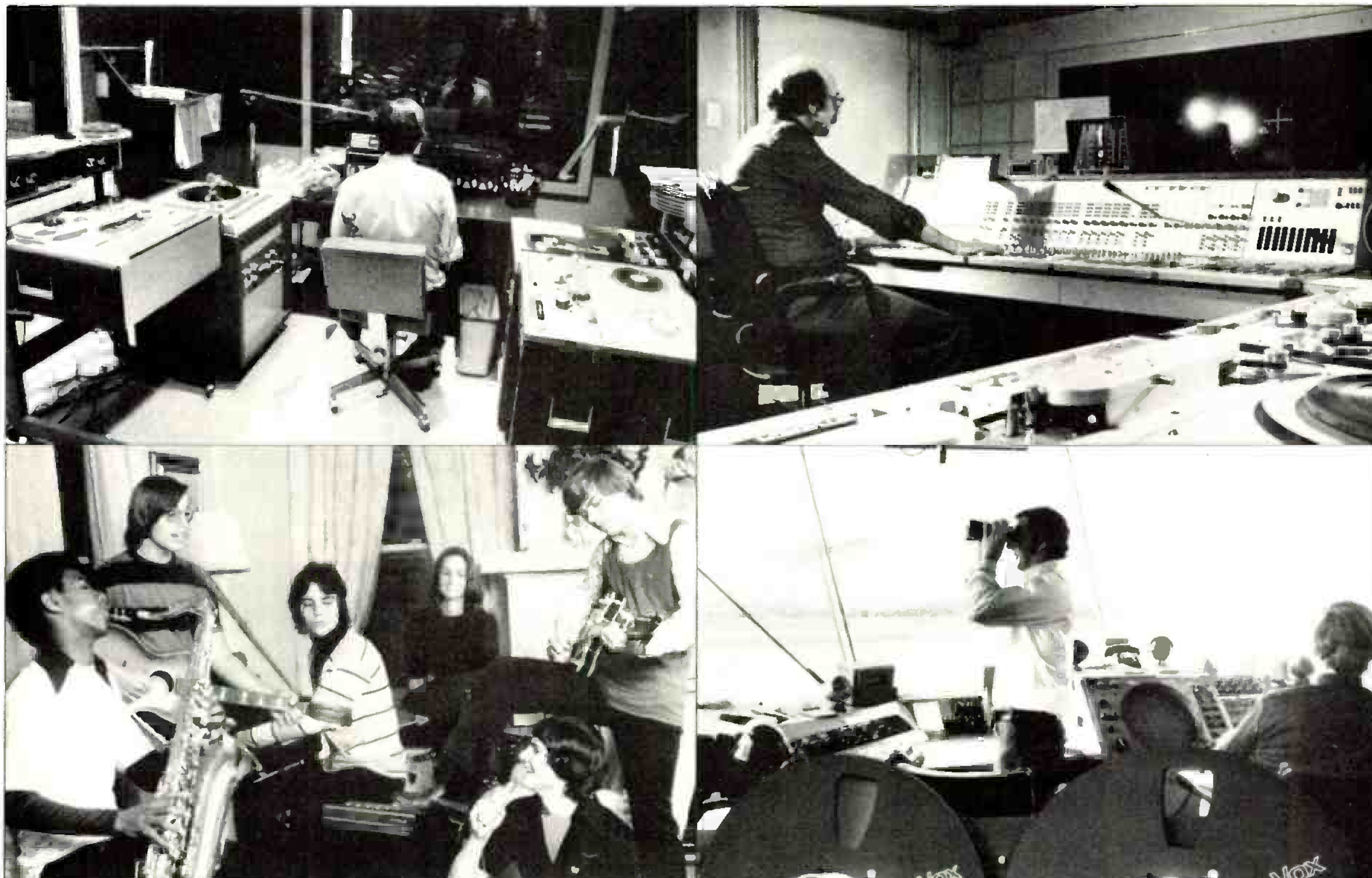

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I made one when I was 11 years old, in 1916. At the same time I was interested in photography, so I became a double menace to my family in that I pre-empted the only bathroom to develop my films and probably annoyed everyone with my discovery of the Morse Code. What one could hear, in those days, was mainly the "chirping" sounds of code transmission, mainly long wave. I was once surprised to hear a woman singing, about that time, and thought for years that I had heard Marconi playing a record from his yacht, the Electra. It was only a couple of years ago, in Zurich, that I met a member of the Sapphire Club (whose name I forgot! Forgive me, whoever you are!) who told me that what I heard was actual live sound, broadcast from a naval vessel through a carbon microphone inserted in the ground leg of the antenna! I hope for confirmation of this marvel.

After a while I became an expert wireman—I think, the first male employee of Automatic Radio, which at that time (1923-24?) was making 5-tube tuned rf radio receivers. We learned what was what by experimenting with tubes and all kinds of circuits. I became an "expert" in tubes when there were only a few tubes, WD11 & 12, UV 199 and 201A and 200, plus a couple of Western Electric high power bottles. We learned about loudspeakers by making them. I think, in the very early days, we made practically everything but the tube sockets—wound our own coils and made our own variable condensers.

These days an earnest youngster can find what he wants in almost any first-class college. The study of wave phenomena, the physics of light and sound, particle physics, magnetic energy, solid state physics, A-C mathematics and so on. And there is nothing to prohibit him or her from experimenting, it is easier now to experiment than it was when I was a kid. There's more stuff to experiment with!

As for us old timers, well, we can write letters to Editors telling them how it was—and does anybody out there still have a DeForest 3 circuit tuner?

JOE TALL
Washington, D.C.

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Copies of all issues of **db—The Sound Engineering Magazine** starting with the November 1967 issue are now available on 35 mm. microfilm. For further information or to place your order please write directly to:

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MARTIN DICKSTEIN

db Sound With Images

AV or Not AV, That Is The Question

• Although audio visuals have become a big and important medium recently—growing bigger yet with multi-image, multi-screen presentations using computers to program action so fast it looks to the viewer like a movie—the questions being asked now are what audio visuals to use, when to use them, and even should they be used at all.

HOW TO MAKE A PRESENTATION

At a recent seminar, there was a session on how to make a presentation. The "star" of the presentation, according to the expert giving the seminar, should be the speaker, and not the slides or the films, or the overhead transparency, or anything else...just the speaker. The leader of the meeting video taped the attendees during a four-minute talk on any subject of their choice, and then the group critiqued the playback.

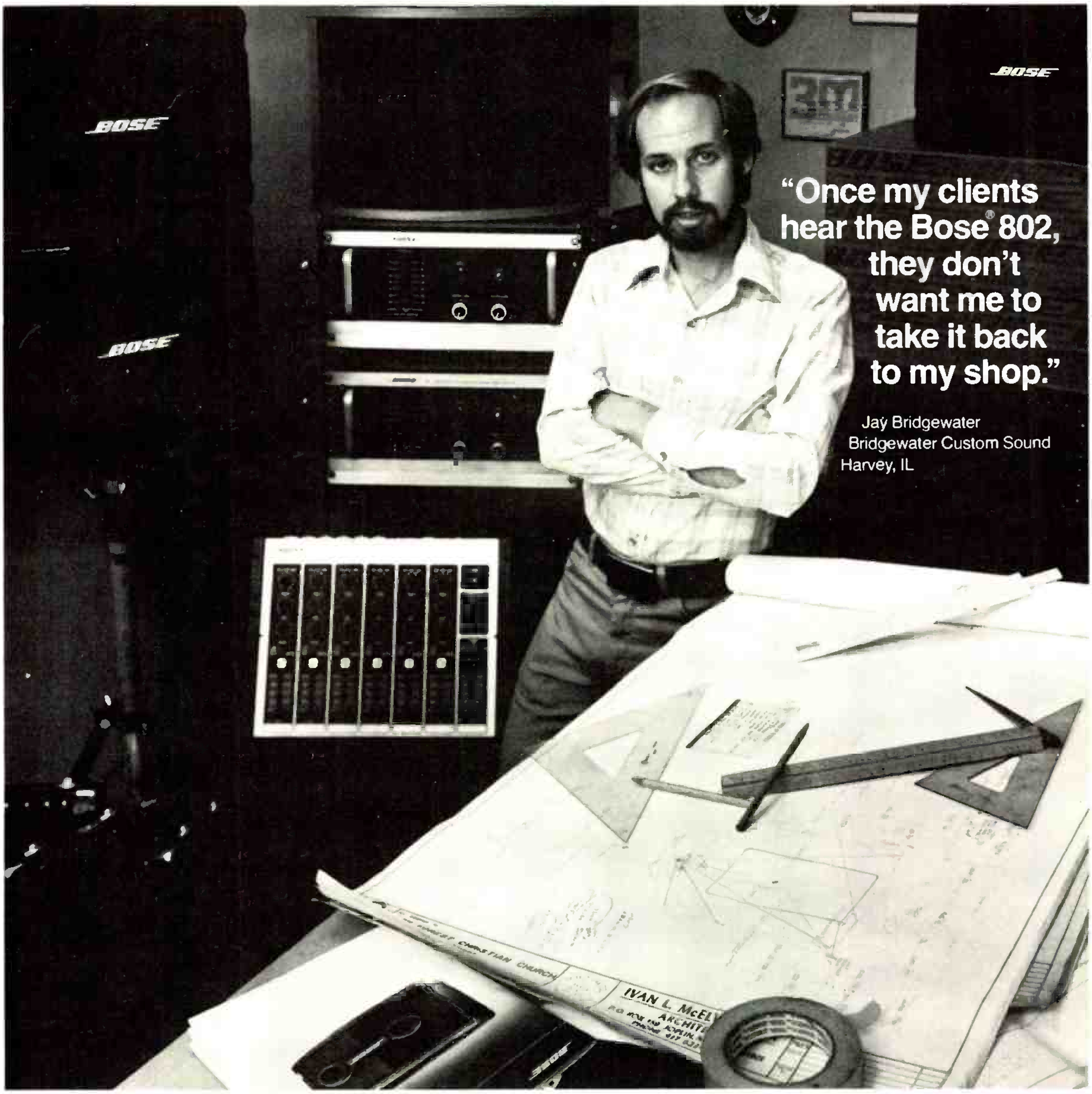
It became quite obvious that some speakers continually looked at their script material, and only looked up at the audience during the pause to take a breath. Others, who did look up, looked at only one side of the audience. Some speakers leaned on the lectern for dear life—perhaps to hide their nervousness and keep their hands from shaking, hoping their voice would not give them away.

Those who used visuals, such as slides, turned toward the screen to read the slide, and consequently their voices were lost as they got off-microphone. Those that used film also had the room darkened, as did those that used slides, and this allowed them to hide from the eyes of the audience. As a result, their posture became careless and their reading of the script was "blah"—as though the film would carry them through.

Some who used slides, used a pointer, and moved away from the lectern—thereby losing voice intelligibility, in addition to having their back to the audience. The ones who used overhead transparencies shuffled them neatly enough to keep them in order, but set them on the projector at all angles, left the projector light on during long portions where no transparency was used, spoke down to the projector instead of to the group, and turned toward the screen behind them to read some of the material on the projected image.

PRESENTATION TIPS

The expert then told the group how to rid themselves of tension and nervousness, how to "eye" the audience...the entire audience from one side to the other, from front to back, how to project the voice, how to use the script as an aid rather than a crutch, how to stand without hanging on to the lectern, how to use the hands, etc. He pointed out that the one thing that seemed to come between the speaker and the audience the most was the visual being used; and with that, recommended that visuals be left out of the presentations altogether. With words to the effect that it was much better to say whatever had to be said in full light, without the hindrance of the visuals, the expert told the group that *they* were the important visual, and that they should be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seen as well as *heard*. In essence, the visuals take the spotlight away from the presenter and the darkened room gives the audience a chance to take their eyes off the speaker. The talk becomes less important, and the psychology of hiding behind the visuals gives the speaker a chance to become lax in his or her presentation—thus losing the sharp edge hoped for in the speech.

Many people may agree with this philosophy. The members of the audience, all of whom were either first or second in command of their respective offices of large companies throughout the world, disagreed. They claimed that it was impossible for them to make client presentations, or even internal presentations, without the use of some visuals. Some of the information they had to offer could, perhaps, be transmitted verbally, but much of it had to be shown, to be explained.

The leader of the session suggested that the material that had to be shown could possibly be included in printed form and given to each person in the audience. This suggestion was noisily rebuffed. The attendees agreed unanimously that it was a good idea to give some material in printed form so the clients could take it away with them for future reference; but at the meeting itself, the introduction of printed pages would give the audience something else to look at during the presentation, and this was undesirable.

NO SUBSTITUTE FOR GOOD VISUALS

The expert agreed and suggested that there must be some other way to show the information without the use of projected visuals, but no substitute could be found. Discussion on this subject continued for some time, but not one alternative was acceptable to all. Some suggested the use of a large pad on an easel. This was turned down because the presenter would have to write the information and it was time consuming, some of them did not have good handwriting, and generally they felt awkward away from the lectern.

Another suggestion was the use of large boards with the material already printed on them. Some agreed, but others said that this medium was good with a small group, but could not be used in large group situations.

Without any alternative to the use of slides and film, and with general agreement that overhead transparencies had applications also, the discussion turned to the use of visuals—but visuals used properly. Even the session leader, agreed that visuals had a place. Now the question became: How to use them?

First, it was decided that the wrong times to start meetings with visuals was: early in the morning when the audience might still have some sleep in them, and the darkened room would be too sug-

gestive; and immediately after lunch while the people might still be feeling the effects of the drinks and heavy food they had at lunch. (Nothing was said about after dinner, but this might also be a good time to keep the room well lit for a while, at the beginning of an evening meeting.)

WHEN TO INTRODUCE VISUALS

The session leader then suggested that no meeting should start with visuals where the room had to be darkened right after the presenter came to the lectern. First, the person speaking has to establish that he or she is an authority on the subject to be presented, and give the audience the feeling that all visuals are only visual aids—used to help present essential information to the whole audience at the same time. At no time during the presentation should the speaker lose sight of that fact. The dominance of the speaker, over the visual aid, has to be maintained at all times. There is a correct way to point to a spot on a slide, either with a pointer (use the inside arm to keep the face of the speaker toward the audience), or with a flashlight-type pointer (where the image of an arrow could be aimed at the spot on the screen being discussed).

There is also a proper way to handle transparencies so that the speaker does not turn away from the audience, nor place the transparency on the projector incorrectly, thereby losing the position of importance in the presentation. It was also suggested that where no visuals are to be used for any length of time, such as several pages of a written speech, or during lengthy talk periods, the lights in the room should be brought up so the audience could again focus attention on the presenter.

PREPARATION OF SLIDES

The very important point of properly produced visuals also came under discussion. Slides should be made with as few words as possible—and large enough for everyone to see. When large columns of numbers had to be shown, the portion of importance should be shown in blown-up size for easy reading. The same holds true for transparencies—key points only in visual form, the rest should be left for the presenter to say. It was also mentioned that there is greatest retentivity when the important points are seen as well as heard; and this led to having only key figures, percentages, words, etc. on the screen, for greatest impact. Color of slides and color of words also came up, with general agreement that a dark background with bright type such as white or yellow should be used; but not a black background, as this seemed to produce eye fatigue over any length of time.

Several other factors of a similar nature were also discussed, but it was generally agreed that the presenter is the important factor of any presentation, and that visual aids are precisely that—aids. Made properly, and used properly, visual aids have a definite place and reason for being used. It is up to the presenter to maintain the proper balance, and not become the aid to the visual.

Lest you think we forgot, we do wish you a very happy, successful, healthy, and peaceful New Year. ■

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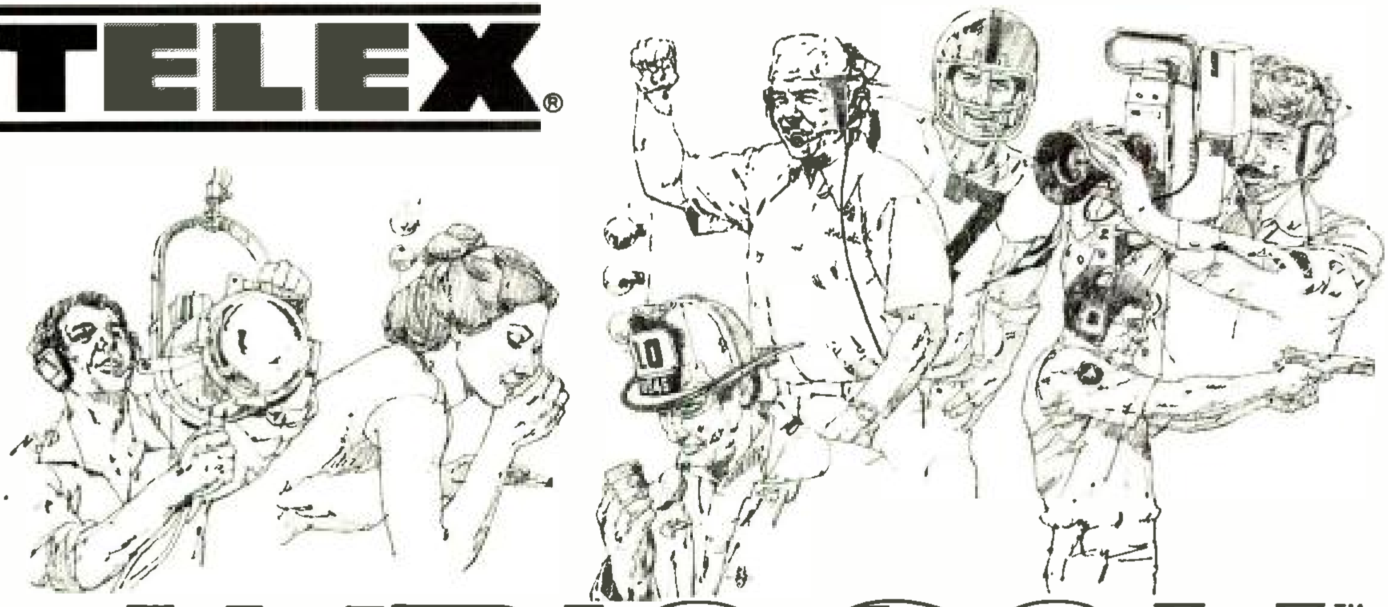

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NORMAN H. CROWHURST

System Phasing

• A letter just came my way, actually arising from something published in another magazine. But the reader also reads **db**, and this column in particular, so he asked me to cover the subject—and it is one that needs it. The original discussion relates to the use of 3-pin microphone connectors, with balanced or unbalanced lines. But the reader's main concern is with phasing.

He enclosed quite a volume of manufacturer's literature, showing how each manufacturer recommends connections be made. In all of them, pin 1 is ground (or "earth" if the manufacturer is British) and is connected to the cable sheath, or shield. The question where differences creep in, concern whether, when you go to unbalanced line, the jumper should bridge pins 1 and 2, or pins 1 and 3.

"HOT" OR "COLD"?

Many of the instructions state that either pin 2 or pin 3 (they do not agree on this) is "high" or "hot," and that the other one should be grounded for unbalanced connection. And the question arises what is meant by "high," or "hot". If you have a balanced line, that is a good question, because in theory, each side is equally "hot." Ground should be at a mid point between them, although actually they should be floating.

In some systems, to force a better balance, a transformer or center-tapped resistance is used to connect ground to a mid point, so that the phasing of the leads wired to pins 2 and 3 is equal and opposite. This means that, in the conventional sense, both sides are equally hot.

If such a center point balance is achieved with a center-tapped winding on a transformer, then grounding either side will "short out the audio world" as an original writer on the subject expressed it. But what he said was that that would happen, if you grounded the *wrong* side, which thus became a matter for discussion: which was the wrong side to ground?

And if an artificial center point is grounded by means of a center-tapped resistance, or two equal resistors, then grounding one side will unbalance the line and, at the same time, change the loading provided by the resistors. If the microphone has two 330-ohm resistors for this purpose, its balanced loading will be 660 ohms, and its unbalanced loading 330 ohms, whichever side you ground.

This is a situation more often encountered in constant-voltage line distribution, to feed loudspeakers: the output transformer has a center tap that goes to ground. Grounding either side will short out the amplifier output, through that half of the transformer secondary. But in microphone circuits, center-tapping is far less common: in most balanced circuits, the connection floats, so that either side could be grounded, without changing the loading, or so it would seem.

However, it has been known for a system to exhibit a "hot" and "cold" side in such a situation, in that grounding the wrong one produces instability, while the other one works fine. Now, the way I would regard "hot" and "cold," I would say that the one that works fine was the "hot" side. Why should there be such a difference?

CAPACITANCE FEEDBACK

If both sides are truly the same, then it should make no difference which side you ground: there will be double the voltage on the other side. One reason that constant voltage lines use a center point ground, is so that both conductors have equal and opposite voltages, to minimize possible capacitance feedback to low-level circuits, such as microphones.

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Under the lights at discos around the world, more dancers are moving to JBLs . . . by far the leading disco speakers (Billboard's International Disco source-book).

Night club performers rely on JBL's new Cabaret Series loudspeakers—the first JBLs made especially for club musicians.

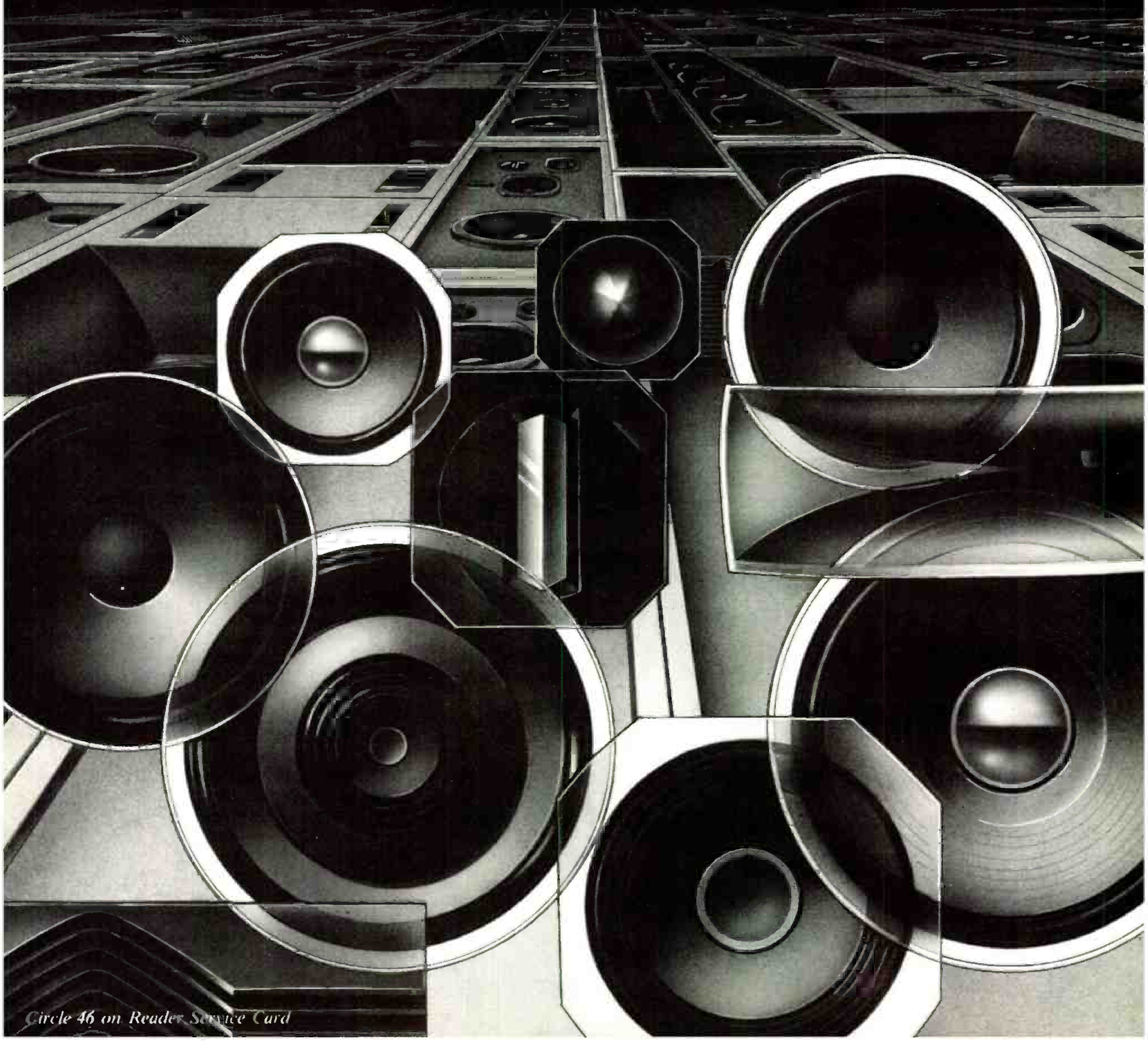
Theater owners who want their audiences to experience today's great new multiple sound tracks are installing JBLs.

And serious musician/songwriters who rely on home recording are following the lead of the big recording studios: They're turning to JBLs, too.

Wherever excellence belongs, you'll find JBL.

James B. Lansing Sound, Inc., 8500 Balboa Boulevard, Northridge, CA 91329.

FIRST WITH THE PROS. JBL



Circle 46 on Reader Service Card

a microphone connection, instead of its center point, or letting it float. And it is possible that one side will do that more readily, while grounding the other side is quite stable. This is usually due to the fact the circuit is not symmetrical, electrically, particularly at the higher frequencies, where capacitance takes effect.

In these circumstances, grounding the "hot" side will provide stability, while grounding the "cold" side will aggravate the "hotness" of the other side. But that is not what determines which side to connect to pin 1, as a matter of "standard connections." This is far more generally connected with microphone phasing, in which there is, unfortunately, not an industry standard—or perhaps it would be more accurate to say that the standard is not universally used.

My inquirer is concerned with system phasing: whether, in the whole system, forward pressure at the microphone ultimately results in forward movement of the loudspeaker diaphragm. He suggests that this is desirable, particularly in handling asymmetrical waveforms, such as voice, and he has a point there. Is that important, or isn't it? Is it only relative phasing of the system that matters?

We have so many compromises to make in audio, that we should look at this from a number of angles. First, what is the nature of the sound wave that propagates from such an "asymmetrical source:" is it a synthesis of frequencies, or is it a composite waveform? If it is a composite waveform, then it does not change, as it passes through space. If it is a synthesis of frequencies then, since every frequency has a different wavelength, the phase relationship must change as it progresses.

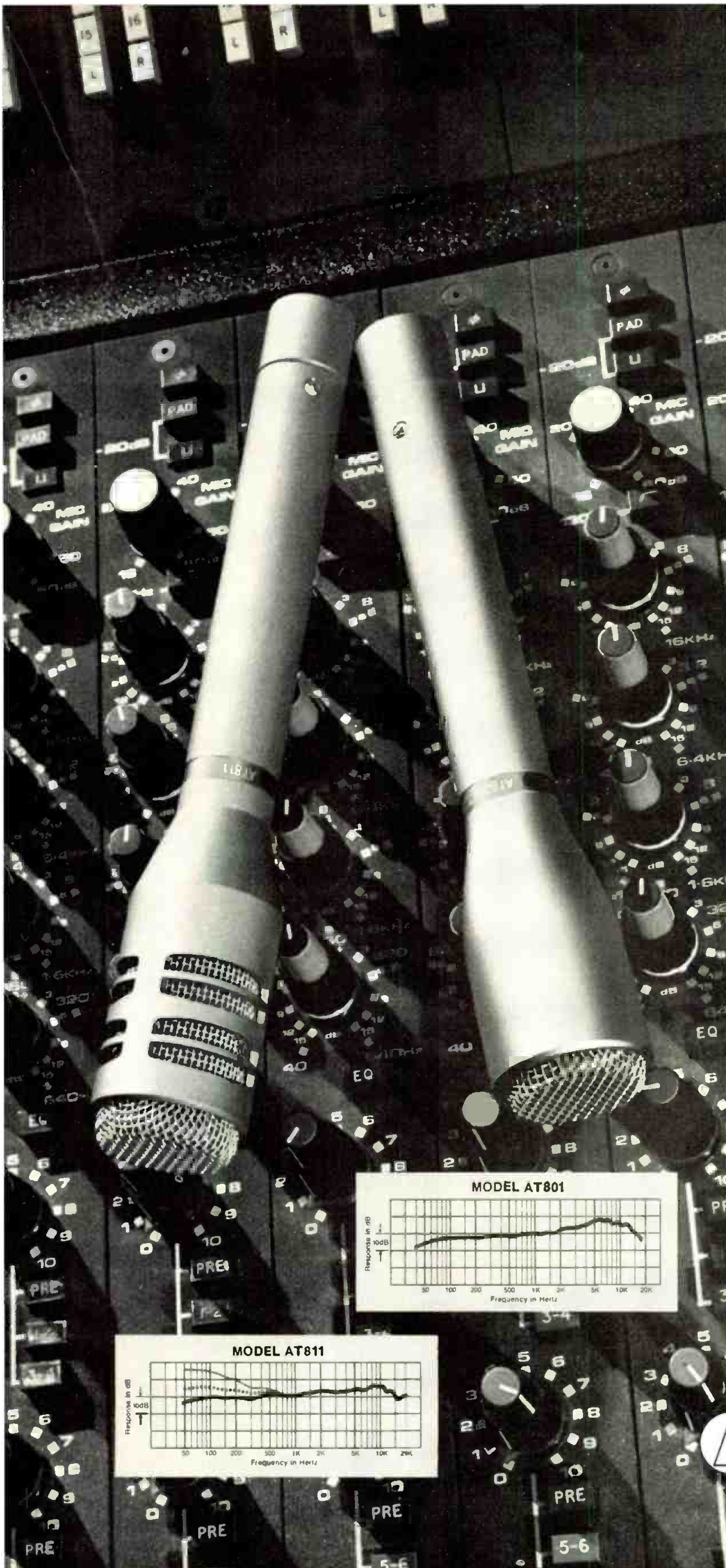
Which is true? Neither, completely. To answer this, you need to think about how sound propagates, by means of a combination of pressure and velocity fluctuations that are mutually self-supporting, once the source has given them form. Sound velocity is not dependent on frequency, so a waveform emitted should not change its shape, as it progresses across space. In most instances, this probably more closely approximates reality.

But because different frequencies, present in the waveform, are affected differently by objects in the path of the wave, changes do occur. In fact, it is this change by which our hearing faculty is most easily able to separate direct sound from reverberant sound. This can easily be demonstrated by taking a mono signal, feeding it to two loudspeakers, fairly close together, and changing the relative phase of their connections.

In phase, the sound appears to come from a point midway between them, particularly when you stand so they are equidistant from your listening position. But connect them out of phase, and the source becomes confused, sounding more like reverberation. What this says is that relative phase is important in wiring the loudspeaker end of a system. This is just as true—maybe more so—for stereo or quadraphonic, as it is for mono.

The same thing has been shown to be true for microphones: they too should be connected in phase. And for vocals, or even for orchestral use, it is also important that each microphone be used to pick up the sounds for which it is responsible, at an adequately higher level than neighboring microphones, to avoid another deficiency that poor phasing can produce: erratic frequency response. This is why it is better to use 2 microphones than 4, for a close quartet, for example, placing 2 members of the quartet close to each microphone, while allowing adequate distance to ensure that the "wrong" microphone does not pick up the "others" at appreciable level, to produce this effect.

But that treats the pickup area, covered by microphones, as separate and distinct from the area served by the loudspeakers. For recording, or radio, or television, this is true. Then the only remaining question, is the one whether reversing phase, so that an instantaneous pressure at the microphone diaphragm



The Audio-Technica philosophy:

EQ should be used to improve the sound... not to fix the mike!

Introducing affordable smooth sound. The remarkable AT801 and AT811 Electret Condensers. With curves so smooth you would have to pay a bundle to match them anywhere else.

Response like this has a number of benefits. First, your EQ is used only to touch up the sound, not to correct built-in errors of the microphone. Which leaves more leeway to control the overall sound.

And without unwanted peaks you have more usable headroom. That's vital when you're working near the dynamic limit of a preamp or line amp. Sound stays clean and sharp. Compressors or limiters sound less forced, because they are controlling peaks in the *sound*, not peaks in the *mike!*

But perhaps the biggest advantage is the versatility of these A-T condensers. Because they have just the right amount of presence for today's recordings, they're not limited to just one kind of instrument...just one type of voice. Put them *anywhere* in the mix: brass, reeds, percussion, chorus, or strings. Then listen. What you hear in the studio you'll hear at the console. Which is a great place to start in miking any session.

At their highly affordable prices, these are two of the best bargains you'll find these days. Reliable, clean-sounding, and the most predictable microphones you can use. Make them a mainstay in your studio today. Write for spec sheets and dealer list.

Model AT801 Omnidirectional Electret Condenser

Model AT811 Unidirectional Electret Condenser

Great sound, right from the start!

audio-technica®
INNOVATION □ PRECISION □ INTEGRITY

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produces a reversal at the loudspeaker diaphragm, diminishes the faithfulness of reproduction, as compared with reproducing a pressure at the same instant, referred to program time.

This would be extremely difficult to standardize. It is one thing to standardize microphone connections, so that microphones are always correctly phased at the "pick-up" end. It is another thing to standardize loudspeaker connections, whether in mono, stereo or quad, so that in a particular system, all the loudspeakers work together correctly. Putting in a phase reversal, so that pressure at any microphone results in the reverse at any loudspeaker (not interfering with the integrity of the system at either end, only with the overall result), is something that can easily happen.

Must pressure at the microphone result in upward modulation of an AM radio signal, or phase advance of an FM signal? And must the same relationship follow-through at the receiving end? Changes of either one could result in failure of correct phasing in this sense. Perhaps some day this will be standardized. The same goes for recording, whether mechanical (disc) or magnetic.

But there is one more factor when we get into sound reinforcement, or public address applications. What happens if you have the simplest auditorium reinforcement system: a single microphone with a correctly phased installation of loudspeakers; and change the microphone phasing?

ACOUSTIC FEEDBACK AND SYSTEM PHASE

This takes considerable analysis. Now the acoustic feedback problem can come into the picture. As soon as acoustic feedback starts up, a standing wave pattern builds. In a standing wave pattern there are nodal lines where pressure fluctuates, but the air does not move, because air moves in and out from both sides of the node. And there are antinodes, where maximum particle movement occurs, between nodal points or lines.

A purely pressure microphone (which usually has omnidirectional pattern) will usually be part of a pattern in which it lies on a node—maximum pressure, minimum particle movement. A directional microphone will be modified by the fact that movement is part of its function. But the point relative to this discussion is that if the system is in phase, as defined earlier, the distance from various loudspeakers to the microphone will be measured as an even number of half wavelengths, at the frequency of feedback.

If the system is out of phase, the distance that causes the build-up will be measured as an odd number of half wavelengths of the frequency that builds up.

This could suggest that reversing the phase of the microphone would cause considerable shift in the frequency at which feedback builds up, and also on the gain setting at which it occurs. However, this is not necessarily true. In the frequency range where such feedback occurs, there are usually a closely spaced sequence of frequencies, with their associated standing wave patterns, so there may be a standing wave pattern of quite different shape or configuration so that frequency may not shift very much, because the active path along which the number of wavelengths is measured, will be different.

In fact, moving the microphone a few inches or, in the case of a directional type, changing its orientation a little, can cause the pattern and its frequency to change, just about as much (or little) as reversing phase can.

As with so many other questions this column has discussed, there are no simple, universal answers. What may be important factors in one situation may assume secondary significance—or perhaps no significance at all—in other situations. And, in any situation, the compromise selected as best for that situation, will *not* be one that can be generalized as the best solution for any problem. ■

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db New Products & Services

CONDENSER MICROPHONES

• Two new condenser microphones, the MSXY-8 and the XY-82 feature extremely smooth ON-OFF axis frequency response from 25 Hz to 20 kHz, with self-noise better than 16 dBA. The MSXY-8 microphone has two variable capsules (each offering omni, cardioid and figure-8 pickup patterns), remote pattern control and a.c. power supply. The XY-82 is a coincident microphone with two cardioid capsules requiring 24-48 v d.c. phantom powering. Both microphones permit one capsule to be rotated through 180 degrees, and each have fet preamplifiers. While offering MS or X-Y stereo recording techniques, and variations thereof, these mics also provide the mixing engineer the possibility to obtain a 35 to 40 dB front-to-back ratio for maximum instrumental isolation when used as a monophonic microphone. By connecting the two mic outputs of the XY-82 to two mixer input positions, with one of the mic outputs inverted 180 degrees in phase, and one of the mic capsules rotated 180 degrees from the other, remote pattern control at the mixing console is achieved. With this set-up, pickup patterns of omni, cardioid and figure-8 are obtained by adjusting the mixer input pad and faders.

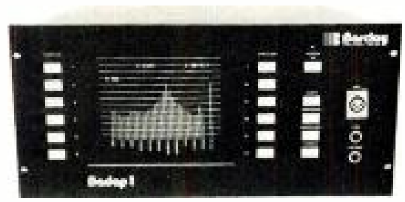
Mfr: PML (Pearl)
Cara International Ltd. (Distributor)
Price: MSXY-8, $2,495.00 (with remote a.c. power supply)
XY-82, $1,495.00 (without power supply)
Circle 51 on Reader Service Card

ELECTRONIC FILTER

• An active stereo filter, the model 4100 is designed to eliminate low frequency (subsonic), as well as high frequency (ultrasonic) interference. The low frequency filter section effectively removes subsonic effects below 20 Hz at 18 dB/octave; while the high frequency filter section eliminates ultrasonic effects above 20 kHz at 12 dB octave. The distortion rating for the model 4100 is 0.002 per cent IM (typical), and less than 0.025 per cent from 20 Hz to 20 kHz.

Mfr: Ace Audio Co.
Price: Kits, $66.50; Wired Units, $97.00
Circle 52 on Reader Service Card

AUDIO MICRO-COMPUTER



• A programmable audio test instrument, the Badcap I Audio Micro-Computer provides measurements of third-octave response, room reverb time, wow and flutter, Critical Distance, and more. The unit employs a full-color display for easier interpretation and comparison of data, and multiple memories for the storage/recall of data. Applications of the Badcap I include: room EQ, spectrum analysis, disc/tape monitoring, sound system design, room acoustical treatment, multi-channel monitoring, etc. All necessary test signals are provided by the unit.

Mfr: Barclay Analytical Limited
Price: $5,495.00
Circle 53 on Reader Service Card

MATCHING TRANSFORMERS

• Transformers wired into rugged cables, or in compact plug-in form, match either 200 or 500 ohm impedances to 2,000, 5,000, 45,000, 50,000 or 80,000 ohms. Isolation transformers are also available with a 200 ohm input and output impedance for use in preventing ground loops and other interface problems.

Mfr: Beyer
Burns Audiotronics, Inc. (Distributor)
Circle 54 on Reader Service Card

Circle 25 on Reader Service Card

MIXING CONSOLE

• The model 5B is an eight-in, four-out mixing console. Each input module provides three-position attenuation, rotary trim from 0 to 20 dB of continuously variable gain reduction for mic, tape or line signals; three-position input selector for mic or line, tape in or EQ; a foldback pre-EQ or fader cue; post-fader echo, and built-in peak and dip equalization. In addition, each input module contains a solo function, direct output, channel assignment, pan, overload led light, and straight line fader. Specwise, the console has a frequency response of 30 Hz to 20 kHz, ±2 dB; overall signal-to-noise ratio (one input, mic or line) greater than 75 dB (weighted), 70 dB (unweighted); and crosstalk greater than 60 dB (at 1 kHz).
Mfr: TEAC Corporation of America
Price: $1,900.00
Circle 55 on Reader Service Card

MICROPHONE

• Designed to provide broadcast journalists and interviewers with a highly dependable tool for field reporting, the D-130 is an omnidirectional microphone, featuring heavy-duty shock-proof construction. Cushioned within a compliant impact-absorbing suspension, the transducer element of the D-130 is virtually immune to handling noise, mechanical shocks and spurious vibrations. In addition, a hum-bucking coil is incorporated into the microphone to cancel the effects of stray magnetic fields. The D-130 has a frequency range of 50 Hz to 13 kHz, with a nominal impedance of 200 ohms and 54.5 dBm sensitivity.
Mfr: AKG Acoustics
Price: $80.00
Circle 56 on Reader Service Card

NOISE GENERATOR

• A new addition to the IMPAC series of modular plug-in card amplifiers, the model 4017 noise generator is capable of providing pink or white noise for use in amplifier bandwidth testing, equalizer testing, speaker analysis, and room analysis. The model 4017 offers outputs from -3 dBm to +20 dBm into a 75 ohm load, with selection of either pink or white noise made by means of wire jumpers on the mating PC board connector. The module is powered by conventional ±15 v d.c. supplies.
Mfr: Modular Audio Products
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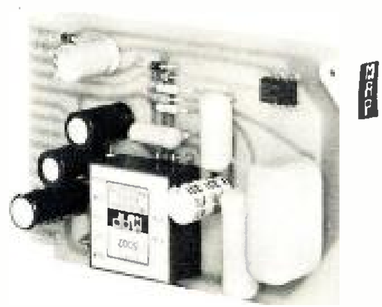
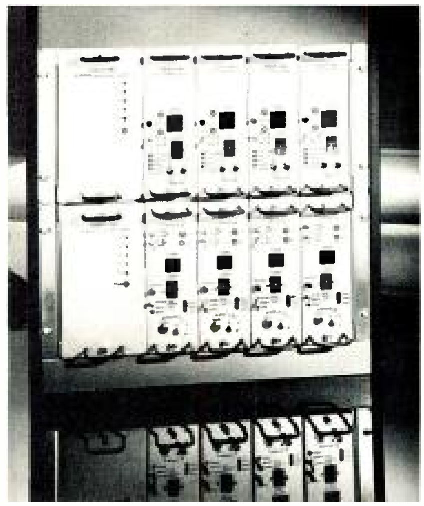
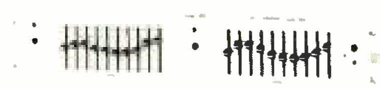


SCPC TERMINAL



• A Single Channel Per Carrier (SCPC) terminal, the model 412 can be used to receive or transmit high-quality audio program material, voice, or data at any carrier frequency in the range of 52 MHz to 88 MHz. Providing the satellite terminal operator with 16 manual or remote selectable channels spaced at least 300 kHz apart, the 412 modulates or demodulates audio, voice, or data directly to, or from the 52 to 88 MHz spectrum. The 412 modulator incorporates a frequency stabilization network to maintain carrier frequency within ±0.003 per cent per year.
Mfr: COASTCOM
Circle 58 on Reader Service Card

EQUALIZER



• Two independent graphic equalizers with one common power supply, the model 535 provides 12 dB of boost or cut at each of 10 preferred ISO one-octave increments from 31.5 Hz to 16 kHz. Filters combine smoothly with minimum phase shift. Gain adjustment via front panel controls is from -10 dB to +20 dB, and the signal-to-noise ratio is better than 110 dB at maximum output.
Mfr: UREI
Price: $424.00
Circle 59 on Reader Service Card

"Once you get your hands on this machine . . . you'll see what we mean."

PERFORMANCE:
Overall Signal-to-Noise: 66 dB unweighted at 520 nWb/m (30 Hz to 18 kHz audio filter).

Playback Signal-to-Noise (electronics): 72 dB unweighted (with audio filter).

Headroom: +24 dB. Maximum Output: +28 dBm.

Overall Frequency Response (15 ips): 30 Hz to 22 kHz ±2 dB.

Playback Frequency Response (MRL test tape): 31.5 Hz to 20 kHz ±2 dB.

RELIABILITY: An unmatched four-year track record of on the job performance for the original compact professional recorder. Day in, night out. Just ask someone you trust.

ALIGNABILITY: Any tape recorder must be aligned to achieve maximum performance. With the MX-5050-B, all primary alignments are on the front panel. So is a 1-kHz test oscillator. Secondary alignments are inside the bottom panel. You or your maintenance people can align it fast and easy. This saves you time, money, and enhances your reputation.

INTERFACEABILITY: With a flick of the output switch you can plug-in to any system: +4 dBm 600 ohm or −10 dB high impedance. No line amps or pads to mess with. A perfect match everytime.

ADDITIONAL BENEFITS: Three speeds, dc servo ±7%, ¼ track reproduce, full edit capability, over-dubbing, noise free inserts, XLR connectors, NAB/CCIR switching, unique three-position alignment level switch.

PRICE: Suggested retail price $2,050 (USA).

MX-5050-B: The best value in a professional tape recorder.

OTARI®

Call Ruth Pruett Ables on 415/592-8311 for the name of your nearest Otari professional dealer.
Otari Corporation, 1559 Industrial Road,
San Carlos, CA 94070 TWX 910-376-4890
In Canada: BSR (Canada, Ltd.),
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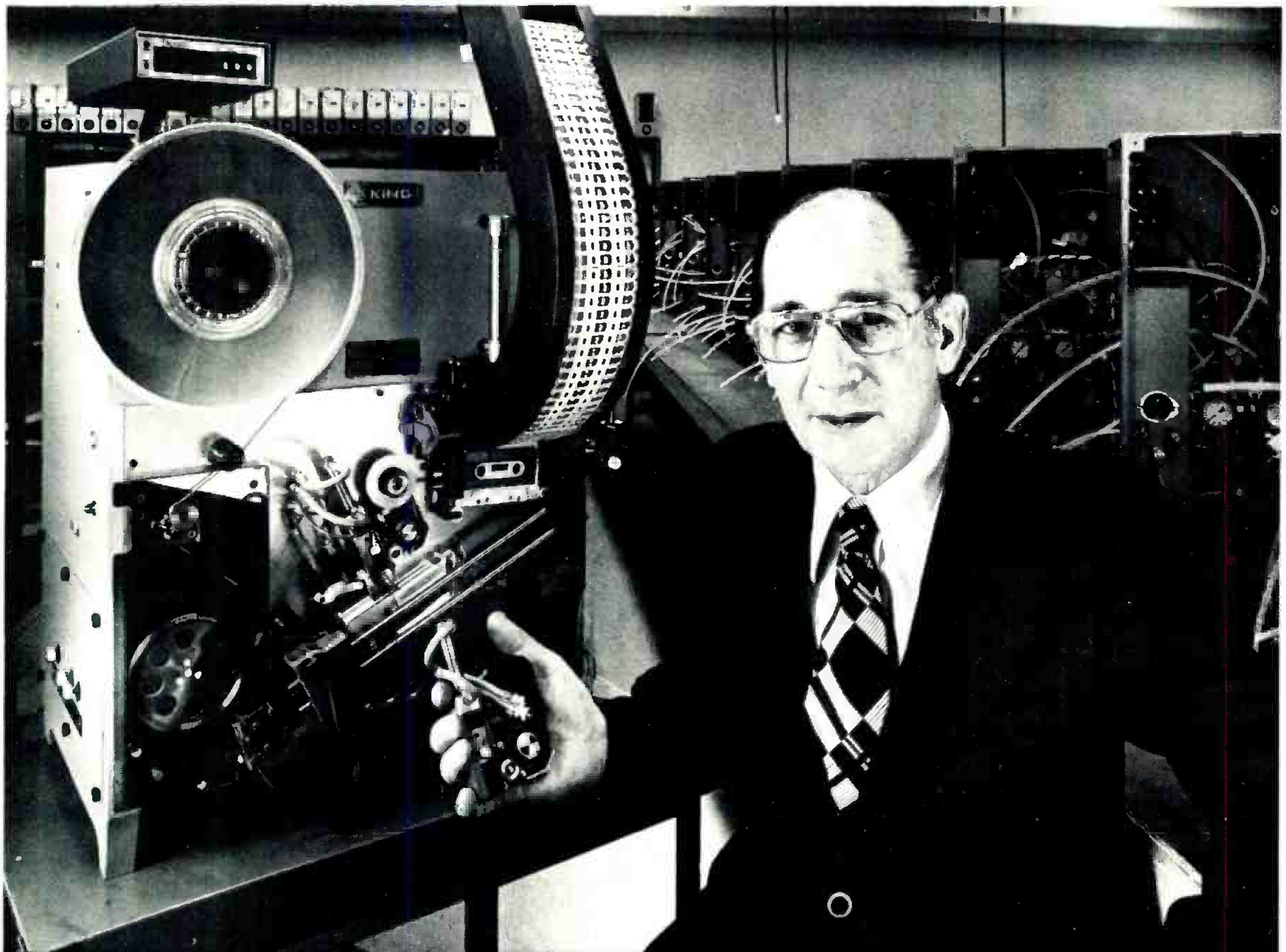

MIXING CONSOLE

• A sixteen channel mixing console, the 1642 features submaster mixing, 4 bands of EQ, multiple monitor and echo sends, phantom power, four separate echo/line channel returns, priority solo circuiting, headphone monitoring, meter switching and access to all busses. Specwise, the 1642 maintains a frequency response of ±1 dB, 15 Hz to 33 kHz; thd, below 0.02 per cent; IM distortion, below 0.05 per cent; slew rate, greater than 10 v/ms.; and a signal-to-noise ratio better than -80 dB (-50 dB input/1 volt output).
Mfr: Biamp Systems, Inc.
Circle 60 on Reader Service Card

MICROPHONE MOUNT

• Providing more than 20 dB mechanical isolation, TENSIMOUNT is a universal microphone mounting and isolation system which accepts any microphone up to 1⅜ inches in diameter. Allowing instantaneous switching of microphones without altering stand set-ups, the TENSIMOUNT converts the microphone to fit a standard ¾-inch mic clamp. The mount utilizes easily replacable, inexpensive elastic elements which can be color-coded for easy microphone identification in the studio and on location.
Mfr.: Brewer Instruments
Price: $9.95
Circle 61 on Reader Service Card

MICROPHONE

• Designed to be virtually invisible when placed on altars, lecterns, tables and in other settings, the SM18 cardioid dynamic microphone is available encased in either a white or brown foam "envelope" with cable colored to match. The microphones are angled at 10 degrees inside their foam "envelopes" to place the cartridge approximately an eighth of an inch from the surface on which they are set. As a result, surface reflected sound waves and direct sound waves reach the microphone at nearly the same time for highly intelligible voice reproduction.
Mfr: Shure Brothers Inc.
Price: $63.00
Circle 62 on Reader Service Card

DIGITAL RECORDER & ELECTRONIC EDITOR

• A PCM digital audio recorder, the X-80 series is completely computer-controlled, with full tape-cut and electronic editing facilities available. The X-80 offers a dynamic range of over 90 dB, less than 0.02 per cent distortion at peak levels, frequency response of 20 to 20,000 Hz (±0.3 dB), and complete freedom from wow and flutter. Tape editing is possible either by tape-cut or with the XE-1 electronic editor, providing automatic editing by SMPTE code. The computer-controlled editing unit has fade-in and fade-out functions and a digital level fader provides adjustable recording levels.
Mfr: Mitsubishi Electric Corporation
Circle 63 on Reader Service Card

X-80 Digital Recorder

XE-1 Electronic Editor

TAPE RECORDER

• The ATR-124 24-track analog audio recorder employs a closed-loop d.c. servo transport providing constant tape tension at each reel in all operating modes without pinch rollers. Transformerless I/O capability eliminates annoying distortions while offering excellent frequency response. Other features include: a variable speed shuttle control, adjustable from slow to 300 in/sec.; 16-inch reel capability; programmable monitoring with memory and a battery-powered backup memory that retains set-up instructions in the event of a power failure; four assignable record, playback and Sel Sync equalizers per channel; and single point search-to-cue with tape looping. The I/O bus capability of the ATR-124 provides evaluation of each channel without continual moving of the I/O cables. Optional features available on the ATR-124 are: a multi-point search-to-cue, designed to replace the standard single point search-to-cue, providing a capacity of 99 memories; and a complete remote control panel, offering all the functions available on the main panel.
Mfr: Ampex Corporation
Circle 64 on Reader Service Card

MODULATION CONTROLLER

• Compatible with any stereo generator/exciter combination, signal enhancement device or format, the Sta-Max is a wideband modulation controller that produces maximum loudness for f.m. stereo signals with complete elimination of overshoot or distortion. The unit is completely solid-state, and uses BNC jacks for all connections to and from the unit.
Mfr: Automated Broadcast Controls
Circle 65 on Reader Service Card

LOGGER

• An open-reel, slow-speed logging recorder reproducer, the model 230L offers a three motor tape transport with 2-speed (15/16 and 1⅞ in/sec.) operation. The logger is available in one-, two-, or four-channel configurations with professional solid state record/reproduce preamplifiers. Well suited for broadcast logging, the model 230L records over 12½ hours of information on 3600 feet of tape at 15/16 in/sec. Push button control with relay-type logic permits full remote control capability.
Mfr: Telex Communications, Inc.
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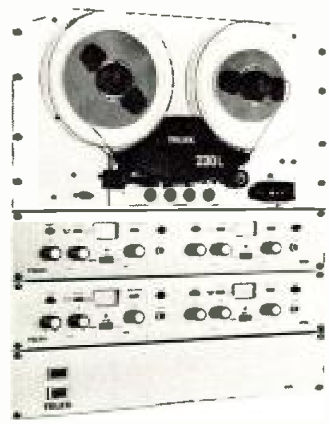
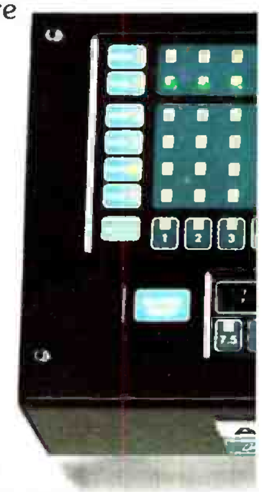


Introducing a present

Once you go through a recording session with the new ATR-124 24-channel recorder by Ampex, you'll want to go through another. Because with each new session you'll discover something new you can do. Things that you can only do with a recorder that's full of features of the future.

ATR-124 gives you the unheard of: Time on your hands.
Which means you can use that time to give clients more of what they're paying for—your creative skills. With the ATR-124 microprocessor-based control system, you can pre-program what you want to do ahead of time so you won't waste studio time setting things up. When their time starts, you're ready to record by touching a single recall button.

ATR-124 also lets you duplicate a technique you may have used earlier in the session without having to rethink what you did. Just touch the memory button and it'll all come back to you. ATR-124 lets you rehearse what you've got in mind, without recording it, to make sure what you've got in mind is right. Tape can be manipulated faster which means you'll get the sound you want sooner. And the chance to try something "a little different." All because of the speed and accuracy that ATR-124 puts at your fingertips.

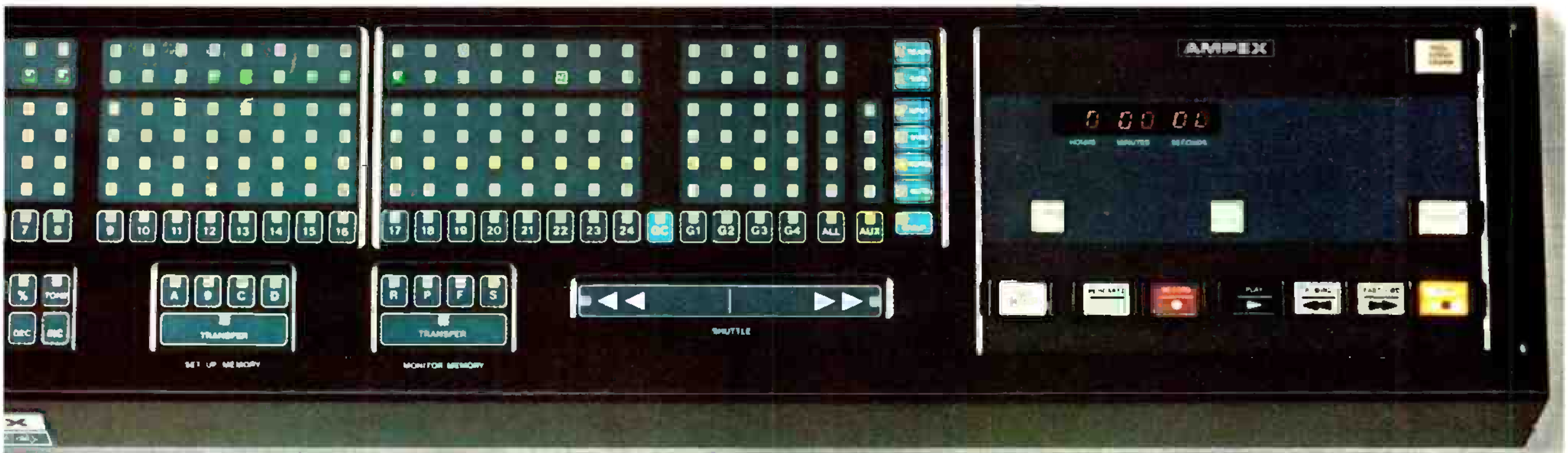


ATR-124 doesn't take away your creativity, it adds to it.
The less time spent setting up, correcting, and redoing, the more time spent creating. And when you add features that help you create to the ones that help you save time, you've got one very potent piece of audio machinery. Take the control panel for instance. It's like nothing you've ever seen. Pushpads linked to a microprocessor give you a new level of creative flexibility. Program a setup, then change it. Then change it back, all with a single fingertip.

A repeatable, variable speed oscillator for pitch correction and special effects is built in. In addition

from the future: ATR-124.

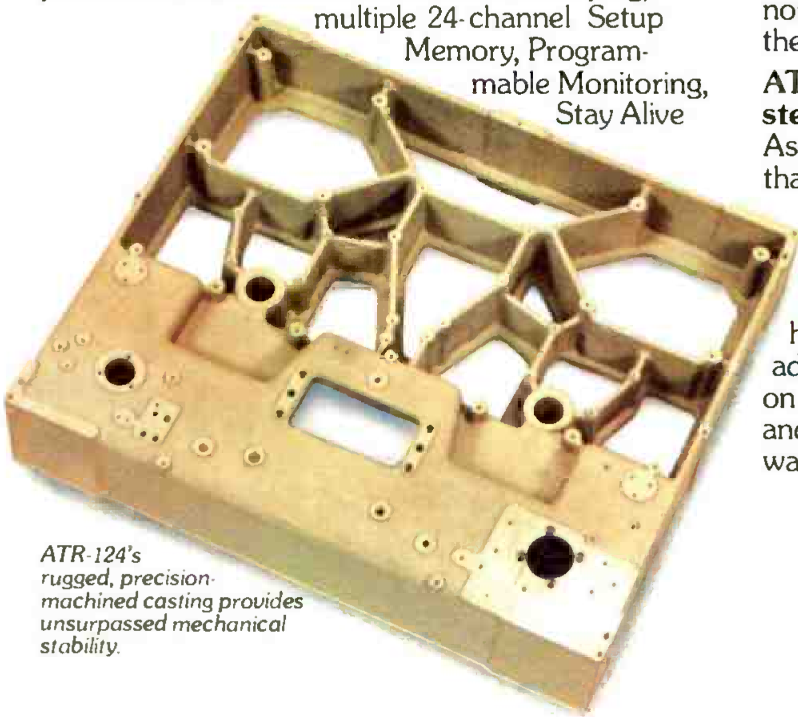
to the standard output, there is an optional auxiliary output with each channel that enhances flexibility. So don't think that ATR-124 is going to



ATR-124's Control Panel. Speed and accuracy at your fingertips.

replace anything that you do. On the contrary, it's going to improve the skills you have, if not help you develop some new ones.

ATR-124 picks up where ATR-100 leaves off.
It's only natural that the people who brought you the ATR-100 should be the ones to bring you something better. ATR-124 offers you 24 channels instead of 4. You also get many new and exclusive features. The kind that have set Ampex apart from the crowd for the last 30 years. Features like balanced, transformerless inputs and outputs; a patented flux gate record head; 16" reel capability; input and output signal bus for setup alignment; membrane switch setup panel; fingertip-operated shuttle speed control; and microprocessor-based synthesized Varispeed -50% to +200% in .1% steps or in ¼ tone steps. ATR-124 also features microprocessor-based control of Channel Grouping, multiple 24-channel Setup Memory, Programmable Monitoring, Stay Alive Memory, and Record Mode diagnostics. The point is this: If you like the ATR-100, you're going to love working with the ATR-124.



ATR-124's rugged, precision-machined casting provides unsurpassed mechanical stability.

ATR-124 options.
As impressive as the ATR-124 itself.
With the addition of a built-in Multi-Point Search-To-Cue (MPSTC), you can rehearse edits and control five tape-time actuated events and be compatible with SMPTE time code. Separately controlled auxiliary output amplifiers with each channel provide simultaneous monitoring of normal and sync playback as well as all other monitoring modes. A roll-around remote control unit can also be added to the ATR-124 which contains all control features normally found on the main unit.

ATR-124's Multi-Point Search-To-Cue (MPSTC). Provides 100 cue locations.

ATR-124. Your next step is to experience it firsthand.
As you scan the points we've covered, remember that you're scanning just a small portion of ATR-124's story. We haven't even begun to discuss the accessibility of key components for easy servicing and minimal downtime, or the features we've built in to give you greatly improved tape handling. To find out more, write to us at the address shown below. We'll send you a brochure on ATR-124, our latest audio effort. Better yet, call us and we'll set up a demonstration. It's really the only way to listen to the future.

ATR-124. Pure 24-Channel Gold From Ampex.

AMPEX

Listen to the future

Ampex Corporation, Audio-Video Systems Division,
401 Broadway, Redwood City, California 94063 415/367-2011

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db Editorial

ANALOG AUDIO—someday, historians may look back on it as the primitive science that got the audio industry through most of the twentieth century.

For the moment however, it's still very much a part of our present, and it's likely to remain with us for some time to come. Although it's risky to make predictions about the directions of future technology, we'll risk a bet that the microphone, at least, will remain an always-analog device. The prospect of little A/D converters hiding in every mic housing seems unlikely, if not downright impractical. (Not impossible—just impractical.) Of course, we can hedge our bet by guaranteeing that digital microphones will surely arrive before digital loudspeakers.

To get our analog issue underway, we offer two new looks at microphones in the recording studio. The Pressure Zone Process is about as radical a departure from the "we've always done it this way" style as we've seen in quite a while, and Ed Long gives us a first-hand glimpse at what it's all about.

And, from Bruel & Kjaer, Philip White suggests trying out some instrumentation microphones on your next recording session. Sharp-eyed readers will spot a slight difference of opinion between our authors, as to what effect a microphone has upon its environment. (In fact, we usually spend most of our time worrying about what effect the environment has on the microphone.) A free subscription to the first few readers who discover what we mean.

Next, a db Special Report on the new ATR-124: a firm comittment from Ampex for at least one more generation of analog audio tape recorders. It's interesting to see that some of the functions usually associated with console design have found their way into the ATR-124's control system. And, if we remember our August issue, we recall seeing some tape recorder functions appearing on the latest generation of recording consoles.

We'd be willing to bet that eventually, the tape recorder and the console will merge into a single recording "system," in which one control panel handles all the chores, from gain riding today's sessions, to playback of last month's. Once that happens, digital technology should really come into its own, especially with a new generation of digital signal processing devices thrown in for good measure.

In the meantime, given an analog tape recorder whose flexibility is pretty much a function of creative software, it seems we can enjoy the best of both worlds: the time-proven simplicities of analog, and the high-technology power of digital.

Old timers may remember when, once upon a time, you had to go to New York or Los Angeles for those really important recording sessions. Anywhere in-between was nowhere. But nowadays, any city big enough to have its own post office can have its own recording studio. And probably attract a fair share of clients who have had quite enough big-city hassle. Recently, Irv Diehl came from Alabama, with his note pad (no banjo) on his knee (actually, it was in his brief-case, but...). He tells us about what happened in Muscle Shoals, Alabama when four local musicians got tired of seeing the really big sessions get away.

Although level-indicating read-outs for music will always remain analog (remember what the word really means?), digital electronics are now being seen behind the face plate. So, to make the transition between analog and digital, this month we look at the analog side of metering as Sidney Silver compares the characteristics of VU and peak-reading meters, and, in our forthcoming digital issue, we'll examine the digital electronics that may be driving the meter movement.

And finally, we have Barry Hufker's backward glance at some microphone designs that didn't quite make it much beyond the drawing board. Although it's doubtful that the sausage-skin diaphragm will ever stage a come-back, the ribbon microphone is certainly nowhere near passe yet, thanks to some innovative designs from Beyer and Shure. (Are we forgetting any others?)

While we wait for the eventual era of the all-digital recording studio to arrive, let's not under-estimate our old friend Analog. After all, our ears are still analog, and they don't seem to show much sign of changing. Still, with the slow pace of evolution, who knows...? ■

EDWARD M. LONG

The Pressure Recording Process

A novel answer to an old recording dilemma.

HAVE YOU EVER wondered why, if the microphones are placed at a distance from an orchestra, the resultant recording sounds dull? Yet, when you attend a live performance you find that, although you may be seated out in the audience at some distance from the orchestra, the balance of the overall sound can be quite good. Why should this be so?

Have you ever adjusted a graphic equalizer to obtain a flat response from a loudspeaker in a room, using a spectrum analyzer and a calibrated microphone, only to find that the resultant sound was judged to be too bright? Have you ever thought about what might be causing the difference between what we hear and what the microphone picks up?

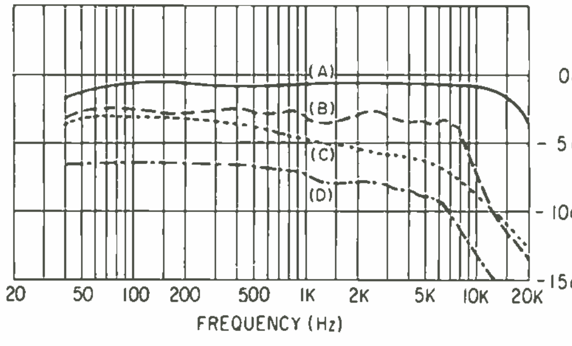
The answer—while quite simple—is not at all obvious, and a brief exposition of the problem would seem to be worthwhile. When a microphone is placed in a sound field, its physical presence in the field causes a pressure increase, which is frequency- or wavelength-related to the size of the microphone. This pressure increase is maximum at a frequency whose wavelength is about twice the dimension of the microphone diameter. Most studio microphones have diaphragms of one-half inch or greater, so the pressure build-up will cause a rise in response inside the audible band. Since all good microphone designers know this, they compensate for the effect, so that the microphone will have a relatively-flat response characteristic for zero-degree incidence sound.

RANDOM-ENERGY-RESPONSE

However, this means that the random-energy-response of the microphone must necessarily be rolled off at the higher frequencies. If you place a microphone close to a source of sound and then start moving it away, the spectral balance of the direct, or zero-degree, incidence sound, is not greatly changed, even for distances as great as 50 or even 100 feet. However, the ratio of the direct-to-random incidence sound, due to the reverberant field, is decreased. This means that, as the microphone is moved away from the source, the spectral balance becomes more dependent upon the random-energy-response characteristic of the microphone. But remember, that while the microphone's direct-sound frequency response is now relatively flat, its response to random-incidence sound has been allowed to roll off by the microphone designer, to compensate for the rise in response caused by the microphone's presence in the sound field, as previously described. The resultant overall spectral balance is the combination of the response of the microphone to direct sound, due mainly to the zero-degree incidence response characteristic, and to reverberant sound due to its random-energy-response characteristic. Therefore, it is plain to see that as the microphone distance from the source is increased, the spectral balance of the sound will be perceived as increasingly dull, until a point is reached where the random-energy-response of the microphone predominates.

FIGURE 1 compares a microphone's direct and random-

Figure 1. Direct and random-incidence response of a microphone. (A) zero-degree incidence response, (B) random-incidence response, (C) actual room curve due to absorption of high frequencies, (D) total sound as picked up by a distant microphone.



Ed Long heads the independent audio consultation and design firm of E. M. Long Associates, Oakland, California. Mr. Long is the co-inventor, along with Ron Wickersham, of the Pressure Recording Process.

incidence response. The absorption of the higher frequencies in the reverberant sound also has an effect upon the final spectral balance, but it should be remembered that this effect is also present for a listener at the microphone position, and does not play a direct part in the perceived discrepancy between live audition and microphone pickup with which we are presently concerned.

Now, what about the previously-mentioned discrepancy in the room equalization situation between equalizing a loudspeaker for flat response in a room, and perceiving the resultant spectral balance as being too bright? The answer is that, in equalizing the total sound to be flat, the rolled-off random-energy-response of the microphone has been compensated for as well. Of course, the microphone should not introduce an error in the spectral balance, but it does. This error causes the resultant equalization to be incorrect, with the higher frequencies being boosted too much.

FIGURE 2 shows the amplitude-versus-frequency response of two microphones designed for flat response under two different conditions: zero-degree incidence and random-incidence sound. It can be seen that each microphone, while providing a flat amplitude-versus-frequency characteristic under one condition, gives quite a different result for the other condition.

In the past, these effects upon the amplitude-versus-frequency response at high frequencies were not widely understood. With the great strides being made in recent times, in other areas of the recording and reproduction of sound, it seems appropriate to consider these effects. One of the main reasons for the development of the Pressure Recording Process was to obtain a method of transducing acoustic energy into electrical energy without discriminating between direct and random-energy sounds. How can a microphone be operated in such a manner that it will respond to the direct and random-incidence sound without discriminating with regard to their respective spectral balance? Looking at the response characteristic curves of the pressure-type microphone shown in FIGURE 2 suggest that it might be suitable if the zero-incidence sound could be eliminated.

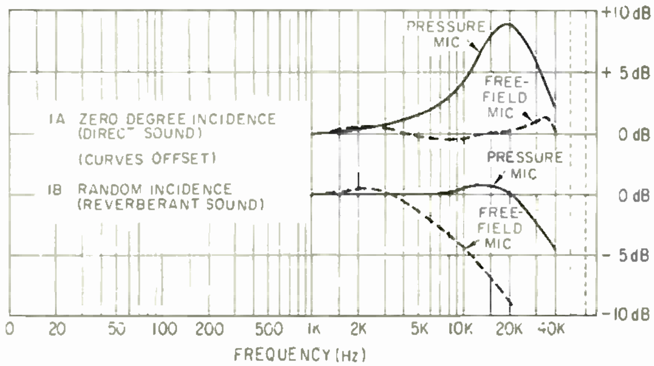


Figure 2. Response of half-inch microphones designed for different conditions to direct and random-incidence sound. (After B&K)

PRESSURE VERSUS FREQUENCY

The essence of the Pressure Recording Process is the use of a true pressure-versus-frequency microphone, which has flat response for random-incidence sound (or at least a smooth roll-off at higher frequencies, which may be equalized), and the positioning of this microphone diaphragm close to, and parallel with, a surface—preferably a major first-order boundary, such as a floor. In the case of an interview, meeting or discussion, a microphone designed according to the Pressure Recording Process might be placed upon a table top. Of course, once the essentials of the Pressure Recording Process are understood, various microphones and microphone uses can quite easily be developed.

Placing the microphone diaphragm, or other major entry port, parallel with, and close to, a boundary surface effectively causes the microphone to respond to pressure variations at the boundary, caused by both the direct and random-incidence sound, without discrimination with respect to their spectral balance. This results in the sound, as picked up by the microphone, having the same spectral balance as the natural sound perceived by a listener. The microphones designed according to the Pressure Recording Process may be placed at a distance from a source of sound without the results sounding dull. Recordings can be made which sound natural and true-to-life, because the spectral balance of the direct and reverberant sound is the same as it is naturally. The Pressure Recording Process also allows loudspeakers in rooms to be equalized correctly so that the resultant sound is perceived to be flat. Other uses come to mind immediately, as soon as one grasps the significance of this truly unique recording development.

The Pressure Recording Process may appear, superficially, to be similar to methods described in the past, but a closer scrutiny will reveal that it is different from anything previously proposed. There is a great temptation to consider the Pressure Recording Process as something quite simple, and therefore to overlook its tremendous potential for solving some of the heretofore seemingly insolvable problems encountered in day-to-day work with sound.

NEW MIC TECHNIQUES

During the past, many different techniques for using microphones have been developed. These generally-accepted techniques are based upon the use of microphones having different characteristics, such as omnidirectional, bi-directional, uni-directional, etc. These techniques could be considered conventions. Since the Pressure Recording Process is really something new, it follows logically that new techniques will be developed for its effective use. These new techniques will mean breaking past conventions for the placement of microphones. We are so used to seeing microphones up on stands or hung from ceilings, that it will be hard, at first, for practitioners of the art, to force themselves to throw off the restraints of these old conventions and take full advantage of the Pressure Recording Process. At least for a while, we will probably notice some strange and interesting microphone arrangements, as people experiment to achieve the best results. One set-up, which was eminently suitable for the situation at hand, was visible on National television recently, during the Papal Mass in Washington, D.C. A Wahrenbrock PZM microphone was mounted in the center of a two-foot square plastic baffle, suspended from an overhead boom, to pick up the choir. At the wavelengths involved, the two-foot square dimensions were adequate to capture the full spectrum of the choral sound, while allowing low frequency noises to be diminished. Wahrenbrock microphones were also used on the altar during the Papal Mass. The winds, which were 20 mph and more, had hardly any effect upon these microphones, while it was obvious that other microphones were effected greatly. The Wahrenbrock PZM microphones are built under the Pressure Recording Process license.

MICROPHONE PLACEMENT

The most difficult aspect of using the Pressure Recording Process effectively will be the overcoming of conventions for microphone placement, which have developed over the years. For instance, the Pressure Recording Process allows the microphone to be placed on the floor in front of a performing group or other source of sound with simply-amazing results. Placing a microphone in this position brings looks of disbelief from musicians and possibly even a comment such as, "That's crazy! Nobody listens on the floor." The usual placement of microphones up over a musical group, in order to increase the brightness of the sound, as picked up by conventional

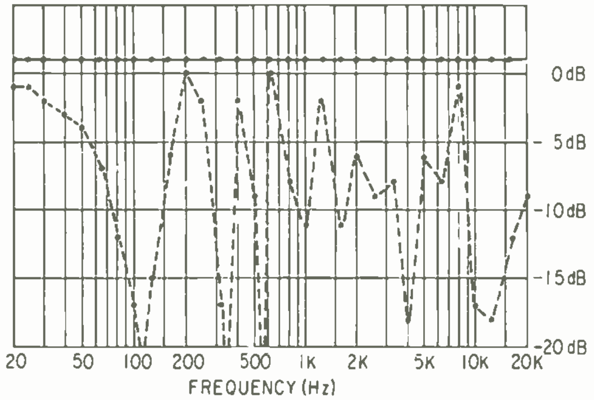


Figure 3. Computer derived amplitude vs. frequency response for conventional and Pressure Recording Process microphone placements. Dashed line represents conventional microphone placement 18 feet up and 10 feet in front of a source located 3 feet above a stage floor; Solid line represents Pressure Recording Process type microphone placed on the stage floor 10 feet in front of the same source. The roughness in response of the sound picked up by the conventional microphone technique is caused by the addition and cancellation between the direct sound and the major first-order reflection from the stage floor.

microphones, which are subject to the effects of discrimination between the direct and random-incidence spectral balance described earlier, is not even thought of as strange anymore, because it is conventional. A benefit of the Pressure Recording Process, which is possible because it requires that the diaphragm be placed parallel with, and close to, a boundary, is the elimination of the bad effect upon the frequency spectrum due to the interaction between the direct sound and the major first-order reflection. This effect has been well-documented in the past and may be the cause of confusion on the part of some persons, as to what the essence of the Pressure Recording Process really is. (See references 1, 2, 3) FIGURE 3 shows a computer-derived response for a conventional sound pickup, of a single source of sound, with the microphone suspended 18 feet up and 10 feet back from the source. The source is 3 feet up from the major first-order boundary—in this case, a stage floor. The dashed curve is the amplitude-versus-frequency spectrum of the sound caused by the alternate additions and cancellations between the direct sound and the major first-order reflection from the stage floor. It seems unbelievable that the direct sound and first major reflection can result in such an uneven response. The only reason that it becomes tolerable in real life is that first, it is a natural phenomenon to which we are exposed daily, and second, the total sound will be the result of the direct sound and a number of reflections, all arriving at later times, which tend to "fill in the holes" so to speak. The effect upon the spectral response, caused by these delayed sounds, can be eliminated by using the Pressure Recording Process, thus increasing the smoothness and clarity. The solid line in FIGURE 3 shows the response for a microphone designed for the Pressure Recording Process placed 10 feet away from the source, as before. The source is still at a height of 3 feet above the stage floor. An article dealing with some of the problems encountered in stereo recording, are discussed in another article.[4]

Any truly-new process should be able to pass two tests. Does it provide an end result which is different than that previously attainable? Is it basically different from anything previously available? Before the Pressure Recording Process, the combination of (1) uniform spectral response to both direct and random-incidence sound and (2) the elimination of the deleterious effects upon the amplitude-versus-frequency spectrum caused by the major first-order reflection, at the same time, across the entire audible spectrum, was not possible using any known technique.

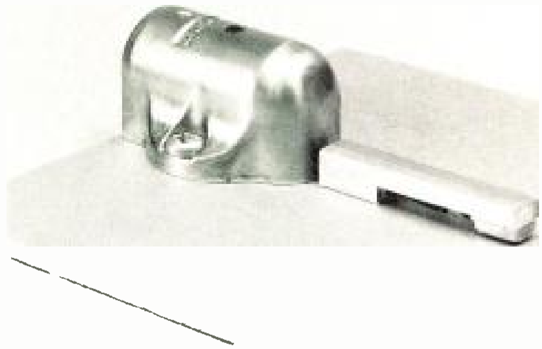


Figure 4. Wahrenbrock's Pressure Zone Microphone.

FIGURE 4 shows the Wahrenbrock microphone, mentioned earlier. There are now two different models, with different configurations and powering options available. At present, these microphones are being used around the world.

It is sincerely hoped that other microphone manufacturers will soon offer microphones designed specifically for use with the Presssure Recording Process. The talent and ingenuity of the world's great microphone designers and manufacturers, should result in some significant advances being made possible in both sound recording and acoustical measurements. ■

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The success of the Pressure Zone Microphone among experienced recording engineers has led to the involvement of a major audio manufacturer. Crown International, in conjunction with E. M. Long Associates, Synergetic Audio Concepts, and Wahrenbrock Sound Associates Ltd., is completing a development program to fulfill an exclusive license to manufacture and market a complete line of Pressure Zone Microphones. The target date for delivery is April, 1980.

PHILIP WHITE

Distortion – Measuring Microphones For Music Recording

The first-rate performance characteristics of air condenser measuring microphones make them most desirable for high-quality studio work.

NOW THAT CONSUMERS have much more accurate reproduction equipment, it means, of course, that recording equipment has to follow suit. And one of the most important components of the recording equipment chain is the microphone. If the microphone distorts the signal in any way, the fidelity of the signal is lost forever. But before we look at potential types of distortion originating from microphones, let's define some general categories of distortion.

LINEAR DISTORTION

In this category we find:

1. Frequency Response. If it isn't flat, it means that steady-state tones will not be recorded/reproduced correctly, with respect to their levels. Put this into perspective by considering slowly-moving music, such as an organ playing chords. If the different tones of the chord are not reproduced at their correct levels, the timbre will suffer.

2. Phase response. This relates to the relative time difference of individual sines of a complex signal. If a steady-state sound is fed into a speaker with a flat frequency response and a poor (non-linear) phase response, no apparent deterioration of the sound will take place. If, however, transient sounds from percussion-type instruments, pizzicato strings, the onset of horns—or for that matter, any transient musical sounds—are reproduced by this speaker, a non-linear phase response will change the apparent timbre. In addition, poor phase response will produce—in stereo applications—confusion of the stereo image, time smear and distortion of distance perception, and will give a wrong impression of the acoustics.

Look at it this way: A triangle is struck, and it produces a transient sound consisting of a fundamental and four harmonics. If there is a phase error in the recording or reproduction equipment, time smear will be produced and it will sound as if the transient tone is originating from five different points in space. We can make an analogy in photography: even if a camera records the intensity and colors (frequency response) correctly, the picture can still be blurred if distance hasn't been set properly. This is similar to music: If the phase response is poor, the music will be "out of focus."

Philip White, of Bruel & Kjaer, Denmark, specializes in acoustical and electro-acoustical applications and is currently working out of the Cleveland-based subsidiary, B & K Instruments, Inc.

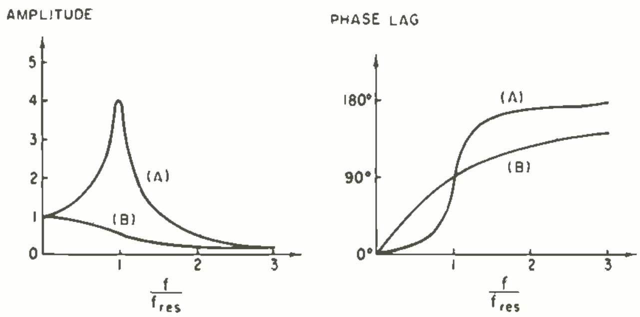


Figure 1. Frequency and phase response of (A) diaphragm with little damping, (B) diaphragm with large amount of damping.

NON-LINEAR DISTORTION

While linear distortion does not add any tones to the sound, non-linear distortion adds harmonics or sidebands to the original signal. This type of distortion is caused by clipping, slew rate limitations, etc. There are three main categories of non-linear distortion:

1. Modulation Distortion. Examples of this are: modulation noise in tape recorders; rumble, wow, flutter and tone arm resonance modulating the music signal in turntables; wow and flutter in tape recorders.

2. Steady-State Distortion. This occurs when a steady sine fed into an audio component produces harmonics, or when two sines produce sidebands (difference frequency and intermodulation distortion).

3. Transient Distortion. This occurs when, for example, an amplifier is excited with a transient signal or by the onset of a steady-state signal, and slew rate limitations cause the amplifier to clip the signal (temporarily).

Now let us look at condenser microphones, specifically, and examine some of their foibles:

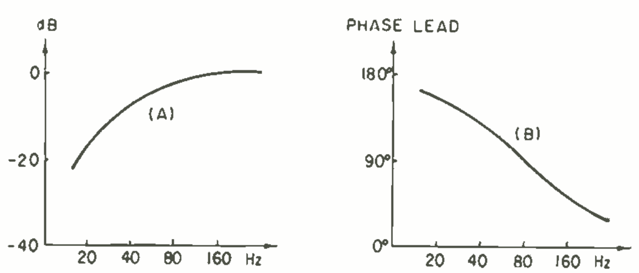
The microphone may (1) emphasize certain frequencies while de-emphasizing others (brilliance, coloration); (2) it may produce "pops" when high peak sound pressures occur; (3) it may produce distortion of "s" and "t" sounds and of high level sustained tones; (4) it may distort the signal waveform because of various phase errors.

Let us examine these four important items more closely:

FREQUENCY AND PHASE RESPONSE

On a pressure-gradient microphone (cardioid) the sound has access to the diaphragm both from the front and from the rear

Figure 2. (A) Typical low-end roll-off and (B) corresponding phase response for a condenser studio microphone.



(presssure-gradient principle). The frequency response is thus inherently rising at lower frequencies. At some medium crossover frequency, the response starts dropping, due to phase cancellation. Above this frequency the sound pressure, on some cardioids, will only excite the front side of the diaphragm; the access to the rear of the diaphragm is restricted by means of an acoustical low-pass filter consisting of various holes, slits and cavities (analogous to Ls, Rs and Cs). In other words, at higher frequencies the microphone behaves like a pressure microphone (omni) and derives its directivity pattern because of diffraction and shadow effects due to geometry. At a high frequency—where the wavelength of sound is equal to the diameter of the microphone—the response will peak up to 12 dB because of reflections. This peak can be brought down by introducing an appropriate amount of damping of the diaphragm.

The rising frequency response at lower frequencies is straightened out by locating diaphragm resonance at a medium-to-low frequency, typically at 1,000 to 2,000 Hz, and then flatten out this resonance peak by introducing a large amount of damping. Generally, a resonance peak will change the phase 180 degrees (lag). FIGURE 1 shows the principle of damping of a resonance peak and the corresponding phase curve. Note that the change in phase starts far below and ends far above the resonance frequency. At resonance the phase shift is 90 degrees.

The frequency response roll-off at the low end, constituted both by acoustical and by preamplifier electronics roll-off, will typically result in 90-degree phase shifts below 50 to 100 Hz, and will go toward 180 degrees at lower frequencies (see FIGURE 2). Imagine what this will do to the waveform originating from basses, bass drums, kettle drums, etc.

By now it should be clear that directional microphones (cardioids) of the pressure-gradient type are quite complicated devices exhibiting a relatively smooth frequency response. This smoothness, though, is obtained at the expense of the phase (transient) response because of (more or less) damped resonance peaks within the audio frequency range.

A simpler microphone is the pressure type (omni), seen in FIGURE 3. Here, the sound pressure excites the diaphragm from the front only. At low to medium-high frequencies, the diaphragm motion is stiffness-controlled, giving an inherently-flat frequency response. The phase response will be linear (ideal) at low frequencies, but will start to "run away" somewhat before the resonance frequency. Since the frequency response would normally roll off above diaphragm resonance, it is extended by means of pressure build-up in front of the microphone which, as mentioned before, peaks at a frequency where the wave length equals the diameter of the microphone. With this type of design, the resonance phenomena are moved to the high end of the frequency range, which should represent some advantage over the more complicated pressure-gradient microphones (cardioid) where the resonance is located right in the middle of the frequency range.

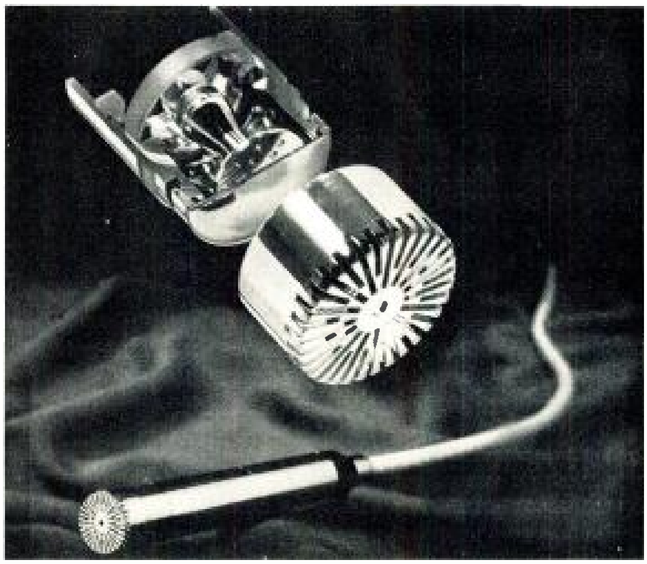


Figure 3. A Bruel & Kjaer 1-inch, free-field microphone, Type 4145.

DYNAMIC RANGE AND DISTORTION

Whereas dynamic microphones normally can not be overloaded in practical applications, condenser microphones can. The upper end of the dynamic range is specified to be where a certain amount of harmonic distortion is introduced; for studio microphones the 0.5 per cent limit is normally specified, whereas the 3 per cent or 10 per cent limit is commonly specified for measuring microphones.

When used in the recording of voices, pronounciation of certain words and sound creates a temporary huge static pressure on the microphone diaphragm, due to air flow, rather than traveling sound waves. This occurs with certain consonants because they are formed by quickly releasing air with the tongue, lips or teeth. Such a word is "pop," hence the name "pop" effect. This temporary huge static pressure will cause the condenser microphone to produce a high output voltage (several volts), causing the preamplifier electronics to saturate.

This is, in effect, a pronounced form of transient distortion. While the preamplifier is in, or close to, saturation, any sound—even of a small amplitude—will be distorted. As the preamplifier comes out of overload, the sound returns to normal. It is worth noting that the preamplifier normally will stay in saturation longer than it takes for the static pressure to subside.

To alleviate this, a high-pass filter in the preamplifier could be switched in, but then the phase response at low frequencies will be adversely affected.

Steady-state distortion occurs when, for example, voices or instruments produce high-level, sustained tones, and the preamplifier will clip the top off the waveform causing harmonic, difference frequency, and intermodulation distortion. The remedy here is to move further away from the microphone. Some microphone preamplifier combinations have a gain reduction switch to extend the upper end of the dynamic range while sacrificing the lower end because of a correspondingly higher noise floor.

Certain "s" and "t" sounds may, even though the sound level is not very high, produce an apparent high amount of distortion. The stronger subjective reaction to this might very well be because difference frequency distortion will in general be several times the level of harmonic distortion. Difference frequency distortion is also more objectionable than harmonic distortion, because some of the harmonics are inherent in the music anyway, while the difference frequencies are not. The higher level of difference frequency distortion, as compared to harmonic distortion, results from an inherent mathematical relationship, plus the fact that the electronics tend to roll off the harmonics.

WHY MEASURING MICROPHONES AS STUDIO MICROPHONES?

Let us first look at some of the desirable characteristics of measurement microphones and subsequently see how these

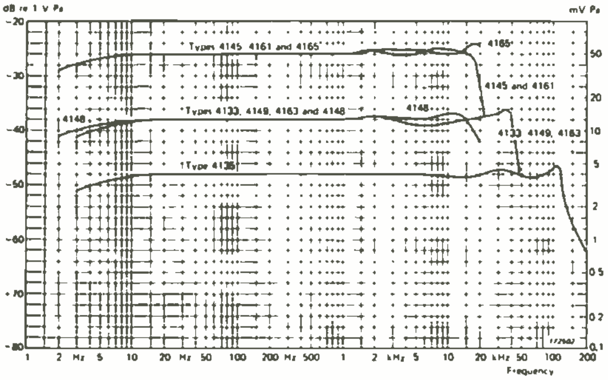


Figure 4. Typical 0 degree incidence frequency responses of the different free-field microphones.

characteristics match up very favorably to the requirements of studio microphones:

1) High long-term stability
2) Flat response over a wide frequency range
3) Directional response
4) High sensitivity, wide dynamic range and low distortion
5) Extensively documented

Precision-made, high quality Air Condenser Microphones meet the above requirements.

LONG-TERM STABILITY

It may not seem very important that the sensitivity of a studio microphone changes a couple of dB, or more. After all, can't you just change the gain in the console by the same amount? The fact is, however, that most microphone parameters, including sensitivity, are closely interrelated. A change in diaphragm tension will affect the sensitivity and, at the same time, the diaphragm resonance. So, the change in sensitivity will affect both frequency and phase response. The microphone no longer works in an optimized fashion.

The measured and extrapolated long-term stability of Bruel & Kjaer air condenser microphones is 1 dB change in sensitivity in 300 to 900 years, depending on type.

FREQUENCY AND PHASE RESPONSE

As seen in FIGURE 4, the frequency response goes from 2-4 Hz up to 18.5-120 kHz, depending on the type. The curves shown are typical: e.g. for the types 4133 and 4165 the recorded frequency response curve supplied with each individual microphone actually rises less than 1 dB at the high frequencies. The microphones and preamplifiers are designed such that the acoustical roll-off, as well as the electronic roll-off, coincide in the 2-4 Hz region. The resulting phase deviation will therefore only be around 4 degrees at 40 Hz, making these microphones ideal for picking up low frequency transient sound.

As mentioned before, pressure microphones (omni) normally have the diaphragm resonance towards the high end of the frequency range. Now take a look at the frequency response of the type 4133, which extends to 40 kHz. Due to the high tension and low mass of the diaphragm, its resonance frequency is outside the audio range at 23 kHz. This results in a deviation of approximately 40 degrees at 20 kHz from an ideal phase curve. This corresponds to a time smear of only 6 microseconds or, translated into spatial distance, with sound traveling at a speed of 344 m/sec, to a distance smear of only 20 mm. This will be of importance to voices and all instruments and, of course, specifically with respect to the accurate reproduction of cymbals and triangles, any percussion type instrument, guitars (especially acoustical models), etc.

When using this type microphone in stereo, multi-channel or differential applications, the phase matching between microphones becomes crucial in order to preserve the sound

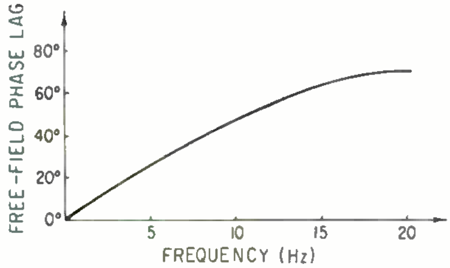


Figure 5. Free-field phase response of the half-inch microphone type 4133.

image. Any two of these microphones will match within 5 degrees. FIGURE 5 shows the free-field phase curve of a type 4133 microphone. The overall slope of the curve is not important because it is only indicative of the group delay—the time by which all frequencies in common are delayed. The 4133 comes quite close to an ideal phase curve, which is a straight line going through 0 degrees and 0 Hz.

A smaller microphone such as the quarter-inch type 4135 shown in FIGURE 4 has an even-better phase and frequency response (4 Hz to 100 kHz) and will be well suited for high sound intensity percussion instruments. Its somewhat lower sensitivity will result in low distortion (0.6 per cent at 160 dB SPL), but give a relatively high noise floor (48 dB(A) SPL).

DIRECTIONAL RESPONSE

For pressure microphones the directional response is governed by the diameter of the microphone... the smaller, the better. At low frequencies the pattern is omnidirectional, at higher frequencies more directional. For the type 4133 the off-axis response is uniform within 5 dB at 10 kHz (±90 degrees): for the type 4135 it is within 2 dB.

SENSITIVITY, DYNAMIC RANGE AND DISTORTION

The one-inch microphone (Type 4145) and half-inch microphone (Type 4165) have the highest sensitivity (50 mV/Pa $\approx$ 5 mV/μbar), resulting in noise floors corresponding to 13 and 18 dB(A) SPL. In general, studio condenser microphones will exhibit a dynamic range of 90 to 110 dB. Due to the high voltage swing capability of instrumentation preamplifiers, and the high-tension diaphragm of measuring microphones, a dynamic range of 112 to 125 dB is achieved (0.5 per cent distortion). The sound level where most studio microphones exhibit 0.5 per cent distortion is around 110 to 135 dB SPL. However, the type 4133 measuring microphone can capture sound levels of 152 dB with only 0.5 per cent distortion—an important characteristic when recording high levels and high peaks. The chance of overload and distortion due to "pop" sounds is very low. The overload point of, for example, the types 4133, 4135 and 4165 is approximately 170, 160 and 150 dB SPL, respectively.

DOCUMENTATION

Due to the fact that air condenser microphones are used in a wealth of routine and also in many difficult scientific applications, a very comprehensive amount of information concerning acoustical, electrical and environmental performance has been gathered (see Ref. 1). In addition, each microphone is supplied with its individual calibration certificate showing the actual frequency response curve and important electrical parameters. Besides being an important final step in the factory quality control procedure, it allows the user to make an accurate match to other studio equipment. For the user to verify the performance, a simple acoustical calibration will provide more information than at first apparent. A substantial change in acoustical sensitivity (more than 1-2 dB) in these microphones, which are highly stable in all parameters under normal use, would indicate that the microphone has been subjected to harsh treatment. Here we should keep in mind that sensitivity is closely related to frequency and phase response.

CONCLUSION

With the availability of higher quality professional recording and consumer reproduction equipment, the performance of studio microphones has become more important than ever. The public has become increasingly aware of good and bad recording techniques and several studios improve the recording techniques by going to direct-to-disc recording, digital tape recording and mastering. This places more emphasis on a most important item: the studio microphone. Air condenser measuring microphones have performance characteristics unequalled by other commercially available omnidirectional microphones, making them immensely suitable for high quality studio work:

Lower distortion at high SPL
Higher overload capability
Wider frequency range
Flatter response
Closer to an ideal phase (transient) response
High sensitivity

■

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db Special Report

JOHN M. WORAM

The Ampex ATR-124

As we enter what appears to be the digital decade of the 80s, Ampex is making a major commitment to at least one more generation of analog professional audio tape recorders.

IT SEEMS AS THOUGH everyone is introducing—or at least, announcing—the arrival of a digital tape recorder. Well, almost everyone. Lately, Ampex has been conspicuous for its total silence on the matter of digital recording. The recent ADD-1 (see our November 1979 issue) acknowledged what everyone suspected all along: Ampex—just like everyone else—was indeed getting involved in digital technology. And now, with a DDL in their product line, could a DTR be far behind?

At the recent convention of the Audio Engineering Society, Ampex finally broke the silence, by unveiling its new ATR-124 tape recorder. And, it's analog!

Ampex seems to have carefully evaluated the status of digital technology today, and concluded, "Not yet." Although others will surely disagree (as we'll see next month), the company proposes one more generation of analog machines. Of course, the new generation takes full advantage of the ubiquitous micro-processor, but the recorded format remains, for the moment, analog.

The usual specs are of course impressive, even if wow and flutter is still measureable, and signal-to-noise is not quite up to digital standards. However, the system electronics have been designed with enough latitude to accommodate future advances in tape formulations. And these will surely narrow—although perhaps not close—the S/N gap. (As an aside, there doesn't seem to be a professional application for the new metal particle tapes, since their advantages are most apparent at the very-short recorded wavelengths found on cassettes.)

Perhaps the most-noteworthy departure from earlier Ampex designs is found within the ATR-124's series of control systems. These are divided into separate sections: a channel-status module, a transport controller, an equalization selection panel, and a multi-point search-to-cue system.

CHANNEL STATUS MODULE

A 29-by-7 matrix provides control over the condition of each track. The usual modes (Ready, Safe, Input, Sync, Repro) are switched by first touching the appropriate mode switch, at either end of the matrix, and then touching one or more of of the channel switches. An additional Mute mode is also available.

There is also a Group mode, in which one or more channels may be assigned to one of four groups. Now, if Input and Group 1 are touched, all channels assigned to Group 1 will go into the Input mode.

MEMORIES

A Setup Memory allows four monitor conditions to be stored in Memories A, B, C and D. The Monitor Memory allows a different monitor condition to be selected for each transport mode. As an example, the following monitoring conditions could be programmed into the system: While recording, the system will monitor only those channels presently being recorded; during playback, all channels will be monitored; at stop, all channels assigned to record will switch to input; during fast forward or rewind, all channels except the time code track will be muted.

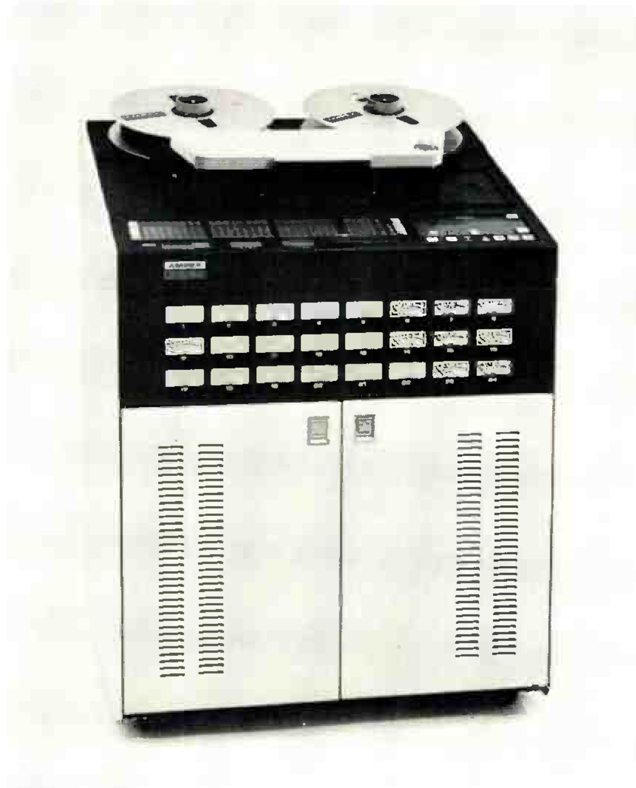


Figure 1. The ATR-124: Ampex's latest-generation analog multi-track tape recorder.

Depending on the synchronizer being used to read time code, it may be necessary to read the code continuously—or intermittently—during fast-wind modes. Usually, the tape lifter is under the control of the synchronizer, and may be permanently defeated when continuous time-code readings are required. Ampex feels that the air film that is produced between tape and heads during either fast-wind mode is sufficient to prevent head wear, although obviously, an audio-muting function is necessary to prevent demolishing the studio monitors.

Presumably, the mute mode could also be controlled from an external point—perhaps via the Group matrix—so that preselected channels could be switched on-and-off during mixdown, as is common on many automated consoles.

TRANSFER MODE

The Transfer mode prevents accidental erasure of previously-stored monitor conditions, and also allows the contents of any memory to be placed in one of the other memories as well. For example, to call up the contents of Memory A, simply depress A. To enter a new condition into Memory A, depress Transfer, A. To write condition A into Memory B, depress A, Transfer, B. This latter sequence can be helpful if a fairly-complex monitor condition needs some minor modifications from time to time. Instead of attempting a point-by-point duplication, simply transfer A into B (A remains as-is). Now, the monitor mods can be made to B.

TAPE SPEED

In addition to three-speed operation (7½, 15, 30 in/sec.) a crystal-controlled frequency synthesizer permits speed change over a 50-to-200 per cent range. Thus, the transport speed may be varied from 3¾-to-60 in/sec. Changes may be made as percentages of normal speed, or, in musical quarter-tone increments, over a range of ± one octave.

Figure 2. A roll-around pedestal containing the channel function and multi-point search-to-cue modules.

AUXILIARY CHANNELS

Some European tape recorders provide separate sync outputs, and, in a somewhat-similar fashion, the ATR-124's optional Aux channels provide a separate set of electronics which may monitor Input, Sync, Repro or Mute. Unlike the main outputs, all Aux channels switch modes simultaneously. Perhaps the major advantage here is to allow tape monitoring during punch-ins, by monitoring the Aux channels in the Repro mode (in the control room). Meanwhile, the studio musicians listen on the normal channels. At the punch-in, the sync monitor goes to Input, so the musicians hear themselves during the insert, in the usual manner.

MULTI-POINT SEARCH-TO-CUE

Of course, it would be difficult—if not impossible—for the engineer to do punch-ins while monitoring tape output, due to the delay between the sync/record and playback heads. Therefore, the just-described feature will no doubt require the use of the Multi-point Search-to-Cue module. This module can store up to 100 cue points in memory.

During rehearsal, the engineer monitors the normal channel, and—using the module's keyboard—cue numbers are assigned to several points on the tape, identified as: Cue Point, Record Start, Record End, and Search-Play. For example, assume the following times have been entered—3:27, 3:38, 3:50, 4:02. The system will start the tape at (Cue Point) 3:27, go into record (Record Start) at 3:38, drop out of record (Record End) at 3:50 and at 4:02 (Search-Play) will rewind back to 3:27 and repeat the cycle. Once cues are programmed, the engineer may switch to tape monitoring, as described above.

In the Auto Pre-roll mode, only Record-Start and -End points need be entered, and the machine will automatically cycle between pre-selected cue points before and after the recorded section. These points are set via a thumbwheel switch, between 0 and 99 seconds.

In the Rehearse mode, the system will simulate a punch-in/-out sequence, without actually going into record. If any of the cues need to be adjusted, they may be changed by frames or seconds. If any cue point is changed in error, the previous setting may be returned, simply by depressing an X/Y Transfer key, similar to the one found on most calculators.

EQUALIZATION SELECTION PANEL

Each audio channel has four complete record/playback networks, and these may be pre-set for various tape speeds, recording standards, tape formulations, fluxivities, etc. Thus, the machine may be pre-aligned for say, 7½ in/sec./NAB 3M tape, 15 in/sec. CCIR/Agfa, 30 in/sec./AES BASF and 15 in/sec. CCIR/Ampex, at "X" fluxivity.

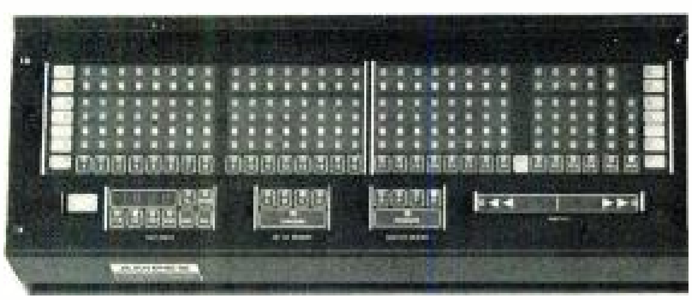
TESTING

Accessible from the front of the machine is an input/output bus, which may be switched to any audio channel. This allows easy access for alignment procedures, without tying up the console itself. It would seem to be a relatively trivial task to design some sort of automated testing system that could perform spot-checks on each audio channel, and perhaps even make corrective adjustments when necessary.

LOOKING AHEAD

Once a "super-mix" has been prepared, it would certainly be disastrous if all the cues were lost every night when the power was turned off. To prevent such calamities, a battery-backup for the memory will hold all cues for about three months, or until the session is over, whichever comes first. However, this won't be of much help when it becomes necessary to take the tape across town, or across the world, to another studio.

Figure 3. The Channel Status Module. A matrix of touch-sensitive membrane switches permits high-density packing of control functions. At the bottom of the panels are switch groupings for tape speed and VSO, plus set-up and monitor memories. The touch-sensitive shuttle bar may be used to rock tape back-and-forth. Position of the operator's finger along the bar determines tape speed in either direction.



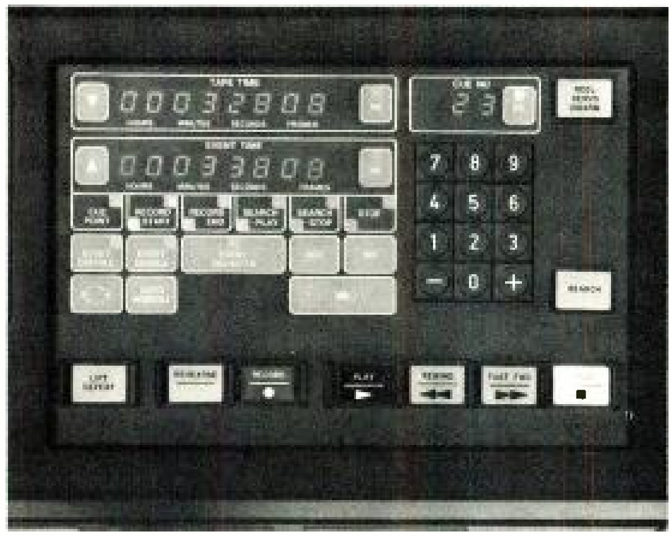


Figure 4. The Multi-point Search-to-cue module.

By touching one of three KE (keyboard) switches, the keyboard may be used to enter tape time, event time or cue number.

The six buttons immediately below the event time readout allow the operator to program the system for a variety of production chores.

Split-screen illumination of the tape transport switches indicates the actual mode of the transport, as well as the last command received. Thus, during decelleration from fast forward-to-stop, half of each button would be lit, indicating the machine has been commanded to stop, but is still in motion. As soon as the tape comes to a complete halt, both segments of the stop button are lit, and the fast-forward light is extinguished.

Again, it would seem to be within the realm of possibility to design some sort of memory-reading system, that would permit all 100 cues to be stored on a floppy disk—or perhaps, at the head of the master tape itself. That way, the cues could accompany the tape wherever it goes, freeing the machine memory for new tasks.

IN-STUDIO PERFORMANCE

It's still too early to report on the ATR-124's performance under actual recording studio conditions. But by now, Ampex's position in the pro marketplace has been well-established, and there's every reason to expect that the ATR-124 will match—and probably surpass—the performance of earlier models. With an electronics system based on the legendary ATR-100 Series, the system with no doubt attract the interest of studio-owners who wish to stay with analog audio for at least one more generation (of machines, that is—not of owners).

There's a certain amount of "leap-frogging" within the studio-to-home listener signal path, as various elements become the weakest link. Multi-track digital recording certainly removes tape from the weak-link category, but it doesn't do anything for lousy pressings or degraded f.m. broadcasts (see our recent editorials for more on this).

On the one hand, digital tape recorder technology is no doubt going to be a part of every studio's future. On the other hand, prices, proven reliability, and standards still stand in the way of an overnight swing to digital.

The question is no longer "If", but "When?". Everyone has an opinion, from sooner to later. The Ampex ATR-124 represents but one of those opinions. But—all things considered—it's a pretty strong opinion. ■

IRWIN DIEHL

Muscle Shoals Sound Studios

More than just a 24-track two-studio complex, M.S.S. has become one of the South's leading record production facilities.

WHAT'S THE MAIN INGREDIENT for a successful recording studio operation? Ask a dozen studio owners, and more than likely, you'll get almost as many different responses. Some will stress the large and steady flow of capital that's needed to outfit and operate the studio competitively, others may say contacts and friends in the business are the essential factor, and some might suggest long and hard hours are the key.

Certainly, each of these must be factored into any comprehensive success plan in the recording industry, and probably should not be ignored. However, the success plan for Muscle Shoals Sound Studios reveals one more interesting factor, not often found in studio management.

For more than a decade, the studio owners—Jimmy Johnson, David Hood, Roger Hawkins and Barry Beckett—have been known as the Muscle Shoals Rhythm Section. One record company executive regards them as the world's only remaining rhythm section from the 60s era.

In 1969, the four purchased a small 16-track studio on Jackson Highway in Sheffield, Alabama. Before long, the little studio became successful because its owners were—and still are—among the best studio musicians in the country. However, after producers got their Muscle Shoals rhythm sound, they would then go off to Nashville or New York to finish off the project.

Although business was good, it was not very satisfying to see so many projects leave town before completion. And so, it was soon agreed that what Muscle Shoals really needed was an all-new 24-track, two-studio complex that would attract producers, and then keep them in town right through the mixdown session.

The entrance displays the laurels of just a few of the Muscle Shoals' projects that turned to gold.

Irwin Diehl, co-founder of the Institute of Audio Research, New York, is currently employed as an independent consultant.

A view of the Studio A control-room.

One of Muscle Shoals' two mirror-image studios.

In April, 1978, Muscle Shoals Sound Studios moved to its present quarters, and, according to Jimmy Johnson, "We've taken down the 'tracks-only' sign. Artists and producers now come here to begin and finish projects." The studio owners continue to play on sessions, but are now also busy with writing, producing and engineering chores as well.

THE STUDIO COMPLEX

The "new" building is a 31,000 square-foot structure built at the turn of the century, originally designed to accommodate the City of Sheffield power plant. Somewhat more recently, it was used as a Naval Reserve Center during the second world war.

Snuggled into a hillside some fifty yards from an embankment of the Tennessee River, the building would be more than adequate insurance against outside noise and street rumble in any large metropolitan area. On its actual site, the structure is triple-indemnity against any hell-raising Tennesee River catfish.

While the outside setting and building skeleton might be considered an acoustical designer's dream come true, the new owners have devoted no less attention to inside construction and isolation details, as might be expected in far-noisier, industrial/metropolitan areas.

The Muscle Shoals facilities were designed and built by Claude Hill of Audio Consultants in Nashville, and includes two near identical studios, laid end-to-end with control rooms situated back-to-back. The outer walls are comprised of 10" and 12" concrete block, filled with sand to give added mass. Continuous voids or air-spaces separate walls of adjoining areas that must be acoustically isolated.

Wood two-by-fours are laid to the concrete block surface to provide studding for interior surfaces. Between studs, fiberglass materials are installed to reduce resonances that might result from otherwise-hollow cavities.

ISOLATION BOOTHS

The three drum and isolation booths built into each studio follow the "room within a room" principle, with carpeted, ¾"-plywood floors resting on 2" x 4" studs. The spaces between floor studding are also filled with a fiber batting material.

Isolation booth ceilings are approximately 8-feet high, secured to 2" x 6" joists, with pegboard and 18" fiberglass resting atop the entire ceiling structure. Double-paned windows in twin, acoustically-isolated walls permit visual contact with the audio while keeping sound leakage to a minimum.

The studio ceilings have been dropped to 18-feet and are backed with 6" fiberglass. between the dropped ceilings and building roof, an intermediate structure or sound barrier with 6" of foam insulation is added.

A variety of surface materials have been used to control the studios' acoustic properties. Carpeting, hardwoods and fabrics of various densities and textures have created a visually attractive space while maintaining a flexible sound environment.

Walls between studio and control room are of double construction. The studio wall is 8" block and the control room wall, 10". An air-space separates the two walls, which rise to the intermediate structure between roof and dropped ceiling. I-beams just above the control room windows support this heavy construction over the window framing.

Control-room ceilings are dropped to about 9-feet, treated in the fashion described previously. Fabric-covered sound traps

A Designer's View of M.S.S.

Our design philosophy has always been to design each of our studio projects from scratch, with a strong emphasis on client needs. We do not believe that one standard design can fit all needs and applications. As applied to M.S.S. this design philosophy is represented in the control-room shape, which follows the lines of their previous facility on Jackson Highway. I affectionately refer to the old studio as the "burlap palace," a name which is well deserved.

The other attributes of the control-room design are...

1. Adequate size and volume to accommodate any console befitting a "world class" studio.

2. A broad and deep monitor field designed to allow reasonable freedom of movement by producer and engineer at the console without compromise in monitor accuracy. Particular attention was paid to the phantom stereo imaging, as the clients use many panoramic and time domain effects in their mixes.

3. All equipment items were provided with locations that provide access for operation and maintenance while allowing full freedom of movement for all operating personnel and talent.

4. The AUDICON ALPHA ONE Monitor System was designed initially for the control rooms at M.S.S., as no commercially available monitor system could provide for the requirements of the clients.

5. Two identical "mirror image" control-rooms were built. Particular care was taken to assure full isolation between studios A and B and was achieved by the "back to back" configuration and the isolation walls constructed between. Both studios and control rooms exist on a common 16 inch steel-reinforced concrete slab, which has existed since the 1930s. There is no measurable or audible transmission through this slab.

6. As in the studio design, control-room reverberation time was calculated in the design, then accurately measured just prior to completion of construction and minor adjustments were made in the design to yield uniform RT60 of 45 milliseconds from 50 Hz to 8 kHz ±20 per cent.

The two studios were designed to yield identical performance. The single difference between them is the dimension between the control-room/studio wall and the wall which makes up the front of the isolation booths. The acoustical treatment of the smaller studio B was designed to duplicate the character of the larger studio A. Both rooms were adjusted during the final stages of construction to yield the same 1.5 second RT60 reverberation time over the 50 Hz to 12 kHz band. The high, 20-foot ceilings provide the volume of air which I strongly feel is necessary for achieving tonal quality of the classic orchestral instruments. The three isolation booths are of heavy double-wall wood construction and yield broadband isolation of more than 50 dB between booths and between booths and the studios. The booths in both studios are identical and have adjustable acoustic surfaces.

—Claude Hill

are built-into the control-rooms to control low frequency phenomena. Hardwood, carpeting and tile placed strategically around the room tailor acoustics in the upper frequency ranges.

Isolation between Studio A and Studio B is, for all practical purposes, 100 per cent, due to more than 3-feet of triple walls made of sand-filled concrete block, separated by 4" air-spaces.

This approach to studio construction suggests careful attention to proper design, as well as a concern with detail of construction. This theme established in the building stages was continued when it came time to equip the complex.

STUDIO EQUIPMENT

Both studios are fitted with Neve boards. Room A sports a 32-input Model 8068, and in B there is a 44-input Model 8088. MCI 24-track, 2"-recorders handle the multi-track tape duties, while Ampex and Studer share the ¼-inch mixdown chores.

Muscle Shoals Sound has everything the studios in larger metropolitan areas offer, plus a charming and comfortable setting. The combined studio activities of the Muscle Shoals, Alabama area do not yet require rental services such as those found in busier regions. However, the studio lacks nothing in the way of instruments, amplifiers, signal processing and special-request items. Because studio rental services are essentially non-existent, M.S.S. has purchased all the myriad of instruments and devices necessary to record production. Though this increases audio overhead by a small margin, a hidden advantage accrues to the client in that equipment normally rented is routinely provided and also routinely maintained along with all other studio equipment.

Muscle Shoals Sound has not only become one of the South's leading record production facilities but has gained recognition and respect throughout the U.S. Our report may have underscored a few of the more prominent reasons for their success. ■

SIDNEY L. SILVER

VU Meters vs Peak Program Meters

While VU meters provide a reliable indication of levels on mixed or integrated programs, peak-reading devices are useful in monitoring instantaneous peaks—giving clear warning of overload conditions.

As PRESENT-DAY MULTI-TRACK technology becomes more sophisticated, there is a need for more accurate metering of program levels, to keep audio signals within the limited dynamic range available within the recording system. Because of the stringent demands imposed upon the system by today's transient program material, it has become increasingly clear that program indicators with standard VU characteristics are inadequate for properly monitoring peak energy levels, which are the limiting factor in recording operations.

Normally, standard VU meters can be used to reliably indicate levels on mixed, or integrated programs, whose crest factors (peak-to-average values) are more-or-less predictable. But when individual musical instruments, whose peak energy content is many times normal, are recorded on separate tracks of a multi-channel system, VU meters can be very deceptive in detecting the onset of overload distortion. This problem has become more acute because of the growing trend to abandon

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.

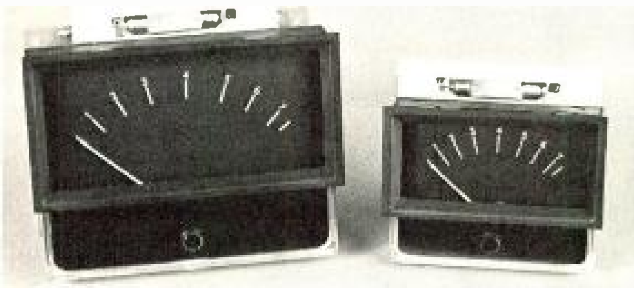


Figure 1. British Standard Peak Program Meter. (Courtesy Surrey Electronics)

"true" VU meters in favor of less-costly devices, which do not comply with standard performance requirements. Nowadays, it is not uncommon to find pseudo-VU meters in professional equipment whose sluggish response and high overswing can lead to substantial errors when adjusting audio levels.

In order to provide a reasonably accurate indication of transient peaks, and give a clear warning of overload conditions, peak-reading devices are now being used to supplement the array of VU meters in audio consoles and multi-track recorders. Peak program meters (PPMs) differ from the standard VU meter in that their ballistic behavior is electronically synthesized to yield a rapid and reliable measurement of the crest values of the audio signal. These instruments have long been favored for general use by European broadcasting and recording studios, and, to date, there are over half a dozen standards in existence defining PPM characteristics. As yet, however, no single PPM standard has been adopted in the US.

There are some commonly-held misconceptions regarding the relative merits of the VU meter and the PPM, and this has led to a great deal of controversy concerning the use of these two instruments. On the one hand, some engineers believe that the VU meter has outlived its usefulness and should be completely replaced by the PPM. Many others feel that both instruments working together can resolve the conflict that arises in virtually monitoring program levels, i.e., registering the instantaneous peaks accurately, while at the same time reading out the integrated program energy on a continuous basis.

As a basis for understanding the present situation, it would be helpful to briefly review the use of program indicators in past years, particularly with regard to their dynamic characteristics.

HISTORICAL BACKGROUND

In the early days of broadcasting and recording, there was a wide variety of audio level meters, with various speeds of movement, different calibrations and reference levels, and very little correlation between their readings. These old-style dB meters were very difficult to interpret, having a large amount of overshoot, with decibel scales extremely cramped at the lower end. To remedy this condition, broadcast and telephone engineers came together some 40 years ago and developed a new type of volume indicator—the VU meter—whose dynamic response was rigidly specified and controlled. Subsequently, the VU meter was adopted as an effective standard by most users of program indicators, but was not generally accepted by motion picture studios. Here, sound engineers working with optical recording systems preferred using a level indicator with a considerably-faster response. They complained that their light modulators could easily be overloaded by sharp peaks eluding the moving pointer of the VU meter, causing serious distortion. To provide a suitable peak-reading device for optical sound, a neon-light indicator was developed that was sensitive enough to register both positive and negative excursions of the light modulator. The unit consisted of a row of 15 neon lamps arranged to record program material in 3-dB steps, ranging from 45 dB to +3 dB with reference to 100 per cent modulation. It is interesting to note the remarkable similarity between this early instrument and the precision light-type devices used today.

During this period of time, PPMs were the predominant level indicators used by European sound engineers to visually monitor program material, and this tradition has continued to the present time. One type of PPM in common use was the moving light-beam meter based on the principle of the mirror galvanometer. This instrument employed a precision optical system whereby a narrow beam of light was projected onto a mirror attached to a moving coil arrangement. The light beam, in turn, was reflected off the mirror surface onto an in-line, ground-glass scale calibrated in decibels. In a modern version of the moving light-beam meter, the light intensity increases and the display color changes to red when the program level exceeds zero reference, thus giving ample warning of the system overload point. While this device is a highly accurate level indicator, it is not found in general use today because of its high cost and the precision optics involved.

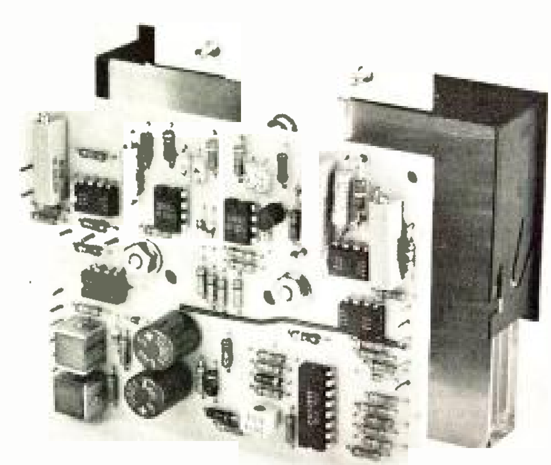
Since the early Seventies, advances in solid-state technology have resulted in the development of new light-type meters which are gradually finding their way into multi-channel equipment, both in the US and Europe.

THE STANDARD VU METER

Let us take a close look at the dynamic characteristics of the much-maligned VU meter. Basically, it is a rectifier-type a.c. voltmeter which responds to signal level components lying somewhere between the average and peak values of a complex waveform. There is no simple relationship between the power of such a waveform and the measured value, so that the actual reading will depend on the particular waveshape at the moment. In order to comply with ANSI Standards,[1] it should take the indicating pointer between 270 and 330 msec to reach 99 per cent of zero deflection in response to a suddenly-applied tone at that level, and overshoot the steady-state value by 1.0 to 1.5 per cent. Moreover, the fall-back time, required for the pointer to reach its rest position upon removal of the tone, should closely approximate the rise time.

Owing to meter-movement inertia, it is obvious that the standard VU meter cannot accurately register the short-duration peaks, varying both in frequency and amplitude, that

Figure 2. Driving amplifier circuit board may be mounted on rear of PPM case or wired separately. (Courtesy of Surrey Electronics)



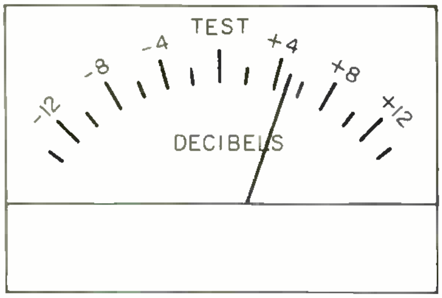


Figure 3. Scaling format for IEC Type IIb Peak Program Meter used by the European Broadcasting Union.

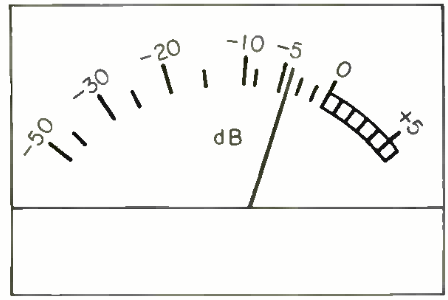


Figure 4. Scale layout for German DIN Standard Peak Program Meter.

occur in speech and music. In bringing the maximum peaks up to the zero mark on the meter scale, it is possible that momentary transients that barely indicate on the meter, but nevertheless require headroom, will exceed the modulation capabilities of the system. Close-miked pianos, muted brass, and snare drums are prime examples of these types of signals. When such high crest values are encountered, the recording engineer may find it necessary to "ride gain" in an area considerably lower than what the VU meter would indicate as a normal, or "safe," level. Of course, with some types of program material it is possible to "peg" the moving pointer and still not approach excessive levels. Only by experience can the operator make the subjective judgement as to how much under-modulation or over-modulation to allow on the VU meter, in order to avoid degrading the signal.

It is generally assumed that the VU meter registers typical program material at about 10 dB below the actual amplitude. Under controlled conditions, however, certain complex waveforms have been observed with crest factors up to 16 dB above the VU meter reading. Many of these transient peaks are too short to cause detectable distortion and, in general, the shorter the peak, the greater the amount of momentary over-modulation that can be tolerated. Experience has shown that significant peaks of program material, observed on a peak-responding meter, tend to be some 6 to 8 dB higher than the highest deflection indicated on a VU meter.

Another point of consideration is that most of the waveforms generated by the human voice (speech or singing) and synthesized musical sounds are asymmetrical in character; that is, the signals have a greater amplitude for one polarity than the other. Ideally, any errors introduced by asymmetrical waveshapes should be unmeasurable when the polarity of the signal is reversed. However, unless the meter employs a precision full-wave rectifier incorporating the correct averaging characteristic, there may be some variation in meter reading, if the polarity of the applied signal is reversed.

PEAK PROGRAM METERS

Essentially, the PPM responds to the quasi-peak values rather than the true, or absolute, values of an audio signal. Unlike the standard VU meter, which requires special electro-mechanical components to achieve the desired speed of movement, the PPM has a driving amplifier which allows the dynamic response to be adjusted electronically. The time constant of the meter movement is controlled by an RC network which holds the peak values of the signal long enough for the indicating pointer to rise rapidly to the correct scale reading, then fall back slowly for easy readability.

Because of the fast rise time (on the order of 10 msec) and the slow decay time (about 2 or 3 sec), the meter integrates over a number of peaks of the signal, so that the indicated reading actually follows the contours of the peak levels of the waveform. The governing factor in selecting a suitable speed of response, is that an otherwise excessively-fast meter, responding to the briefest transient spikes—say, in the microsecond range—would provide too much detailed information. The operator would then be confused by the wildly-fluctuating pointer, and tend to ride gain at too low an average modulation level. Only a true peak-reading instrument, such as an oscilloscope, would be capable of displaying all the instantaneous peaks of the signal waveform.

It should be noted that many PPMs are provided with an optional "slugged" mode of operation, which effectively slows the response of the meter movement. Under these conditions, program peaks tend to be more-clearly defined and can thus be read off separately. This makes it possible to compare signal levels at distant points in the system with a reasonable degree of accuracy. Interestingly enough, this "peak-checking" function was an important consideration when the ballistics of the standard VU meter was originally being formulated.

PPM STANDARDS

According to which peak-indicating standard is applied, there are some discrepancies in the ballistic behavior and scale factors of commercially available PPMs. FIGURES 1 and 2 illustrate an example of the British Standard PPM,[2] originally based on BBC specifications and used by all broadcasting authorities in the UK. It can be seen that the scaling consists of white calibration marks, numbered 1 through 7, on a black background with a white indicating pointer. This arrangement was adopted to help reduce eye strain in situations requiring long observation. The use of a semi-log law in the amplifier circuit enables the scale divisions (calibrated in decibels) to be reasonably linear. Each scale increment represents a level difference of 4 dB, so that the scale range is −20 dB to +4 dB with respect to mark 6, which indicates 100 per cent modulation.

Normal practice is to line up the meter on a sinusoidal signal, say 1 kHz, so that mark 4 is registered when the tone corresponds to 40 per cent modulation. This reference voltage (0.775 v) is called "zero level," and the intensity of any continuous tone is expressed in decibels above and below this value. Mark 6 (about 1.95 v) represents "zero volume," i.e., the intensity of program material that peaks frequently to a point 8 dB above zero reference level. The range from mark 6 to mark 7 indicates the extent of overloading.

The dynamic performance of the instrument is measured by introducing isolated 5-kHz tone bursts of varying lengths to the input and observing the results. If, for example, a 10-msec. tone burst is applied to the meter at a level at which a steady-state tone would give a reading of mark 6, the peak deflection should be −2.5 dB relative to mark 6. This width of tone burst gives an indication which nominally reads 80 per cent of the steady-state modulation level. By convention, it is referred to as the integration time of the instrument. If the same signal is repeated continuously at a repetition rate of 5 msec. there should be a steady-state deflection which is −4 dB relative to mark 6. The fall-back time, defined as the time taken for the pointer to decay from mark 7 to mark 1, should be between 2.5 and 3.0 seconds. Damping is determined by applying a direct current to give a pointer deflection to mark 7, and the resulting overswing should not exceed 5 per cent of the steady-state value. For slugged operation, the response of the meter to a 5-kHz tone burst of 100 msec. duration should yield a steady-state deflection which is -1 dB relative to mark 6.

FIGURE 3 shows the scale layout of the IEC Standard PPM[3] used by the European Broadcasting Union. Here, the principle scale markings are divided into 4 dB increments above and below the TEST level mark, which is at mid-scale position. Zero reference is taken as the level where the application of a 1 kHz sine wave of 0.775 v rms results in a deflection to the TEST point. The scale range is -21 dB to +3 dB, relative to the scale mark corresponding to +9 dB (2.19 v), which is the point where program peaks should kick frequently. If a 5-kHz tone burst is applied to the input, there should be a steady-state deflection of +6 dB, with an overshoot not more than 5 per cent. The release time taken for the pointer to return from +12 dB to 12 dB should be 2.8 sec. Expressed another way, the rate of fall-back should be 8.6 dB sec.

The German DIN Standard PPM,[4] widely used in Central Europe, calls for an attack time of 10 msec. to indicate 1.0 dB below zero reference level, in response to a 1-kHz tone burst. The decay time should be 1.5 sec. for the indicating pointer to fall from 0 dB to -20 dB. Overshoot should be less than 1.5 dB above the steady-state value. Referring to the scale format shown in FIGURE 4, it can be seen that the meter covers a wide dynamic range (55 dB).

LIGHT-TYPE METERS

So far, the discussion has focused on traditional program indicators using electro-mechanical meter movements. There are some drawbacks, however, in applying these instruments to multi-channel situations. For example, the console operator may be required to scan the moving pointers of, say, 24 meters, in order to judge the relative gain. Since the visual angle of the human eye is severely limited under these circumstances, it would be difficult to observe the meters at opposite ends of the console simultaneously. It has been demonstrated, however, that the eye has the ability to accurately perceive moving, flashing displays of color (mostly red) with a much wider viewing angle.

Accordingly, new light-type indicators are readily available which offer far better readability than conventional meters, and are directly compatible with ICs and other solid-state drive circuitry. These are rugged devices which do not depend on the inertial movement of a fluctuating pointer, and hence have no overswing. And since they do not have the parallax problem of pointer-type meters, they can be read at angles up to 150 degrees. Moreover, they can be viewed at greater distances without the need for excessive size. For the most part, these indicators consist of led displays ranging from simple peak overload detectors mounted within the VU meter case, to more complicated multilevel displays matching the effective ballistics of their electro-mechanical counterparts. As these displays become more refined, they are gradually being joined by other display technologies, including plasma and fluorescent types, and multicolor lcds.

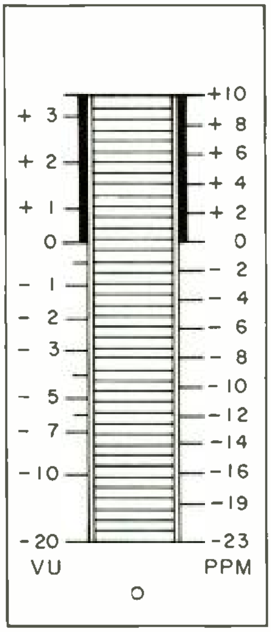


Figure 5. A high-resolution led program indicator with selectable VU or PPM ballistics.

FIGURE 5 shows a typical example of a high-resolution led bar-graph indicator yielding a highly-visible, flicker-free display. This instrument can function either as a peak-reading or average-responding device (using separate scale factors), whose ballistics are switchable depending on the mode of operation desired. Level indication is accomplished by a small vertical column of 36 linear illuminated segments whose display color changes from yellow to red when a peak overload condition is indicated. In the PPM mode, the first yellow segment comes on at -23 dB, and the transition from the top yellow to the first red segment occurs at 0 dB. In some units, a 3-color system is used to indicate modulation levels—green for under-modulation, yellow for normal modulation, and red, of course, signaling over-modulation.

An important feature incorporated in some models is a "peak-holding" capability which permits the operator to store the highest signal value reached, either for a short interval or until the indicator is reset manually. If this setting is exceeded, the red sectors will flash at twice the normal brilliance for a predetermined period of time. This is a useful function in disc cutting or tape duplicating operations, where the gain control can be adjusted to accommodate the maximum peak level encountered in a recording session. Also, display intensity can be remotely controlled to suit ambient lighting conditions, from the bright environment of a recording studio to the dim environment of a tv control room.

To further reduce "head-swiveling" in multi-channel recording, there are bar-graph indicators available which display up to 28 channels of audio data closely spaced on a tv monitor screen. Each illuminated bar represents one channel or meter bus. The vertical bars are always visible as a background reference, but as the level of a channel increases, the bar indicating that channel increases in height and intensity. The lower two-thirds portion of the monitor is a blue filter indicating the safe area, and the upper one-third a red filter indicating the danger, or overload area. ■

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3. IEC Standard 268-10a also specifies Type IIb PPM used by the European Broadcasting Union, Brussels, Belgium.

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BARRY HUFKER

Re-inventing The Microphone

The search for improved microphone design has often left inventors with interesting, creative, and, at times, unconventional results.

EVER SINCE THE INVENTION of the first microphone there have been many innovative and practical transducer designs which have helped to advance the quality of recorded and broadcast sound. The capacitor, moving coil and ribbon are such microphones whose construction and history have been well documented over the years.

Leaving the well-worn path of conventional microphone design, various inventors have tried to create "a better microphone" by at times, literally "re-inventing" it. Their results have often been highly creative, if unconventional. The histories of some of these transducers are scant because they never achieved popularity, or in some cases, practicality.

Although starting with reasonably worthwhile ideas, experimentation would soon reveal to the inventor that the creation possessed a "fatal flaw." The flaw would prove to be so basic in the design as to make it insurmountable, preventing the originator from using his device as a means of making any real or lasting contribution to audio. Whether inventing the first microphone or trying to gain notoriety for having built a successful new type, there comes a time when one realizes some ideas are better than others.

EARLY MICROPHONE DESIGN

Examining the construction of the human ear for clues to a practical transducer, J. P. Reis, working in the 1800's, imitated the eardrum in his microphone by employing a stretched diaphragm, to which were connected loosely-contacting metal parts. Having some pieces of the microphone puzzle, Reis was hampered by his choice of diaphragm material—sausage skin. The finished device could transmit single frequency tones but was plainly inadequate for the conversion of more complex sounds such as speech. Coming so close to success, it was Reis' inability to overcome the inadequacies of his device which prevented him from gaining recognition as the inventor of the first practical microphone.

In contrast, the capacitor microphone has been an important and enduring contribution to broadcasting and recording. It has evolved in a variety of forms capable of high sound pressure levels, low noise, excellent frequency response and multiple polar patterns. Thus, when seeking a means to duplicate the capacitor microphone's success, inventors thought it a reasonable jump to the inductor as the possible solution.

THE INDUCTOR MICROPHONE

While the modern moving-coil microphone does indeed operate on the principle of inductance, and shares many of the capacitor's qualities, there was at one time an "inductor microphone." The inductor was very much like the dynamic microphone of today, in that it too was a moving-coil device. Suspended in a strong magnetic field, the coil consisted of a single turn of wire attached to a parchment diaphragm. The inductor microphone was an omnidirectional, pressure instrument, with an impedance-matching transformer used to match the coil to the output line. This particular scheme had several of the necessary elements to be successful but performed poorly and was made obsolete by other moving-coil designs.

The invention of the vacuum-tube amplifier enabled traditional generating elements, such as the capacitor and magnet, to be used in microphones. In a creative but less conventional approach to inventing microphones, G. M. Rose of the Tube Department of the RCA Victor Division developed several types of "mechano-electronic" transducers.

Harry F. Olson, writing in the 1947 *Journal of the Acoustical Society of America*, thought it "rather interesting and

Barry Hufker is a recording engineer/producer at KWMU, University of Missouri—St. Louis.

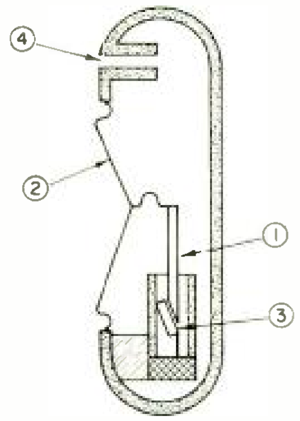

Figure 1. Sectional view of mechano-electronic transducer. (1) variable tube element, (2) paper diaphragm, (3) fixed element, (4) low frequency port.

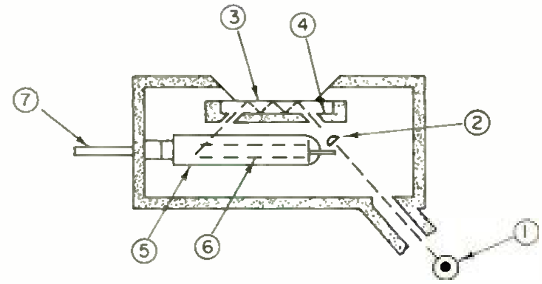

Figure 2. 1938 Banks optical microphone. (1) light source, (2) lens, (3) metal diaphragm, (4) fixed reflector, (5) electron multiplier tube, (6) cathode, (7) output.

somewhat surprising that the direct method of transduction, instead of the indirect method of employing a conventional transducer and then amplifying the output with a vacuum tube, has not been developed, particularly in view of the fact that the electronic transducer appears to possess many advantages over the conventional transducer."[1]

"ELECTRONIC" MICROPHONE

In one embodiment designated RCA type 5734, the mechano-electronic (or "electronic") microphone was constructed much like a vacuum tube (see FIGURE 1). One of the tube's internal elements was connected to a two-inch paper diaphragm comprising one side of the tube's glass container. Impinging sound waves fluctuated the diaphragm, varying the distance between the moveable element and the fixed one. This action resulted in a flow of electrons proportional to the sound wave's amplitude. The port in the casing enabled the microphone to accentuate the low frequency response while still achieving a sharp low cut-off frequency. The greatest performance demands fell upon the diaphragm which had to cope with a vacuum on one side, 15 pounds per square inch of static pressure on the opposite side and yet be able to react to sound pressure of less than a millionth of a pound.

Tests of the RCA electronic microphone revealed it to have low linear distortion, but a useable frequency response of only 100 to 6000 Hz. High hopes were held in 1947 for the mechano-electronic transducer as developed by RCA. It was even thought that in a slightly different form the principle could be a

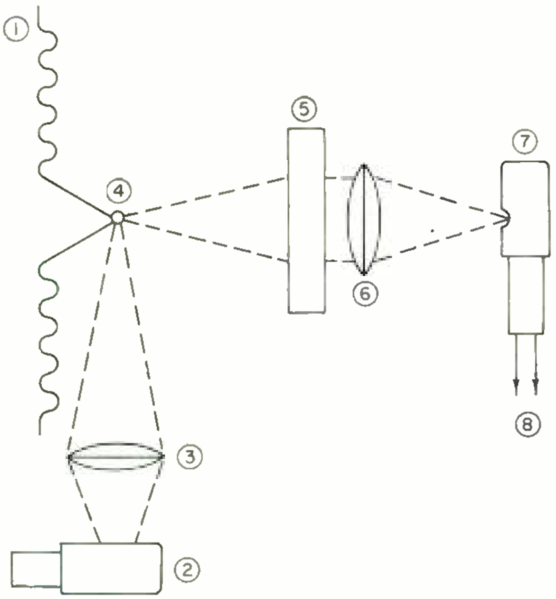

Figure 3. 1951 optical microphone by MacDonell. (1) diaphragm, (2) light source, (3) lens, (4) reflecting cup, (5) grating, (6) lens, (7) photo tube, (8) output.

succcessful phono pickup, yet Rose's invention was never commercialized.

THE OPTICAL MICROPHONE

1938 saw the advent of still another unusual transducer. Englishman George Banks invented the first optical microphone and assigned the rights to RCA. Banks' optical microphone was assembled in a metal housing with an opening at the side as seen in FIGURE 2. From a light source at this opening, a beam wave focused through a lens onto a reflective metal membrane. This diaphragm was exposed to the sound source. Parallel to the membrane was a fixed reflector which received the light bounced from the diaphragm. The light then zig-zagged at 45-degree angles between diaphragm and reflector, until it passed through a hole onto an electron multiplier tube, where it was amplified. When at rest, the diaphragm let approximately half the light reach the cathode of the amplifying tube. The diaphragm's movement varied the amount of light shining on the tube and an output signal was generated. The electrode opposite the cathode was built of light-permeable mesh and served as the tube's output.

John MacDonell re-invented the light microphone in 1951 using a "circular diaphragm with concentric corrugations."[2] (See FIGURE 3) This time a light source provided a beam focused through a lens onto a cyllindrically-shaped cup, fixed to the center of the diaphragm. Light from the cup was beamed onto a Ronchi ruling, or grating, comprised of alternating opaque and transparent evenly-spaced parallel lines of equal width. From this grating, the light travelled to another lens, where it was focused upon a phototube. Movement of the diaphragm modulated the light reflected from the cup to the grating. As the light passed through the grating, it was focused on the phototube, where the modulated light would be amplified as an output signal.

Patents 2,259,511 and 2,666,650 were awarded to Banks and MacDonell for their microphones, which for all their apparent promise were never commercialized, simply because they didn't work. A full discussion of the optical microphone's shortcomings would be an article in itself and indeed, one was written by Virginia Rettinger in 1965. Briefly and simply it can be said that Banks' transducer assumes soundwaves will displace the diaphragm sufficiently to cause the light beam to be modulated. In reality, the displacement is on the order of micro-

Figure 4. The Shure SM-33. A modern ribbon microphone, designed for studio use.

inches and is not enough to make any measureable difference in the angle or intensity of the beam and hence no modulation occurs.

MacDonell's microphone fails for a slightly different reason. Simply, the diaphragm's displacement due to the sound wave is too small to force the beam to transverse the lines of the modulation grating and subsequently no output is produced.

THE RIBBON MICROPHONE

The velocity transducer, or ribbon microphone, has been an important tool in audio for many years and has endeared itself to recording engineers and broadcasters alike. It has been characterized by high sensitivity, smooth frequency response, a subjective "warmth" and in early models, an extremely fragile ribbon easily stretched beyond use by relatively strong puffs of air. Modern versions are hardier, preserve the famous qualities of past types and provide all the best the ribbon has to offer. Unfortunately, somewhere along the line, ribbons fell into an undeserved decline in popularity,having been dropped in favor of the capacitor or moving-coil. Engineers who have not picked up a ribbon in a while will find they can hold up like other dynamic microphones, and are invited to re-experience them so that a viable microphone type will not go the same way as so many others of lesser value. ■

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FOR STUDIO or high end sound reinforcement system—JBL 4350A top-of-the-line studio monitor speakers. New. Mfg. price $3,972. Asking $2,650 plus shipping. Also, top end AKAI 570 DII cassette deck. Originally $1,200—will sacrifice for $700. Call **(201) 569-7867** after 7:00 P.M. EST.

EVENTIDE CLOCKWORKS HARMONIZER—Model H910, 1 year old, mint condition, $1,100.00. **Cathedral Sound Studios, Rensselaer, New York (518) 465-5689,** Ask for Carol.

STOCK CLEARANCE

REVOX, OTARI, TECHNICS & other quality recorders, mixers, headphones, mics, pre and power amps, speakers, etc. Lists from—**Entertainment Sound Services, Inc., P.O. Box 66, Madison, Ala. 35758. (205) 772-0251.**

UREI: FOR IMMEDIATE DELIVERY most items **UAR Professional Systems, 8535 Fairhaven, San Antonio, TX 78229. 512-690-8888.**

SERVICES

CUTTERHEAD REPAIR SERVICE for all models Westrex, HAECO, Grampian. Modifications done on Westrex. Avoid costly down time; 3-day turnaround upon receipt. Send for free brochure: **International Cutterhead Repair, 194 Kings Ct., Teaneck, N.J. 07666. (201) 837-1289.**

MAGNETIC HEAD relapping—24 hour service. Replacement heads for professional recorders. **IEM, 350 N. Eric Drive, Palatine, IL 60067. (312) 358-4622.**

MULTI-TRACK RECORDING specialists—1-2-4-8-16-24 tracks, authorized dealer for Tascam, Otari, Ampex, Teac, Technics, AKG, AB Systems, Crest, SAE Pro, dbx, Orban, Tapco C12, Audioarts, Loft, Lexicon, Ashly Audio, Altec, PAS, PSL, Shure, and many more. Single items or complete studio packages. Studio design and construction. Phone or write for a prompt written quotation. **Professional Sound Labs, Inc., 42 North Franklin St., Hempstead, NY 11550. (516) 486-5813.**

JBL AND GAUSS SPEAKER WARRANTY CENTER. Full lines stocked. Instant recone service, compression driver diaphrams for immediate shipment. **Newcome Sound, 4684 Indianola Ave., Columbus, Ohio 43214 (614) 268-5605.**

AMPEX SERVICE COMPANY: Complete factory service and parts for Ampex equipment; professional audio; one-inch helical scan video systems; video closed circuit cameras; instrumentation consumer audio; professional audio motor and head assembly rebuilding. **2201 Lunt Ave., Elk Grove Village, IL 60007; 500 Rodier Dr., Glendale, CA 91201; 75 Commerce Way, Hackensack, NJ 07601.**

ACOUSTIC CONSULTATION—Specalizing in studios, control rooms, discos. Qualified personnel, reasonable rates. Acoustilog, Bruel & Kjaer, HP, Tektronix, Ivie; equipment calibrated on premises. Reverberation timer and RTA rentals. **Acoustilog, 19 Mercer Street, New York, NY 10013 (212) 925-1365.**

WANTED

WANTED: USED RECORDING equipment of any kind. Expanding studio will pay cash. **Dennis Reed, Box 50022, Palo Alto, CA 94303 (415) 494-9344.**

MICROPHONES WANTED for broadcast historical archive: Early RCA and Western Electric models. Contact: **James Steele, National Association of Broadcasters, 477 Madison Avenue, New York 10022. (212) 759-7020.**

WANTED: USED RECORDING equipment of any kind. Expanding studio will pay cash. **Dennis Reed, Box 50022, Palo Alto, CA 94303 (415) 494-9344.**

WANTED: PR10 2-track electronics. Must be in good working condition. **(408) 224-1777.**

BUSINESS OPPORTUNITIES

MIDTOWN COMMERCIAL PRODUCTION/DEMO studio forced to move. Will consider partnership with existing studio contemplating expansion, or freelance engineer looking for permanent location. **(212) 575-0295.**

COMPLETE AUDIO STUDIO package available in Tulsa, including MOOG 55 and 3 module Arp synthesizers, 8-track and 2-track Ampex recorders with dbx units, mixing board and much more. Call **(602) 488-3454,** or **P.O. Box 1316, Carefree, Arizona 85377.**

EMPLOYMENT

TM PRODUCTIONS/PROGRAMMING seeks additional maintenance engineer. Degree, five years experience, and familiarity with MCI multi-track gear preferred. Send resume to **Ken Justiss, Vice President, TM Productions; 1349 Regal Row; Dallas, TX 75247.** No phone calls.

SOUND ENGINEER AVAILABLE—experienced and proficient in all aspects of professional theatre sound, especially systems concepts and design. Past experience: engineering services for US Gov't., two universities, theme park and Philadelphia Bi-Centennial committee. Currently president of small company offering systems design, consultation and limited sales. Applicant seeking a challenging, creative, responsible position with manufacturer, distributor, retailer or contractor in the audio equipment market seeking expertise in this area. Possibility for advancement highly desirable. **Keith Arnett, PO Box 139, Kingston, NJ 08528;** Member AES.

WANTED: CHIEF ENGINEER for FM/AM combination in S.E. Wisconsin. FM is 50K, Top 40 with new studios...AM is 500W daytime, partially automated with Harris System 90. Knowledge of both is essential. Need take-charge person with ability for more than just maintenance. Send resume to: **Ron Richards, Op. Mgr., WRKR/WWEG Radio, 2200 N. Green Bay Rd., Racine, WI 53405.** An equal opportunity employer.

db People/Places/Happenings

• Resulting from an increased need for field training and product application in the Language Lab Systems Market, **Telex Communications, Inc.**, Minneapolis, MN, has appointed **Robert E. Krolow** to the newly created position of system sales manager for the company's audio visual products group. In addition, **Steven R. Henriksen** has been named director of customer service and technical training; responsible for conducting service training seminars, both domestically and internationally, for the company's aviation, audio-visual and broadcast professional audio product groups. Mr. Henriksen will also provide technical support and emphasize both preventive and corrective service training for technicians at the institutional and user level.

• Recent additions to the staff of **Philips High Fidelity Laboratories, Ltd., Fort Wayne, Ind.**, include: **Dick Quaid** assigned to the position of national sales manager; **Johan Koppier** named product manager; and **Ed Williams** appointed advertising/public relations coordinator. Mr. Quaid was formerly the national marketing manager for the market introduction of Magnavision, from **Magnavox**. Mr. Koppier was previously in audio product management at the **Magnavox Consumer Electronics Company**, and Mr. Williams was most recently a member of the advertising staff of Magnavox.

• **Recordex Corporation**, formerly located in Northwest Atlanta, has moved its manufacturing and national headquarters to new and larger facilities in Marietta, Georgia. The new address is: 1935 Delk Industrial Blvd., Marietta, Georgia 30067. Telephone: (404) 955-7368

• Marking the culmination of a major five-year engineering effort, **National Public Radio** has begun program transmission via Western Union's **Westar I** satellite. Earth terminals, provided by **Rockwell/Collins**, will be operating at NPR member stations in Oregon, Washington and Utah, as well as the main origination terminal in Bren Mar, Virginia which is shared with the Public Broadcasting Service.

• **Queen Village Recording Studios**, Philadelphia, PA, has named **Joe Campellone** general manager of the studio. Before joining the Queen Village staff, Mr. Campellone handled sales, marketing and merchandising for **Dominion Music**, and was in charge of their East Coast operation. Elsewhere at Queen Village, **Wally Hayman** has been promoted to studio manager, overseeing all studio operations and acting as liaison for ad agency business; and new to the staff is engineer **Bill Olszewski**. Also, the studio has recently installed a 40-track Neve recording console, in addition to their 24-track facility. Busy times at Queen Village.

• Recently relocating from Beaverton to Portland, Oregon, **Biamp Systems, Inc.** has more than tripled the size of its production facilities. Biamp's new address is: 9600 S.W. Barnes Road, Portland, Oregon 97225. Telephone (503) 297-1555.

• **Jay A. Clark** has been appointed director of public relations for **Ampex Corporation**, Redwood City, CA. Prior to joining Ampex, Mr. Clark served as manager of media relations at **Rockwell International Corporation**, Pittsburgh, Pennsylvania.

• Providing professional exhibit design, audio visual, copywriting and promotional services to commercial, political, industrial, financial and international organizations, a new multi-media company, **Show-Off International, Ltd.**, has been formed. Headed by **Ronald K. Chedister**, veteran audio visual designer, the firm is located at: 6870 Elm Street, McLean, Virginia 22101.

• Two recent additions to the staff at **Martin Audio/Video Corporation**, New York City, were: **Gordon L. Clark** as technical manager; and **William H. Dexter** to the pro audio sales department. Promoted to the position of service manager, at Martin, was **Tim Holmes**.

• **Barry Evans** has been appointed national sales manager of the Revox division of **Studer Revox America**. Mr. Evans brings with him, varied experiences from **Fuji** tape, **Akai** and **Uher** recorder sales.

• A number of top level management positions have been filled at **Cetec Gauss**, North Hollywood, CA. **Larry Phillips** has been named marketing director for loudspeaker products; **Jim Williams**, formerly quality assurance manager for Cetec Gauss products, has been appointed director of engineering for the company; **Jerry Fisher**, previously affiliated with **CBS** as manager, process engineering, tape duplication operations, has replaced Mr. Williams as quality assurance manager; **Walter Dick**, formerly with **JBL** as manager, transducer engineering, is the new chief engineer for Cetec Gauss loudspeaker products; and **Bart Bingaman** has been named chief engineer, duplicator products.

• **N. Sakoda** has been appointed to the position of director and president of **US JVC Corporation**, a wholly owned subsidiary of **Victor Company of Japan, Ltd.** Mr. Sakoda succeeds **Mr. S. Hori**, who will be assuming new responsibilities at Victor world headquarters in Tokyo. Previously, Mr. Sakoda was general manager of the export administration division at Victor headquarters.

• Responsible for sales training and conducting consumer seminars, **Peter Hoagland** has been appointed manager of customer service at **Signet**, a division of **A-T U.S.**

• In an expansion move, **Ferrofluidics Corporation** has relocated its corporate facilities. The company's new address is: 40 Simon Street, Nashua, NH 03061. Telephone: (603) 883-9800.

• **Bruce Scrogin** has been promoted to the position of vice president for international sales at **James B. Lansing Sound, Inc.**, Northridge, CA. A member of the JBL marketing team since 1972, Mr. Scrogin recently served as director of the international division. In addition, **Nina Stern** has been promoted to the position of public relations manager at JBL.

• Relocating into larger premises, **Tweed Audio (USA), Inc.**, has moved their USA office to: 4545 Industrial Street, Suite 5-E, Simi Valley, CA 93063. Telephone: (805) 527-6854.

If Technics RS-1500 meets the high standards of A&M Records, why did we improve it?

After the music is recorded, and before it becomes a record, how do the top executives of A&M Records listen to Peter Frampton, Chuck Mangione, and their other stars? On the Technics RS-1500. Why? Because of its outstanding frequency response, constant tape speed and low wow and flutter. In fact they were so impressed, A&M Records bought seven more.

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Electronically, the RS-1520 is equally sophisticated. And the reasons are as simple as IC full-logic controls. A highly accurate microphone amplifier. FET mixing amplifier. And separate 3-position bias/EQ selectors.

The RS-1520. It meets the high standards of A&M Records for the same reasons the RS-1500 does: FREQ. RESP.: 30-30,000Hz, ±3dB (−10dB rec. level) at 15ips. WOW & FLUTTER: 0.018% WRMS at 15ips. S/N RATIO: 60dB (NAB weighted) at 15ips. SEPARATION: 50dB. START-UP TIME: 0.7 secs. SPEED DEVIATION: ±0.1% with 1.0 or 1.5 mil tape at 15ips. SPEED FLUCTUATION: 0.05% with 1.0 or 1.5 mil tape at 15ips. PITCH CONTROL: ±6%. TRACK SYSTEM: 2-track, 2-channel recording, playback and erase. 4-track, 2-channel playback.

RS-1520. A rare combination of audio technology. A rare standard of audio excellence.

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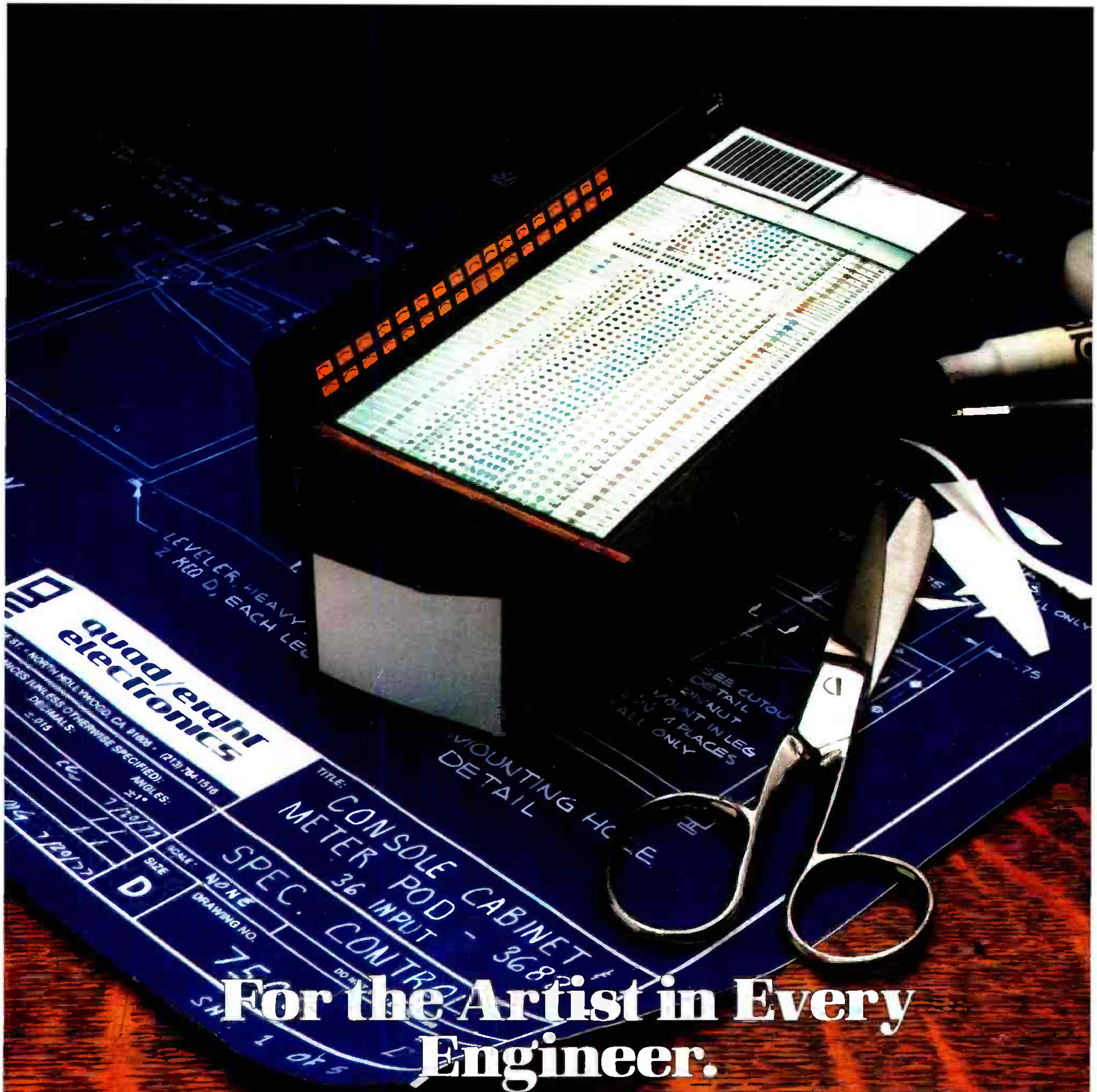

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