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THE SOUND ENGINEERING MAGAZINE
August 1970 75c

Optical Sound Track Processing
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This is what happens when a big name thinks small. It's the TEAC A-24, and it's making cassette history. This deck is powered by a unique hysteresis synchronous outer rotor motor for compact convenience, powerhouse performance. And it comes complete with all the craftsmanship it takes to make a TEAC.

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Coming Next Month

●September is the preview issue for the fall AES Convention to be held in New York. We will have the complete program and maps of the exhibits.

Edward J. Gately, Jr. has prepared an article entitled IC's: The Coming Revolution The author has prepared a strong case for these sub-circuits which he illustrates fully in actual product evaluation examples.

And in October, we will feature a special issue devoted to compressors and limiters. Included will be a directory of products and a round-table discussion by several manufacturers on these components.

And there will be our regular monthly columnists: George Alexandrovich, Norman H. Crowhurst, Martin Dickstein, Arnold Schwartz, and John Woram. Coming in **db**, The Sound Engineering Magazine.

About the Cover

●A fish-eye lens peers at the announce studio of AFRTS-Washington. A story on the construction of these studios appears on page 28.

db

THE SOUND ENGINEERING MAGAZINE

AUGUST 1970 • Volume 4, Number 8

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Letters

The Editor:
Mr. F. C. Hervey, in his May letter, has suggested three-channel stereo as being more simple than four and still capable of a surround of sound. Unfortunately, the use of three channels, as proposed, does not simplify such a basic medium of reproduction as the phonograph disc. However, the idea of matrixing the third channel *can* give a simple system with full disc and radio compatibility.

If the output of a rear microphone is mixed into the two stereo channels *in opposite phase*, it can be successfully played through a rear speaker *differentially* connected. This is the precise mirror image inversion of the so-called phantom channel, in which a front center speaker uses the sum information of the two channels. The psychology of listening acts in our favor in these two connections. The in-phase information forms an apparent source in front, while the out-phased information will not form a center. The rear speaker gives the out-phased information a 6 dB advantage as compared to its amplitude in the side speakers. This gives a strong rear source for rear signals which can be either ambience signals or other instruments or voices.

This system can be played as normal stereo or as the enhanced stereo with the extra speaker. The extra speaker can be two speakers in series or parallel in the two rear corners, if desired. The system can be used on discs or any other medium without limitation of bandwidth, and it gives the sonic benefits of three independent channels. If discs and tape were prepared this way, the user would have his choice of two-or three-channel reproduction without having to add anything but an extra loudspeaker with a potentiometer to control its level.

A further extension of this method is to use a front-centered microphone, the output of which is mixed in phase in the two channels. A front speaker is fed with the sum of the two channels. The two side speakers can be separated more than usual so that the center speaker is effectively an additional sound source.

In this extension, it can be seen that there is no back information in the front speaker and no front information in the back speaker. The time lag from front to back of the recording hall is precisely reproduced in playback from front to back speaker. Thus all the hall ambience is reproduced. It is also practical to have four instruments playing simultaneously with four distinct sound sources, so that this simple system can give all the benefits of four separate channels.

It would seem that efforts to promote four-channel sound with four-track tape are unnecessary when the sonic benefits can be derived from two channels properly recorded.

David Hafler
President, Dynaco Inc.
Philadelphia, Pa. 19121

ERRATA

At the bottom of the first column of last month's article Synchronous Recording Techniques, reference was made to 7/8-inch tape. This should, of course, been ¼-inch tape. We are not attempting to set new standards in tape width.

Page 34 of that issue illustrates an eight-channel mixing module and indicates it to be made by Gately Electronics. This is a mislabeling; the module illustrated is manufactured by Suburban Sound Inc.

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This article will help to reduce all that microphone noise you've been hearing.

The din keeps getting louder: claim and counter-claim.

With three different powering systems available, no wonder the users of transistor-equipped condenser microphones are confused by all the "power politics."

This article will help. Originally published by *db* Magazine, it gives the pros and cons of each system. Fully explained. Carefully diagrammed and documented.

It's *not* promotional literature. But we'll be glad to send you a reprint free. (And extra copies, if you need them for distribution.) Because, frankly, we've never come across a clearer explanation of why Neumann elected to use a 48V PHANTOM® powering system.

Last on the scene with a transistorized condenser microphone, Neumann had *every* powering option open to it. This article will show you why the world's most prestigious manufacturer of electro-acoustical transducers made the electronics decision it did. And why thousands of users are so pleased with the result.

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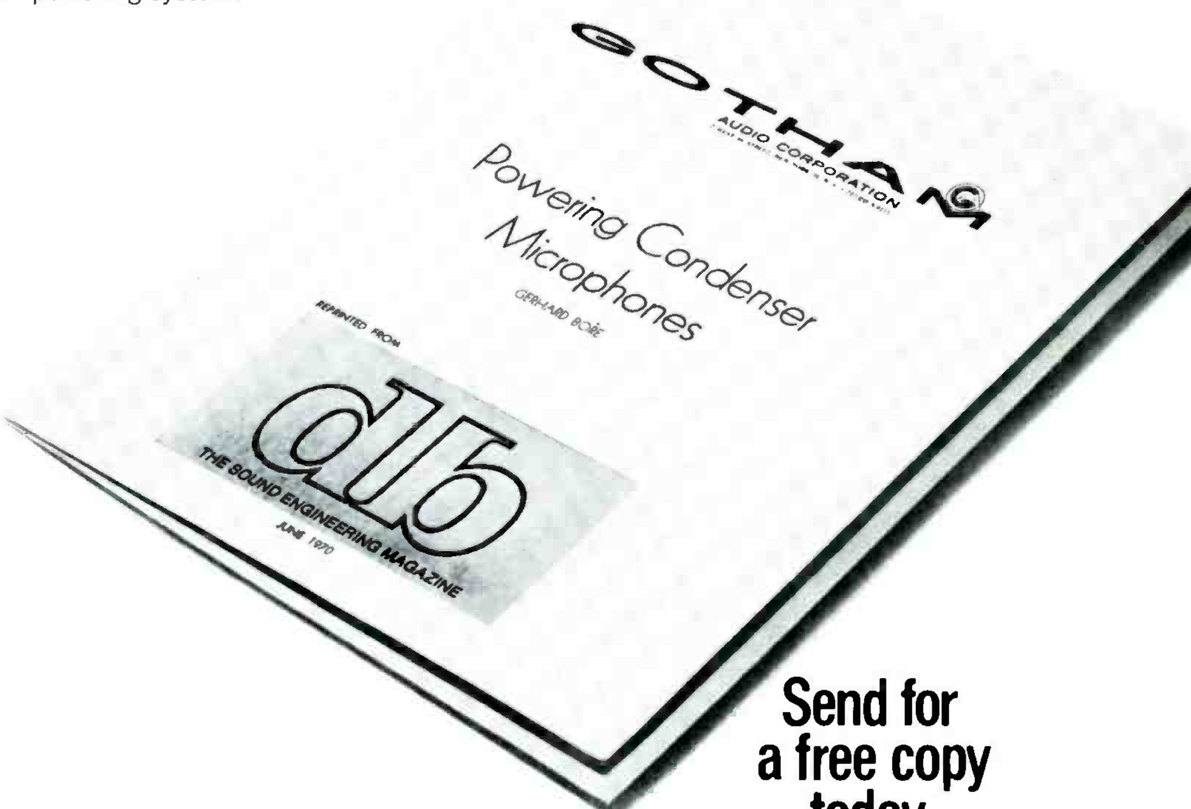


Send for a free copy today.

The Audio Engineer's Handbook

GEORGE ALEXANDROVICH

MEASURING S/N RATIO

● The problem of correctly measuring and interpreting s n of a system or a device has been with us from the days of Edison. If you survey the performance specifications of audio equipment today, you will be amazed at the variety of ways noise specs are written. And most of the times they are expressed in such a way so as to appear better than others.

This month I would like to summarize all the basic facts about noise specs and measurements and outline simple basic rules for their correct interpretation.

Let us start with the definition of *noise*. Noise in a system is any unwanted electrical signal or disturbance appearing and measured at the output of a device or a system after the signal source feeding the system has been removed. Noise can be random or periodic. Periodic type of noise for example, can be caused by picking up r-f signals by unshielded wires or components, or by picking up 60 Hz hum or any other source of periodic signal. Random noise can be acquired by the system through pickup or generated within a system itself. In our measurements of s n we include all types of noise signals mentioned above, regardless of their origin. Noise frequencies beyond audio spectrum can be disregarded, providing they don't cause other adverse effects, (such as distortion of audio signals, by generating beat frequencies or slope detection of r-f signals causing audible interference). The amount of interference at supersonic frequencies can be determined if, let us say, a 20 kHz filter is inserted into the output circuit and a decrease in noise level is read. But, let us take an ideal condition when there are no outside interferences and the only noise that we are concerned with is the one generated by the system or a device.

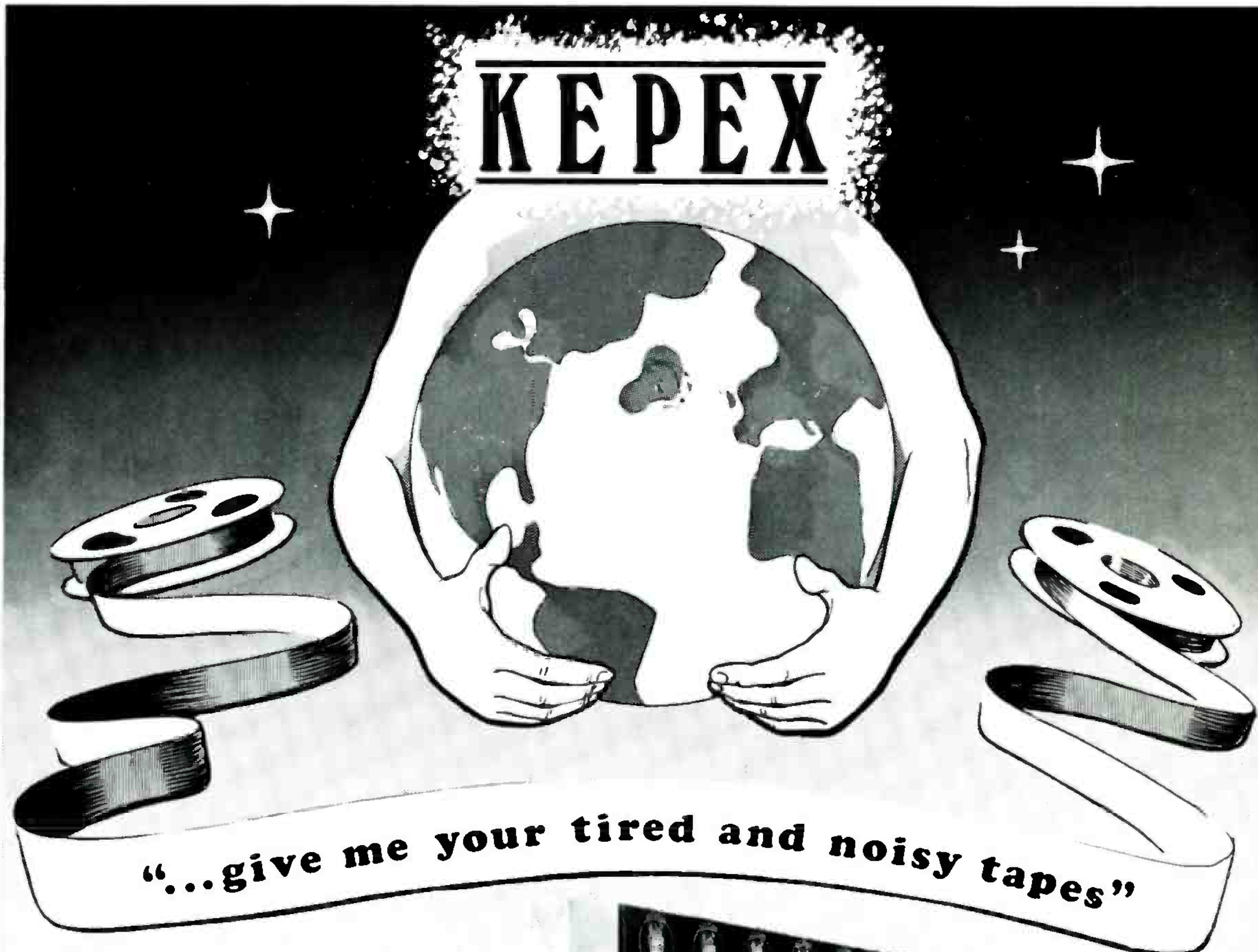
We have heard quite often the term *equivalent input noise*. What does it mean? It is a noise generated by the input stage of our device and normally is in order of −110 to −127 dB. What do *these* figures mean? They mean that noise generated by the components in the input stage has an amplitude of 0.5 μV (−125 dBm) across the source impedance of 600 ohms. If one wanted to feed 0 dBm level into this stage, at this point he would measure a s n of 125 dB, or if one would connect a mic to this input producing −60 dBm he could measure a s/n of 65 dB. (−60 dBm level is 65 dB above −125 dBm noise level).

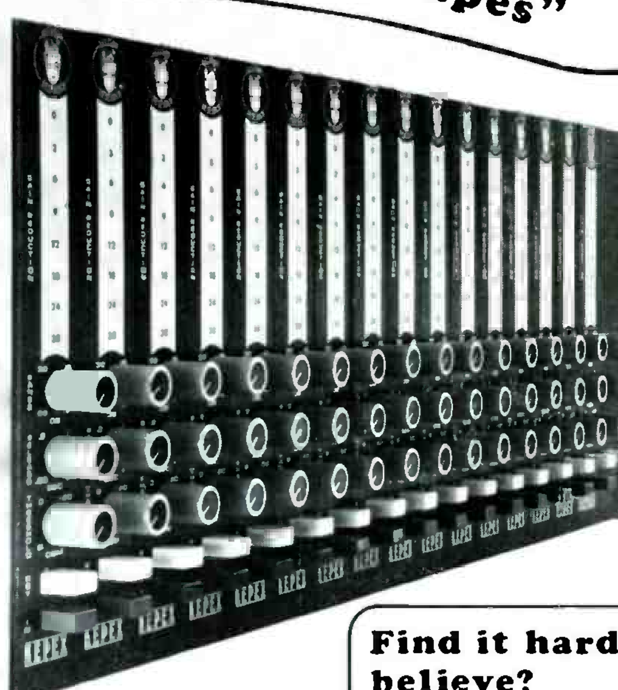
Now, let me point out that minimum *theoretical noise* in the case of the audio spectrum is about −130.9 dB. In other words, free electrons in conductors, resistors and other components produce about 0.27 μV of steady noise. The only way to change this condition is to use cryogenic techniques (supercooling conductors and other parts) something that is beyond our scope of discussion and out of the realm of practicality.

If −130.9 dBm noise level is the lowest we can obtain (wide-band measurement), then the closer the audio signal in our system drops to this −130.9 dBm level, the smaller the s/n we should expect, afterwards.

Let us consider first the high level input of the 10 input console. We feed 0 dbm 1 kHz signal into the input transformer. Signal goes through the fader dropping to perhaps −20 dBm level. Then we feed it into the mixing network which has a loss of, let's say 20 dB. Level at the output of the mixing network is measured at −40 dB. (See FIGURE 1.)

If there is no booster amp after the mixer network, the signal goes into the master fader where it loses another 20 dB. The signal feeding the output amplifier is now −60 dBm. We need 64 dB of gain in the line amp to restore the lost gain. Before the signal enters the booster it is only 70.9 dB above the theoretical minimum noise. Most of the amplifiers have noise generated by their input stages (the most vital point in noise generation) which measures from −110 dBm to −127 dBm. (As little as 3 dB above the limit). This small increase of about 3 dB lowers the possible s n of 70.9 dB to 67.9 dB. As the line amp boosts the level to +4 dBm (64 dB of gain), it also amplifies the noise generated by its first stage by the same amount (−127 dB +64 dB = −63 dB below 0 dB level). But we





WHAT'S A KEPEX?

In today's multitrack studio we are constantly at war with the build-up of undesirable noise. *Not just tape hiss*, but a host of equally offensive noise sources such as: *guitar amplifiers, air conditioners, room echo and leakage effects*, etc.

Consequently, the mixdown session often becomes more of a noise abatement session! *That's where KEPEX can really help.*

With a little programming help from you, KEPEX can quite accurately distinguish signal from noise. It can then go on to preserve the signal intact, while *reducing unwanted noise* signals by as much as *60DB!*

Since you are doing the thinking, KEPEX can be made to perform in a whole lot of ways, ranging from mild noise reduction to some really far out *Electronic Music Effects.*

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Find it hard to believe?

For those of you in doubt I've prepared a demonstration tape so you can hear the results of KEPEX in actual use! Write me for your free copy!

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have an output signal of +4 dBm so noise below our signal output would be 63 dB +4 dB = 67 dB.

Now, if we didn't allow our signal to drop to −60 after the master fader but instead would have a 20 dB amplifier, let us say, after the mixing network, then our levels would look like this: Into an amplifier with 20 dB of gain and with input noise of −127 dB, we would feed signal of −40 dBm, which is 87 dB above the noise level (maximum possible s/n is 20 dB higher now!). (See FIGURE 2.)

Boosting the signal 20 dB boosts the noise to −107 dB. Later, the fader attenuates 20 dB bringing the signal to −40 dB level and noise to −127 dB again. Now, we need only 40 dB of gain to bring the signal to line output level. An output amplifier with 40 dB of gain can produce s/n of 87 dB. This shows that keeping the signal as high as possible will produce a better s/n.

Now, let's review the methods of measuring noise correctly. First of all, a signal generator and vtvm should be on hand (most of them today cannot be called vacuum-tube voltmeters but tvm or fetvm) and preferably, also a 'scope. Vtvm and scope should be connected across the output of the system or device. Even before turning the power on, make sure the noise you read on the vtvm is at least 10 dB lower than the lowest expected reading with the power on. If you read something like −50 dB with the power off, you will never know what your system's noise really is. In this case, recheck your grounding and shielding of the output stage.

The next step is to turn the power on and feed the appropriate signal into the input.

Adjust all faders and or gain controls for normal operating gains and read appropriate output signal at the output of the system or device. Observe the wave form for a clean wave shape. Operating level should be at least 10-14 dB below the clipping level. You can check that by turning your oscillator output 10-14 db higher (and noting the level at which clipping or system overload occurs).

Now that you have all gains adjusted and signal at the output measured, remove the test oscillator from the input and replace it with appropriate dummy load resistor (well shielded) equal to the output impedance of the source (shorting input may produce lower noise reading but will not represent actual operating conditions.) Read the noise at the output. Let us say you obtain noise of 60 dB below your signal output. To find out what this figure of 60 dB represents, you may put a 15 or 20 kHz low-pass filter in the output and determine if excessive noise is above the audible range. You may find that with filter switched in, the noise reading may drop another few dB producing a s/n of, let us say, −65 dB.

To determine what stage is limiting further improvement in s/n, turn your master gain control down. If the noise level drops, then the noise is generated before the master control. If not, then look for cause in the stage following the master.

Usually, the biggest offenders in generating noise are mic preamps. Input levels are sometimes low (−60 dBm or lower) and preamp gains are high (40-60 dB). The best mic preamps have equivalent input noise on the order of −127 dBm, meaning that with mic signals of −60 dBm the best possible s/n one can hope for would be 67 dB. The input stage of a mic preamp contributes to 1/f and white noise, mic transformers and wiring to the preamp input in the average system additionally contributes to the input noise. Balanced transformer inputs cannot be ideally balanced nor can you entirely prevent pickup of hum and rf. Modern techniques using supply voltages for condenser microphones are quite ingenious, but using dynamic mics in the inputs wired for condenser mics may cause some deterioration of s/n, especially if input balance is not near perfect and if the power supply is not pure d.c. These are extremes, but I feel one should be made aware of possible causes of poor s/n. ■

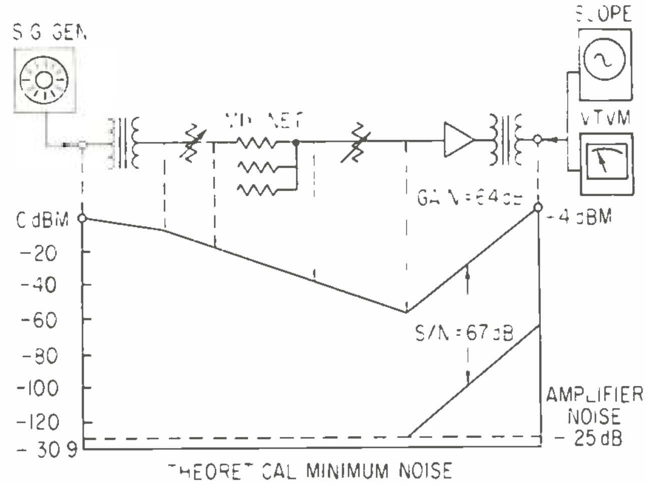

Figure 1. A diagram of level distribution.

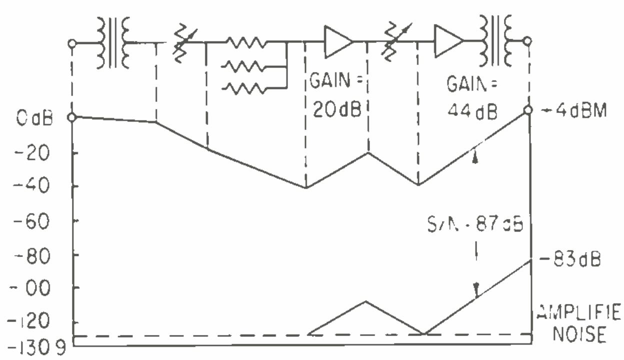

Figure 2. The effects of a booster amplifier on signal to noise.

Sony portable studio $995.00 (VW optional).

What happens when the talent can't come to the recording studio?

The recording studio goes to the talent. Simple move.

If you own the new Sony Model 772.

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The Sony 772 with its 7½ and 15 ips makes it uniquely suitable for master recordings—in the field.

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The Sony 772 is available at select Superscope Special Application Products dealers.

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SONY SUPERSCOPE

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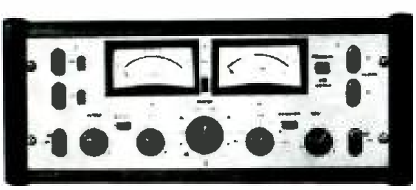
NEW IM ANALYZER

ADVANCES STATE OF THE ART

Extended measuring range and high-speed readings are the outstanding features of a unique new Intermodulation Distortion Analyzer introduced by Crown International recently. The American firm is known for its line of Crown precision professional tape recorders.

This analyzer was developed to meet the production line requirements of the Crown DC300 lab standard amplifier. The first need was for accurate measuring capability through 0.01%. This analyzer guarantees a residual IM level of less than 0.005%, with seven full-scale ranges from 100% to 0.1%.



The second requirement was for an instrument simple enough to be operated by production-line personnel and rapid enough to make sequential readings across the entire power band. The Crown analyzer meets the challenge by reducing measuring time from minutes per reading to just seconds. This is accomplished by a "tracking" function, using two meters and a ganged input/output gain control. The input level is set using the calibrate meter, and distortion is immediately read on the percentage distortion meter. Successive readings at 5db increments take under five seconds each. The entire operation is completed in less than one minute.

Solid state construction, utilizing FETs, makes the Crown analyzer highly stable and uniquely compact, measuring 7x19x7 inches. Rack mount list price is $595. Write for spec sheet to CROWN, Dept. DB-8, Box 1000, Elkhart, Indiana, 46514.

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The Feedback Loop

ARNOLD SCHWARTZ

● Two months ago I asked the question, "What is the upper frequency limit of the disc recording/playback system?" In trying to answer this question, an important distinction was made between ***frequency response***, and ***wavelength response***. The former is defined by the combined frequency response characteristics all the devices in the record/playback chain. The latter is a characteristic of the information (we call music "information") process. This concept of wavelength response is somewhat unfamiliar in disc recording. Last month I covered the wavelength response of the recording process. This month I will discuss the wavelength response of the complementary process, playback.

PLAYBACK WAVELENGTH RESPONSE

The phono cartridge wavelength response is also called *playback loss*. The output of the cartridge at high frequency tends to fall off at inner diameters, that is, where the wavelengths are small. However, since the cartridge does have a frequency response, the wavelength response, or playback loss will be superimposed on it. The basic cause for playback loss is that the disc is indented by the playback stylus. To understand the mechanism of playback loss refer to FIGURE 1 which shows a playback stylus in a modulated groove; the groove being viewed 45 degrees to the record surface. The following critical features are identified: wavelength, tip radius, tracking force, and the stiffness of the record material. In FIGURE 2 the relative indentation of the playback stylus is depicted for three possible cases. At the left side of the drawing the tip is located on a section of unmodulated groove; the dotted segment indicates the amount of indentation. In the center, the tip is located on a concave section of modulated groove; the groove contour tends to cradle the stylus, and the amount of indentation is less than for the unmodulated segment of the groove. On the right side of the drawing the tip is shown on a convex section of modulated groove; the indentation will be greater than both the unmodulated and concave modulation cases. Stylus indentation is shown (see FIGURE 3) for a longer wavelength, and the indentations are virtually the same for all three cases. The difference in indentation between concave and convex segments of the modulation depends upon groove curvature as compared to the stylus radius. As the wavelength decreases the groove increases—that is, convex and concave sections become more pronounced. As this curva-

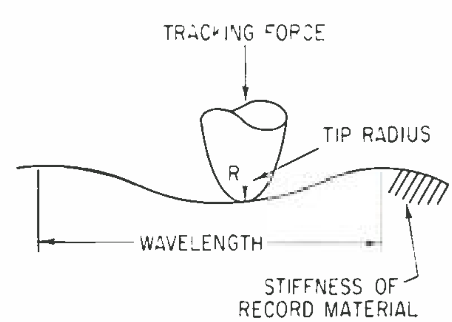


Figure 1. The playback stylus in a modulated groove.

How to record on a higher level!

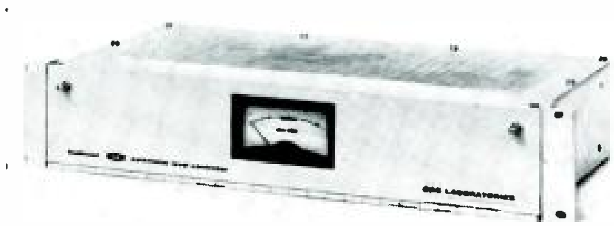
With automatic peak control

The CBS Laboratories' Recording Volumax Model 420 automatic peak controller for recording:

- permits higher recording levels without overloading.
- eliminates the distortion of clippers and the thumping and pumping of conventional limiters.
- provides the maximum peak output at all frequencies even as a function of record diameter.

The completely solid-state Model 420 Recording Volumax is unconditionally guaranteed.

- Stereo model also available.



With automatic level control

Unconditionally guaranteed to outperform ordinary compressors, or AGCs, the solid-state Audimax III offers the ultimate in automatic level control for recording engineers.

The exclusive **Gain Platform** principle permits gain to remain on a stable plateau over a wide range of input levels rather than the continuous rise and fall — with consequent distortion, thumping and pumping, and audio "holes" — so frequently encountered with ordinary AGC amplifiers. Its unique **Gated Gain Stabilizer** acts to bridge through program lapses and thus eliminates "swish-up" of background noise.

A special **Return-to-Zero** feature returns gain to normal during standby conditions. In short, the Audimax reacts to any gain situation in exactly the same manner as an engineer. Price: $625. Stereo model also available — $1,250.

With the world's first 60 dB-wide linear scale meter

The Model 600 Wide Range Program Monitor is the first monitor to combine the latest developments in audio technology with the reliability of solid-state design for accurate audio measurement and analysis on a **single** 60 dB-wide scale.

Where the standard Volume Indicator measures only the top 23 dB of signal level logarithmically, this new Program Monitor displays information from +3 to −57 dB on a single **linear,** decibel scale, thus permitting accurate reading of low level audio material as well as line noise during program pauses. The 0 dB reference settings are adjustable from +18 to −22 dBm.

The 600 is also equipped with a separate DC output for graphic logging over the full 60 dB range or to drive a second meter for remote monitoring.

While not intended as a replacement for the standard Volume Indicator, the 600's meter ballistics are such that its readings are compatible with VU indications. It's a practical program monitor as well as a valuable measuring tool.

It is also available in a standard 19-inch mounting rack from which it can be easily removed for portable use. Price: $505. Rack mounted: $550.

FOR FURTHER INFORMATION, WRITE:

PROFESSIONAL PRODUCTS

CBS LABORATORIES

A Division of Columbia Broadcasting System, Inc. 227 High Ridge Road, Stamford, Connecticut 06905

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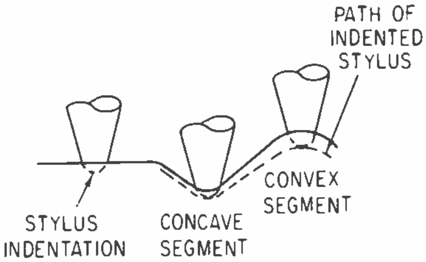


Figure 2. A playback stylus tracing a short wavelength.

ture increases the *difference* in indentation between concave and convex segments increases. The dotted lines of FIGURES 2 and 3 trace the path of the stylus tip, at its indented point of contact, with the groove wall for a long wavelength and for a short wavelength. The playback amplitude of the short wavelength is attenuated due to the differential indentation—although the recorded amplitude is the same in both cases.

Static groove indentation is affected by three factors.

Stiffness of the record material. If the record were infinitely stiff, no indentation and therefore no wavelength losees would occur. On the other hand, the less stiff (more compliant) the record material the greater will be the playback loss at short wavelengths. Since lacquer discs are less stiff than vinyl, playback

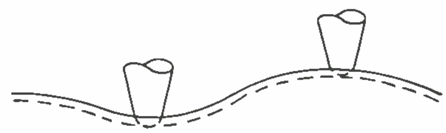


Figure 3. The same stylus as in Figure 2. This time it is tracing a long wavelength.

loss is more evident on the lacquer than the vinyl. This accounts for the difference in high frequencies sometimes encountered when comparing a lacquer recording with the identical recording on a vinyl pressing.

Playback tip radius. As the tip radius is reduced the differences in groove curvature becomes relatively less important. FIGURE 4 describes this effect. A very small tip radius, and a very large tip radius are shown tracing the same short wavelength. The indentation of the small tip is relatively uniform between convex and concave segments, and corresponds to little or no playback loss. The large tip shows a large difference in indentation between the segments and corresponds to a significant playback loss.

Tracking force. If the tracking force were zero, there would be no playback loss. As tracking force increases, indentation effects become more pronounced and playback losses increase.

PLAYBACK LOSS CURVES

Playback loss is best illustrated by the pickup response to a constant-frequency, variable-wavelength recording. This recording is made by feeding a low-level high-frequency tone to the recording head, and allowing the head to spiral in as far as is mechanically possible. Next the record is played back and the pickup output is plotted; the resulting curve will look like that shown in FIGURE 5. There is another way of displaying playback loss. In this case we record a glide tone (such as that found on the CBS STR-100 test record) at the minimum possible diameter, at 33-1/3 rpm. By playing the sweep back at 16-2/3 rpm we avoid the resonant region of the cartridge, and obtain a reasonably flat frequency response. When we plot the playback output we get the curve shown in FIGURE 6. This curve looks like a standard frequency response curve, which it is, but it is valid only for that one diameter. If the same recording and playback were made at the maximum record diameter the resulting response curve would be relatively flat to 15 kHz.

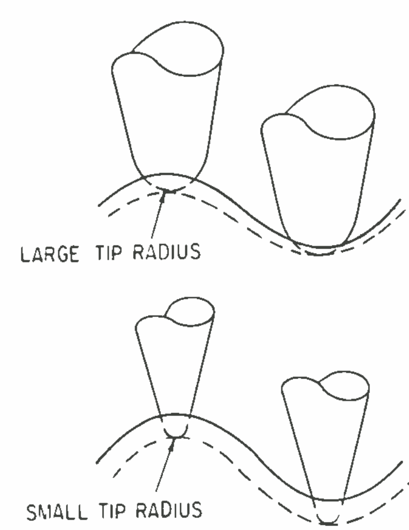


Figure 4. A comparison of large and small tip radii as they trace a short wavelength.

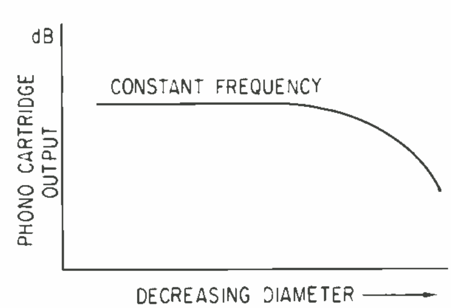


Figure 5. Playback output as it appears when tracing a variable-wavelength recording at constant velocity.

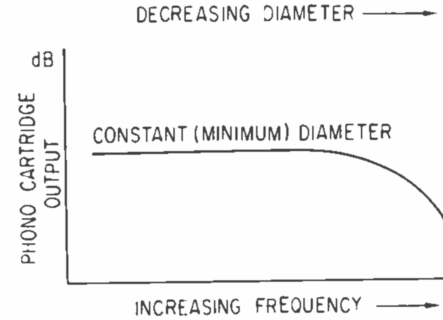


Figure 6. The same situation as in Figure 5 except that the disc diameter is the constant factor.

If you buy a "stock" console, be sure you know how it's stocked

By the time you finish adding what you really want, your "stock" price is a long way off your budgeted figure.

That's why we don't have a "stock" console. Nobody keeps things like consoles just sitting around. Instead, we have a basic plan. The RC 168. And it's a lot less basic than anybody elses.

First of all, our basic plan is really *planned*. Thoroughly. From every standpoint.

For instance, it comes with complete patching. And, any input may be audexed with any output channel. And, the wiring is all concealed. And, there are three-knob equalizers on all the inputs.

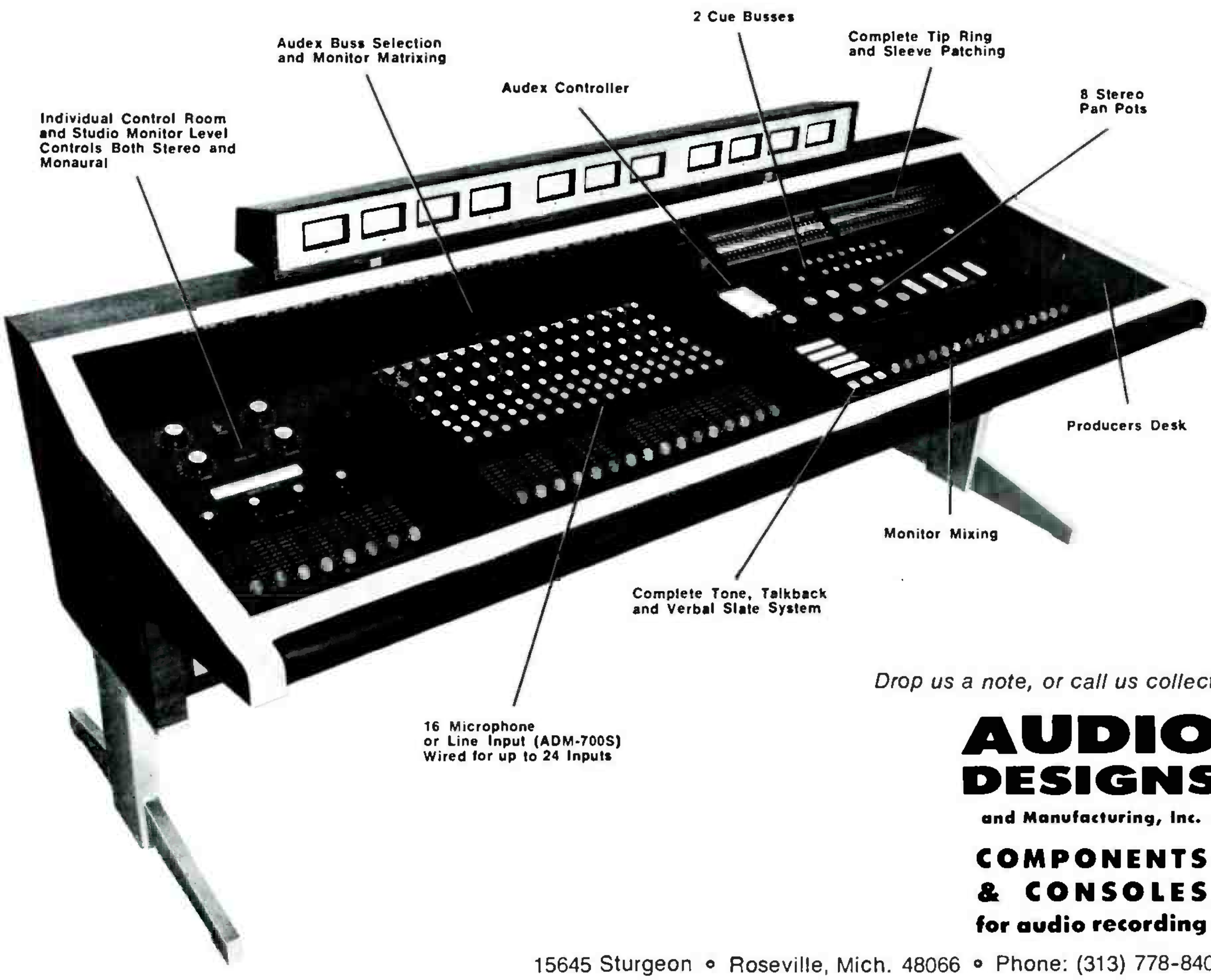
Moreover, this is one basic plan that stays effective for a long, long time. It's a *true* 8-track . . . meaning it will record all 8 channels plus stereo and monaural at once! And we've designed in an expansion plan to give you up to 24 inputs at minimum cost and no additional wiring.

What's more, we never forgot that a mixer only has so many fingers, so long a reach, and can only see so many things at once. So, we made it easier for him. It's called "human engineering."

Like eight pan pots. And slide pot monitor mixing. And an eye level meter turret. And a complete monitoring system. And more.

You can buy a console for less. But you get less console.

We'd like to tell you more about our RC 168. Because there's lots more to tell. And when you hear our prices, you'll know it's worth hearing about.



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From this discussion of playback wavelength loss we can see that because of the indentation of the disc at short wavelengths, the tip radius and the tracking force set an upper frequency limit to the playback system at inner diameters. Extension of the playback bandwidth involves reduction of both of these quantities.

SUMMARY

The upper frequency limit of the disc recording, playback system depends upon a number of factors and upon the recorded diameters. These factors are: Frequency response of the component electromechanical devices; wavelength response of the recording process; and wavelength response of the playback process.

Current best practice indicates that at outer diameters the system bandwidth extends to at least 15 kHz without significant losses; at minimum diameters losses are about 3 to 4 dB at 15 kHz. ■

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Products of unusual interest to the Audio Engineer

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Checks frequency and phase response, noise and distortion, speed variation. Designed to provide a precise and rapid evaluation of disc playback equipment. 45 rpm disc of highest quality, no other equalization is required to interpret the recorded square waves. The 1 kHz square wave has a tilt of less than 1% at a velocity of 7 cm/sec. Overshoot and ringing purely a function of replay.
1 kHz sine wave useful for level and distortion measurements. Typical distortion measurements (using a 500Hz high pass filter) less than 2%.
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A most useful tool for evaluating any disc playback system.
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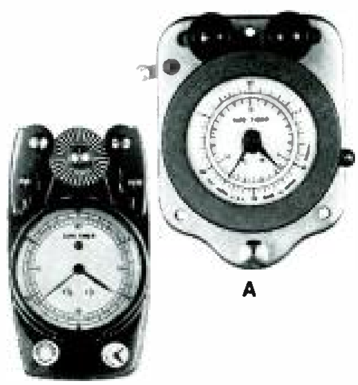
Utilizes advanced degaussing techniques with results equalled only by the most expensive laboratory units. If you use cassettes and require thorough and complete erasure of recorded material then you must try the ERASETTE 200B. You will be delighted at the performance of this amazing, yet inexpensive unit.
ERASETTE Model 200B—$15.95

TAPE TIMERS



A

B

The recorded portion of the magnetic tape can be read at a glance by a scale division of ¼ second as accurately as a clock.
The performance of the TAPE-TIMER synchronized with the tape, prevents such errors as caused by the elongation or contraction of the tape, and by the variation of speed in the rotation of the machine. Fast forwarding of the tape involves the proportional increase of the advance on the TAPE-TIMER. When you rewind the tape, the pointer will be automatically moved back by the space of time exactly corresponding to the rewound length. You are free to stop, rewind, fast forward, or forward the tape without deranging the timing on the machine, thus prohibiting errors. Enables you to simplify the most complex tape editings.
Every rotating part is provided with a precise ball bearing, so that the TAPE-TIMER can be employed at high speed with no need of lubrication.
This trouble-free high precision TAPE-TIMER, within an error of 2/1000, can be simply fitted to any recording or editing machine.
(A) Professional High Speed Tape Timer (7½-15 I.P.S.)—$99.95
(B) Standard Model (3¾-7½ I.P.S.)—$49.95

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Specifications: Casing solid brass with gleaming finish, black dial with white numbers and lettering, red tipped pointer, casing is drilled for wall mounting. Usable in temperatures up to 230 degrees. Dial is direct reading without any calculations whatsoever. Size 6 inches overall, dial face 5 inches in diameter.—**$29.95**

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Extremely versatile amplifier module for a wide variety of uses. Boasts extremely low harmonic distortion of less than .02% at maximum rated output. Uses nine silicon epitaxial planar transistors in unique circuit with over 60 dB of negative feedback plus a constant current load to the drive stage using two transistor circuit instead of the usual bootstrapping technique. High performance, low cost, small size and utmost reliability. Two year replacement warranty.
Power output: 20 watts continuous from 30 volt supply into 4 ohm load, 15 watts into 8 ohm load.
Frequency response: 30 to 300 kHz ±1dB.
Distortion: Less than 0.02% at 1000 Hz at full output and all lower power levels. Distortion on each end of the audio spectrum remains appropriately low.

- **Dimensions:** 3½ x 2¼ x ½ inches
- **Noise figure:** —70 dB unweighted
- **Input impedance:** 100k ohms
- **Supply voltage:** 8 to 35 volts DC
- **Damping factor:** 500

—$15.95

PZ-6 Regulated Power Supply for 35 volts up to 1.4 amps sufficient to drive a pair of Z-30's—$23.95
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Theory and Practice

NORMAN H. CROWHURST

● The previous discussion just about covered the main points of difference with regard to bass speakers. Now we come to something that should be easier: the speakers for the treble, or high frequencies, right? Prior to the advent of stereo, maybe, but not since. This fact was probably demonstrated, as well as any time, by the 3-speaker system introduced by CBS Labs more than a decade ago.

In that system, the bass—below some frequency such as 250 Hz—was channeled from both stereo channels to a single woofer, which could be hidden under the sofa, divan, ottoman, chesterfield, davenport, or whatever you have in your home, while only the middles and highs were fed to left and right speakers. The surprise for most people was that they could not tell that the lows were not also really coming from those same little speakers that handled the rest of the spectrum.

Back in monophonic days, the objective had been to get all the frequencies into the listening area, somehow, and with uniform response at all frequencies. If some frequencies came out "sideways" and others "front-on", nobody really bothered, so long as they came out *equal*. But stereo added a new dimension to change all that.

In talking about stereo, I am assuming that you have passed the stage of judging it by the ping-pong effect, produced by some of the early demonstration records, where a trumpet from the left would be answered by a saxophone from the right. Nowadays, for most of the time, all the orchestra plays at once, or according to the normal score at least, and stereo's job is to reproduce this normal happening realistically.

Much work was done, in the early days, determining the accuracy with which a listener could pin-point each source of sound. But when the chips are down, does it really matter whether the first clarinet is 5 or 10 degrees to the left of center? Is it a tragedy if it wanders as far as 15 degrees off-center, so long as it sounds like a clarinet?

General concensus seems to be that the ultimate objective was not to identify the exact location of every instrument or other source of sound, as much as it was to add a better illusion of realism to the reproduction. With this in view, the exact position apparently occupied by every instrument is not so important as having them sound like real, separate, individual instruments.

In monophonic days, the only separation possible was between one frequency and another, as effected by crossovers. Stereo brings in a whole new concept of separation: between different musical instruments or sources. A clarinet and a violin use most of the same frequencies. To separate such instruments, all the frequencies associated with each instrument must be delivered to different units, or uniformly distributed between them, so the instruments sound as if each is in one piece.

This requires smoothness of frequency response for a quite different reason from what made it desirable for monophonic. Failure of smooth performance on either or both stereo units will result in some of the frequencies belonging to the violin being delivered to the clarinet, or vice versa. Both sounds will appear deteriorated, or effective separation will suffer.

But beyond this general requirement of smoothness of response, which has been brought within reach by modern

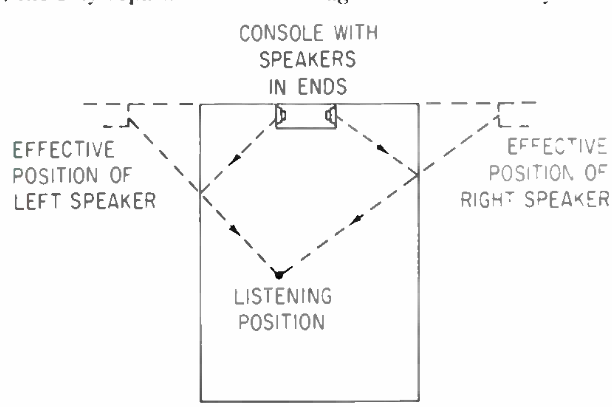

Figure 1. How to use sound reflection to help you, instead of having it fight you. The small console, with its sound reflected from the walls, makes the speakers sound as if they were further apart than is possible in a room of this size.

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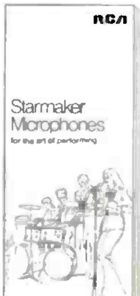
To get specific, ask for the new Starmaker brochure 1S1056 at your RCA microphone distributor or, write: RCA Electronic Components, Commercial Engineering, Section 166 H, Harrison, New Jersey 07029.



RCA

And, remember, for further professional needs, RCA also produces the renowned BK and SK microphone lines.

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technology, computer-aided design and precision production methods, that notion of a single ideal loudspeaker, that performs magic in any listening environment, to which we referred in our previous discussion, is even further from reality than it was 10 years ago.

Whatever is a good system—smooth frequency response—and gives good stereo in one particular environment, may not even sound smooth in a different environment. To achieve stereo, the system must use the acoustic conditions provided by the environment, not try to fight them.

An ideal environment is fairly dead, but not too dead—like an anechoic chamber. As nobody builds anechoic chambers to live in, we need not worry about that extreme. A well-furnished room, with heavy, wall-to-wall carpet, overstuffed furniture in abundance, heavy drapes, acoustic tile ceiling, is pleasantly dead.

In this environment, the classical stereo theory works fine. You put speaker units, which should be simple pressure radiators, appropriately spaced along a wall where you want to visualize the stage, and you are in business. The modern, relatively small, well-engineered speakers serve this purpose well (even CU's check-rated model).

But downstairs maybe you have a recreation room, where you would also like to listen to stereo, and that is another story. Because there you have tiled floor, bare or painted plaster walls —between the plate glass picture windows—and wrought-iron furniture with hessian slung over the iron for comfort: it may be as comfortable as the overstuffed variety, but it is not as acoustically absorbent!

You put the same system that gave good stereo upstairs in here, and the sound batters round the walls, any sense of stereo lost completely. In fact, mono sounds better, using only one speaker. With all that echo, it sounds almost like stereo—more like stereo than stereo, in fact! Does this mean that true stereo is impossible in this kind of environment?

Not necessarily. You see, the classic arrangement does not want this kind of echo to make it work: the reverberation fights you. The trick is to make reverberation your ally, or at least to stop it fighting you. You have two choices.

One of these is the neat little console stereo that seems to have absolutely no separation when you play it upstairs, because the speakers are too close together. But downstairs, you put the console so the speakers bounce the sound off the reverberant walls that were fighting you the other way: use them! Now you have separation—more than the room would otherwise allow (Figure 1). One room's meat is another room's poison!

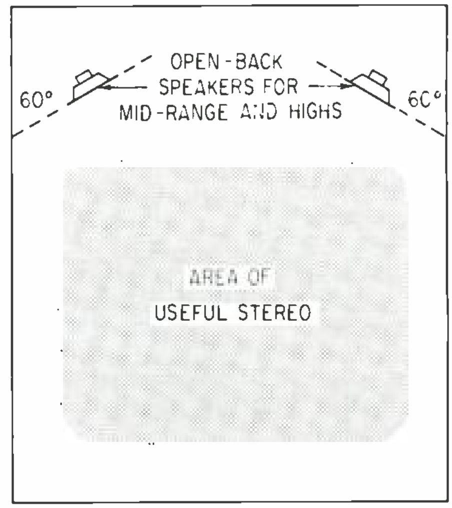

Figure 2. The CBS "isophonic" arrangement, demonstrated over 10 years ago, uses open-backed units, angled and spaced so most of the room receives effective stereo.

Another way utilizes an adaptation of that idea we referred to earlier. The rest of that CBS idea used dipole speakers for the middle and upper frequencies: speakers with open backs. They called them *isophonic*, as used in this way. For the purpose, they arranged the speakers at a special angle (Figure 2), such that their radiation produced interacting particle-velocity fields, throughout the room.

The paper the CBS people presented showed mathematically, that by choosing this particular angle, in terms of accurate location of sound sources portrayed in the program, an acceptably good stereo could be achieved through a major portion of the listening area, which the then-orthodox and often antisocial conventional arrangement did not provide.

Many ideas originating in CBS Labs have been quite sophisticated—perhaps too sophisticated for the average American to appreciate. So their three speaker system never caught on at the time. Many allowed the fact that they knew the lows really came from under the sofa to prejudice them into thinking that the low end "lost separation."

Good program material for checking this is plucked bass strings. That will give you good location of sound, but not by the fundamental low note

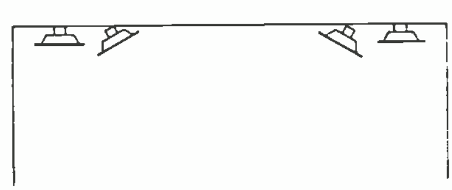

Figure 3. Open-backed units should not be placed flat against a wall, or even angled, close to a wall, as shown here: it destroys their effect.

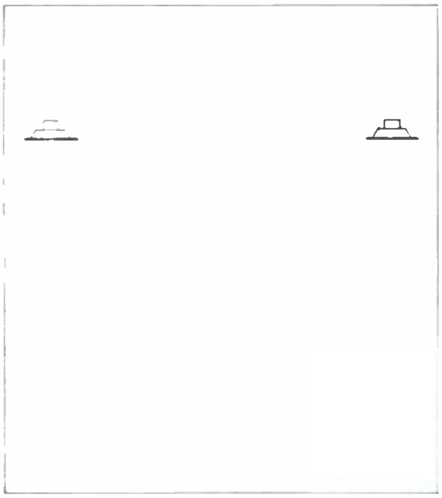

Figure 4. An arrangement that fills a reflective room with good stereo quite effectively. The open-backed units can be housed in open-type room dividers, the rest of which may be occupied by bookshelves, ornamental panelling, or whatever the home-owner fancies.

played. Rather it is the pluck tone, with the follow-through overtones, that cue you. But try listening to an organ record with good pedal bass. Can you locate the source of the bass, left, right, or center, on any system—or even in the original auditorium? It seems to fill the room—that is the nature of that kind of bass, which is all you will really find below about 250 Hz, if you exclude everything above.

So psychology is needed for selling, too.

But what is virtually an adaptation of the same principle is psychologically more acceptable: make those dipoles a little bigger, so they radiate the lows adequately in the immediate vicinity, as a velocity, rather than a pressure wave: normal propagation does not develop for a few wavelengths from a dipole speaker, and then at much reduced level, compared with an equivalent pressure radiator.

All the frequencies now get the same treatment. This is what the so-called *planar* and *co-planar* speakers do.

The ideal angling specified by CBS may be a little difficult to do, in the average room and using the bigger units. To work properly, a dipole should be free and clear of walls. Putting it with its back to a wall, or even angled (FIGURE 3) loses much of its apparent output, particularly at the bass end, because the proper velocity effect is destroyed.

But put the same units edge-on to the side walls, a little way forward from one end of the room (FIGURE 4) and you really have something. Now the "CBS effect" is working again. The wall, symmetrically abutting one edge of the unit, increases its effective size, improving its bass capability. A good stereo illusion, even in a room with fairly high reflectivity, like our recreation room, can come from this arrangement.

This arrangement is also easy to integrate with the decor: just put room dividers, with shelves, decorative panels, or what-have-you, where you want to mount the speakers, which should be about ear-level.

In this situation, the realism is realized because you are being influenced before the reverberation gets to you. The velocity effect of the combined fields from the two speakers reaches you at a higher level than the reverberation. After this little bit of high-level action has convinced you where each musical component of the sound is supposed to be, whether this is accurate or not in precise degree, the relatively low-level reverberation merely strikes you as being normal for the room.

The traditional pressure radiators do not have this advantage, because they can only produce normal propagation, right from their very fronts: they have no accentuated velocity effects to play tricks with your hearing. True the dipole, or co-planar type (and you must use a pair of them symmetrically, for it to work) achieves this effect by an acoustical trick. But isn't getting the most satisfactory illusion what high fidelity—stereo or what-have-you—is all about? ■

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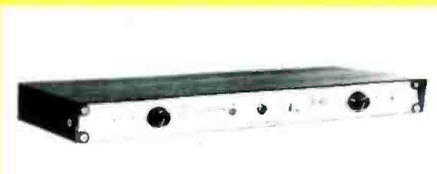

● Into a $1\frac{3}{4}$-inch rack mount form is set a basic stereo amplifier that will deliver 30 watts per channel into 8-Ω loads at 0.05 per cent t.h.d. Hum and noise is 100 dB down from full output, bandwidth ±1 dB is 5-50,000 Hz at full output and complete stability at any load is assured. The circuitry is fully protected against wrong loads, or no loads and will drive any load from 4Ω to infinity. Dual channel level controls and headphone jack are front-panel mounted. There are two rear-panel output and input jacks per channel.
Mfr: Crown International
Price: $225
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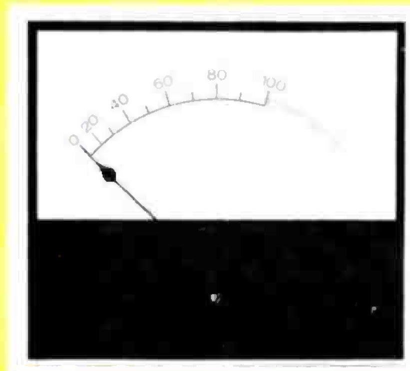
VU METER



● All the standards of ASA Standard C16.5-1954 are met by the model 7045. New design permits mounting on the panel front, or behind the panel with either an optional bezel or lens kit. The meter has a rugged phenolic case and front, with a glass window which is scratch-proof and free of static electricity. Size is $4\frac{1}{2}$ inches. Either A-type or B-type scales may be specified depending on the need.
Mfr: API Instruments Co.
Circle 65 on Reader Service Card

CASSETTE DEGAUSER

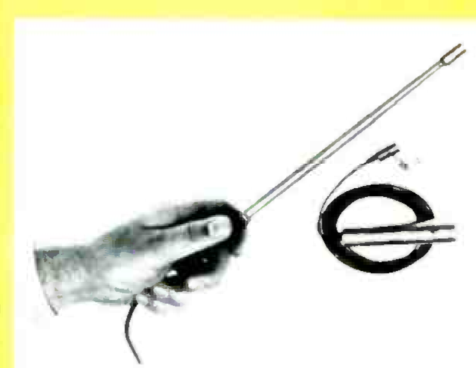
● A new battery-operated degaussing and erasing unit, designed expressly for cassettes has recently been marketed. Recorded signals at full tape saturation are removed to −65 dB. Size of the unit is 4 x $3\frac{1}{2}$ x 2+ inches. The 200B Erasette uses four AA batteries in a self-contained battery pack. The total system is thus self-contained, and a self-storing plastic handle is included to facilitate the handling of cassettes.
Mfr: Magnesonics Corp.
Price: $15.95
Circle 57 on Reader Service Card

2-INCH TAPE SPLICER

● Editing of two-inch audio tape is simplified with the introduction of the KA-2 editing kit. With this kit, editing of wide tape can be done with less than a one-thousandth of an inch spacing between the tape ends. In addition to this block, which is machined from aluminum and uses four retaining fingers; a one-inch tape block also is available.
Mfr: Joel Tall, Inc. (Elpa Marketing)
Price: $75 (two-inch)
Circle 55 on Reader Service Card

ELECTRET CONDENSERS

● Two similar models, ECM-50 and ECM-51 share excellent transient response and sensitivity. Both are omnidirection mics designed for professional use. The ECM-50 is a lavalier tie-tack less than ⅜-inch long and weighing under 1 oz. The battery power supply, transformer and Cannon output connector are contained in a separate package interconnected by a 10-foot cable. Frequency response is 50-16,000 Hz. The EMC-51 has been designed for on-the-spot recording of news-type events. A telescopic wand acts as an extension of the reporter's arm. Frequency response is the same as the ECM-50 model. Both mics use electret condenser principles of operation; for both, battery life is up to 6000 hours with a mercury cell.
Mfr: Sony-Superscope
Price: $195 (each model)
Circle 53 on Reader Service Card.

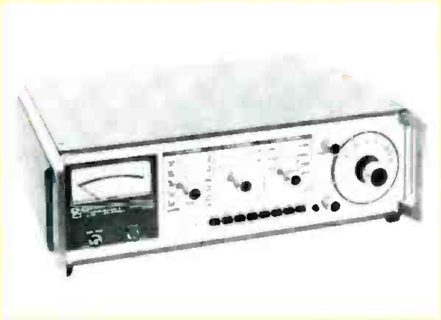


← *Circle 27 on Reader Service Card*

DOLBY B SYSTEM

● This unit, though designed primarily for home use may find its way into limited use professionally. It utilizes the Dolby B circuitry which is single-band and only offers reduction of high-frequency noise (hiss). Although manufactured under license, quality is maintained and Dolby operation within the single channel is carefully controlled for balance. The model 100 is two-channel (stereo) with both record and play compensation circuits built-in and selectable separately. The degree of noise reduction is as much as 3 dB at 600 Hz, 6 dB at 1200 Hz, and 10 dB at 4000 Hz and above. Level balance meters are included as are both a reel and cassette tape of precisely-recorded level to assure proper match of the individual recorder player to the model 100.
Mfr: Advent Corp.
Price: $250
Circle 72 on Reader Service Card

SYSTEM ANALYZER

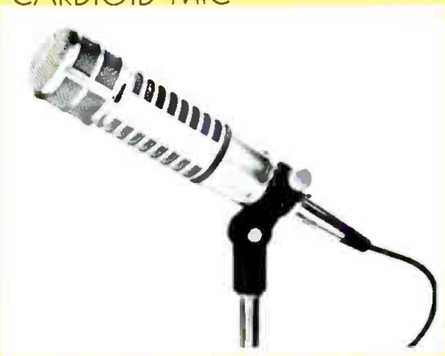


● Here is a multi-purpose solid-state instrument for measuring frequency response and harmonic distortion. The model 6100A System Analyzer contains a low-distortion sine-wave oscillator and a.c. voltmeter. Both cover a range of 10 Hz to 10 mHz. The 2nd and 3rd harmonic analyzer is unusual in that it has both automatic frequency control and also automatic level control. This removes the need for manual resetting of the reference each time input levels are changed. H.d. measurements may be made from 100 Hz to 1 mHz on full scale ranges of 10, 3, 1, and 0.3 per cent. Originally designed for tape recorder measurements, the device is also suitable as a general purpose test instrument for studio and broadcast use.
Mfr: Data Measurements Corp.
Circle 59 on Reader Service Card

CASSETTE DUPLICATOR

● C-1000 is the designation for this new duplicator system designed to copy from reel-to-reel to four cassettes simultaneously. The master machine is two-speed at 60 or 30 in. sec. Up to seven-inch reels are accommodated. Rewind time is approximately 30 seconds for a full reel. The track configuration is two-track mono with both tracks duplicated simultaneously. (A four-track version is also available.) Fixed record speed of the slaves is 15 in. sec. Other features include automatic tape lifters, solid state logic controls to delay cassette starts until full speed at the master is achieved, and one-button system starts. Additional slave systems are available for greater capacity.
Mfr: Pentagon Industries, Inc.
Price: $3750 two track, $4650 four track
Circle 52 on Reader Service Card.

CARDIOID MIC



● This latest dynamic is rugged, extremely wide range, and has been designed for recording studio and broadcasting critical use. The RE20 case is machined from solid steel bar stock, and the unit has built-in shock mounting and electrical shielding. It is unaffected by hard use or abuse. A built-in pop filter eliminates any breath or wind noises, and an external mount including extra shock protection is available for boom or stand use. The cardioid pattern is stated to be extremely uniform. Off-axis response is virtually as flat as on-axis, with maximum rejection designed for typical boom and stand use. A bass tilt-down switch aids in reducing studio rumble. Finish is fawn beige.
Mfr: Electro-Voice, Inc.
Circle 67 on Reader Service Card

AUTOMATED FADING

● FADEX is the name given to this device that goes into a line-level circuit to automate and control variable timing of fading in separate channels. Fade time operates in increments as short as one second and can be set anywhere from there to 29 seconds. The device also makes it possible to program a tape and have a fade initiated at any given point, either *up* or *down*. Repeatability is assured through automation and quality circuitry, and may be tandemed together and operated from a single set of controls. Illuminated pushbuttons on the face of the module always show its state. There are three versions of the FADEX: (424) combination fade up down; (424A) fade down only; and (424B) fade up only.
Mfr: Audio Designs and Mfg., Inc.
Circle 63 on Reader Service Card

LINE AMPLIFIERS

●This series offers encapsulated low-wattage output amplifiers. Three are designed to operate from 110 V a.c. into speaker impedances and the fourth operates from 12-14 V d.c. Sensitivity for full output of each is about 100 mV, input impedance is 20 k, distortion at half power is on the order of 1 per cent. The a.c. models come with power supplies and a 500 k pot built in and have an on/off switch. Bandwidth (−3 dB) of the a.c. models is 50 to 40,000 Hz or better; the d.c. model is limited to a lower range of 150 Hz.
Mfr: Pulse Dynamics Corp.
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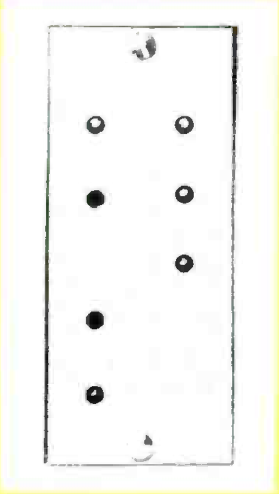
REPLACEMENT RECORD AMP



●Solid-state replacement amplifiers for various Ampex recorders are available. These are direct-replacement electronics sets that are designed to fully match the original equipment on Ampex 300, 350-351, and 354 machines. The factory pre-aligns each amplifier to typical Ampex heads. (In the case of the Ampex 354 the replacement module will be a play-only electronics set.) Replacement of original electronics with these modules is easy and take little time.
Mfr: United Research Lab. Corp.
Prices: vary according to model — $1330 for 2-track 350 electronics
Circle 71 on Reader Service Card

STEREO HEADSET

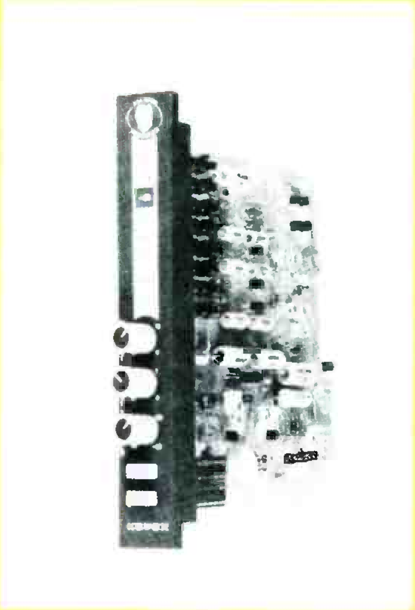
●This is not a new model, but a redesign of the 100 series stereo headsets. Internally, it has been changed to give smoother response, more rugged voice coil structure, better transients and more low-frequency output. Subjectively, the wider bandwidth is apparent. The improved models will carry the designation "A" as in 100A (17Ω), 103A (300Ω), and 106A (600Ω). The transducer is dynamic, moving coil with a Mylar cone and ceramic magnet. Power input can be up to one watt. Distortion over the audio band is stated to be less than 1 per cent.
Mfr: David Clark Co.
Circle 56 on Reader Service Card

CARTRIDGE HEAD ALIGNER

●A new concept in accurate cartridge machine head alignment makes what was impossible, easy. Precision alignment of the heads in all three axes, height zenith, and azimuth, are possible. The Collimeter 11 is built to NAB specifications and may be used with all stereo and mono machines by manufacturers, broadcasters, and service technicians. An internal light source in one mode illuminates the head for alignment of the pole pieces to the height and azimuth crosshairs. In the second mode, the indicator lamp triggers only when sensors detect the proper zenith. The instrument measures 7/8 x 1/2 x 4½ inches.
Mfr: Ramko Research
Price: $13.95
Circle 60 on Reader Service Card

PROGRAM EXPANDER

●Model 500 is a keyable program expander in which the instantaneous gain characteristics of the program material may be varied over a 60-dB range. This may be a function of the program material itself or as a function of an externally-applied keying signal. The uses are many: A multi-track tape noise reduction device of up to 60 dB; an electronic music source through modification of attack and decay characteristics; a device useful for removing echo, room sound and studio leakage, and for achieving presence; and a dynamic range expander capable of up to 60 dB of linear gain expansion. The basic module is a 1- by 7-inch illuminated epoxy-molded strip in plug-in form. The company is offering a 7½ in./sec. tape of actual programs that have been processed (before and after).
Mfr: Allison Research (Kepex)
Price: $275 (console mounting version)
Circle 51 on Reader Service Card



STEREO CONSOLE

- The QRK-5S features a plug-in cue amplifier for its five-channel stereo capability. The amplifier will drive a built-in speaker as well as front-panel phone jack. There is also a built-in 10-watt stereo monitor amplifier together with two plug-in audition amplifiers identical to that used in the line amplifiers. The console weighs all of 32 lbs. and has dimensions of 20½ x 9½ x 12 inches. Up to nine stereo audio inputs can be handled with internal mixdown to a single stereo output.

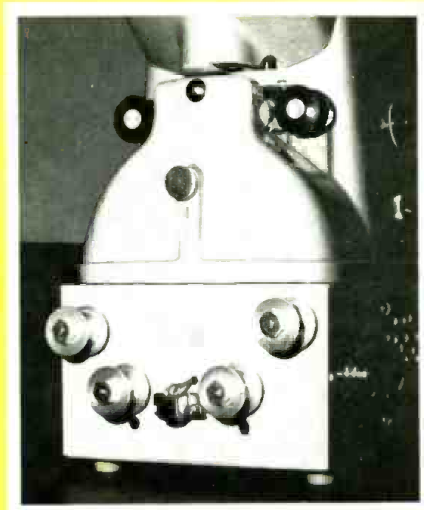
Mfr: QRK Electronic Products Inc.
Price: $1595
Circle 62 on Reader Service Card

LINEAR MOTION FADER

- A recently developed attenuator uses no sliding contacts, thus making it impossible to introduce such noise. The control is unaffected by dirt, dust, corrosive atmosphere, or humidity. Attenuation is stepless and smooth, with infinite resolution. The usual insertion loss is notably reduced.

Mfr: Moser Development Co.
Circle 58 on Reader Service Card

SOUND READER



- A transistorized sound reader for 16-mm magnetic film is now available that offers instant warm-up, one enclosure for sound head, amplifier, and speaker, and easy mounting to the base of most film viewers.

Mfr: Satellite Film Service
Price: $49.95
Circle 68 on Reader Service Card

REVISED LIMITER

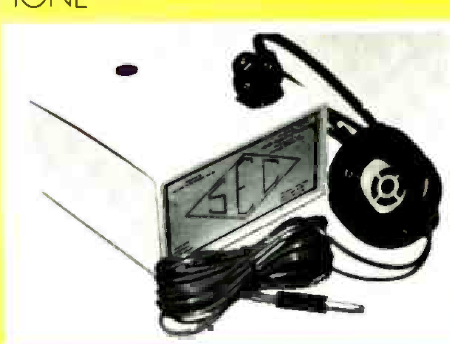
- The 1176 limiting amplifier is now available with a further reduced signal/noise of 6 dB (now at −81 dBm) thus putting it below the level of audibility. In listening tests, this is said to be confirmed. The new model is now designated 1176LN and may be identified by the contemporarily styled black front panel with white letters. Owners of older 1176's will be pleased to note that a low-noise retrofit kit is available for most of the units now in use. The kit, designated 1176RFK will sell for $40.00 and factory installation is available.

Mfr: UREI
Price: $489
Circle 69 on Reader Service Card

EARPHONE

- Model MRH is designed primarily for recording studio operators requiring a single earphone. The unit features 2000Ω impedance for use where multiple headsets must bridge low impedance cue lines. Twenty feet of flexible cable with standard phone plugs are included. The phones are cushioned with padded vinyl.

Mfr: Studio Engineering Consultants
Price: $9.95
Circle 64 on Reader Service Card



EIGHT-TRACK PACKAGE

- The JH-8 is a compact package that offers meters, control modules, and equalizers, all in plug-in configuration. The close placement of illuminated meters and controls facilitates easy determination of volume levels and channel operations status. The system features reproduction and overdub level calibration and high track-to-track uniformity. A 3/16-inch cold rolled steel frame provides ample support for the mounting. The system is expandable to sixteen and twenty-four track configurations.

Mfr: MCI, Inc.
Circle 61 on Reader Service Card

PHONO CARTRIDGE

- A new cartridge with an elliptical playback stylus has been introduced as the model 681SE. It is intended to fill the intermediate range for which the standard ellipticals are too sensitive, and the spherical configurations give less performance than the elliptical. Dependability from a broadcaster's point of view is a primary factor in this model. The trade-off of reliability against quality is resolved, giving the broadcaster the best of both worlds. Other 681 series features including the groove brush built into the removable stylus assembly, excellent shielding, and high output are retained.

Mfr: Stanton Magnetics Inc.
Circle 54 on Reader Service Card

Four-Channel Stereo

JOHN EARGLE

Four-channel stereo sound is here to stay. The consumer market can be depended on to demand more and more of it. However, the methods and configurations in which it will be given to the public are not so sure. A recent N. Y. AES Section meeting was given over to the exploration of proposed playback systems.

On March 17, 1970, it was my privilege to be the moderator at a New York Section AES meeting devoted to four-channel stereo sound. Picking a group of panelists for the evening was particularly difficult, because there are so many qualified people in the several disciplines making up the four-channel scene. We had Jim Cunningham, whose forte is miking techniques; Jerry Minter, who has pioneered in ultra-short wave-length disc recording; Peter Scheiber, whose compatible "4-2-4" method requires only *two* transmission channels; and Leonard Feldman, who with Bill Halstead, has proposed one of several FM multiplex systems for transmitting four discrete channels. We were indebted to RCA Records for providing their new Studio A for the meeting; I don't think that any other room in New York could have provided the space and facilities which were necessary to make the meeting a success.

In the following paragraphs I will review the presentations of each of the participants and give an indication of audience reactions. Following this, I will give *my* assessment of the current status and future of the four-channel stereo phenomenon.

JIM CUNNINGHAM'S PRESENTATION

I insisted on having Jim Cunningham participate in this program because of his pioneering work in the four-channel field, *Tetraphonic*, as he chooses to call it. I first heard his recordings about four years ago, and that material, some of which we heard at the meeting, is still about the most natural that I have heard anywhere.

Cunningham gave an outline of his recording philosophy and provided pertinent examples. Despite an initial mix-up in speaker assignment, the examples came off with astonishing success. It is difficult to imagine 300 people listening to four loudspeakers and hearing as much as they did. Subtlety is the essence of his approach; the rear channels are not so displaced in time and overlayed with reverberation as are many examples currently making the rounds. Rather, the four channels are balanced in such a way that listener location is not so critical. As a result, *more* people could hear the examples well.

JERRY MINTER'S PRESENTATION

Twelve years ago, Jerry Minter demonstrated a two-channel disc which had been cut with a mono cutting head. The technique was analogous to current FM practice; the sum channel was recorded in the usual analog fashion, while the *dif-*

Figure 1. The panel. Left to right they are: Jim Cunningham, Jerry Minter, moderator John Eargle, Peter Scheiber, and Leonard Feldman.

John Eargle is chief engineer of Mercury Records, New York.

ference channel was recorded, laterally as well, as a modulated carrier at a reduced level. The strides made in phonograph cartridge design in the last twelve years have made it possible for Minter to adapt the technique to the geometry of the standard stereo disc. This time, the two analog channels carry information for the left-front channels, and the two carrier channels convey the two rear, or ambience channels. Minter pointed out that the two carrier channels would be limited in both band-width and signal/noise ratio as compared with the main, or analog, channels. As it did in 1958, Minter's method requires sophisticated playback machinery. Perhaps integrated circuitry could make the system more of a commercial possibility than before. Certainly if the disc is required to convey *four discrete channels* and also be compatible, then an approach such as Minter has taken is the only one on the horizon. The system was not demonstrated at the meeting.

PETER SCHEIBER'S PRESENTATION

The mystique which had surrounded the Scheiber system for many months probably contributed in large part to the good attendance that evening. Many people were waiting for a demonstration and an explanation of the system. Demonstrations there were, but Scheiber's explanation of his method was so abstract that very few in the audience were able to grasp even its fundamentals. In reality, his system is very simple and remarkably effective on certain kinds of four-channel program material.

Scheiber is using a combination of standard matrixing and signal expansion techniques. The *encoding* of the four inputs into the two-channel mode is passive; it involves *only* the assignment of the four inputs to specific *directions* in the stereo record groove. One such array might be the conventional left-right sum-difference aspects of the conventional stereo program pair. It is characteristic of such a matrix array that the four inputs can be recovered by simple signal addition and subtraction, again in a passive manner, with each signal crosstalking into the two adjacent loudspeakers down 3 dB. Scheiber's playback system is not content with this crosstalk factor; it is constant by shifting the value from 3 dB over a wide range, and it can equal the channel isolation of a discrete four-channel system under certain input conditions.

The truly unique part of the Scheiber system is this elaborate gain-riding scheme, and Scheiber's recent efforts have been directed at optimizing the gain-riding for undetectible action on the widest variety of program input.

LEONARD FELDMAN'S PRESENTATION

Feldman and Halsted are taking that portion of an FM band normally utilized for SCA broadcasting and reassigning it to a pair of sub-carriers which are used in the transmission of two additional channels. Feldman discussed the transmission scheme in detail, and he mentioned other proposed systems for accomplishing the same ends. In the Feldman-Halsted system, the two front channels are transmitted in normal FM stereo fashion, while the two rear channels are carried by frequency modulated sub-carriers at 69 kHz and 91 kHz respectively. Feldman stated that the system is compatible with normal two-channel stereo listening to the extent that the listener in the two-channel mode will hear only the two primary channels. No demonstration of the system was given at the meeting.

THE OUTLOOK TODAY

In spite of the general bearishness of the economy, the four-channel phenomenon is gaining momentum. RCA is releasing

Figure 2. Peter Scheiber describing his system. The drawing on the board is left from an f.m. system that had been described by Leonard Feldman.

discrete four-channel material this summer using the stereo-eight format. Tracks 1, 3, 5, and 7 will be used for one program and tracks 2, 4, 6, and 8 for the other. Vanguard has been on the market for some time with open-reel in-line four-channel tapes, but the lack of playback hardware has necessarily slowed the growth of that open-reel market. RCA and Motorola will both introduce players for the *Quad-Eight* format, and those players will be compatible with conventional stereo-eight cartridges.

Right now there is a kind of "decibels *vs.* dollars" battle going on, and the Scheiber system is at the heart of that battle. Since the March meeting, the Advent Corporation has become a licensee of the Scheiber system, and they have built a decoder which will sell in the $120.00 range. They have demonstrated the device with great success, and their intentions are obviously to get recording companies to issue their normal two-channel discs and cassettes in Scheiber form. In that form, the product will also play back in two-channel stereo with no degradation, thus going away with any need for double inventory. Advent hopes to convince the recording industry that the differences between Scheiber's four channels and four discrete channels are small enough to be more than compensated by the system's two-channel compatibility. The decision should go beyond the engineering and marketing areas; record producers must evaluate the system and determine for themselves if the varying channel separation of the Scheiber system is good enough for *their* purposes.

In the months since the AES meeting there have been demonstrations of at least two proposed FM discrete four-channel systems, including the Feldman-Halsted approach. These have been local, short-range transmissions, and they have apparently been quite successful. Here in New York there are plans at station WNYC to broadcast four discrete channels experimentally, using the Feldman-Halsted system.

Still another phenomenon has come up recently; David Hafler of Dynaco, long a pioneer in audio and proponent of simplicity, has come up with an elegant but simple scheme of wiring four loudspeakers to two power amplifiers so that the speakers are fed the left, right, sum and difference aspects respectively of the two stereo programs. Now, if a record is prepared with four inputs assigned to the left, right, lateral, and vertical modes of the disc, this record will play back over Hafler's array *exactly* like the Scheiber system *without* gain-riding! Hafler claims also that his array enhances normal stereo program material by isolating the vertical, or difference, mode, which is likely to be heavily reverberant, and having it appear behind the listener. This has been demonstrated with astonishing success, and all that it takes with most systems to implement the difference channel is simply a third loudspeaker, a 10-watt 25-ohm rheostat, and some wire! Dynaco will provide details on request. (See LETTERS, page 2. Ed.)

Figure 3. During four-channel stereo demonstrations, the large crowd in RCA Victor's studio A milled around for comparative listening.

OUTLOOK FOR THE FUTURE

The disc is still king, and the only way it can be a part of the four-channel scene, at least initially, is through a matrix system. Since Scheiber has a running start on the others it is logical to assume that it is the one which will be adopted. Now comes a big question: can a record company opt for both matrix and discrete four channel systems? Putting it another way, if a company feels that the Scheiber approach is good enough for discs, how can they logically justify going to four discrete channels for their tape product? There is nothing new about coexistence in the record business, and it's not necessarily logical, either. My own hunch is that the discrete approach will dominate the two formats where it already has a foothold, reel-to-reel tape and Quad-Eight. The cassette will probably go both ways initially, while the disc will be firmly in the matrix camp. Minter's discrete four-channel disc is still pretty far away, and it can be discounted as a force in the current hardware game.

Even though FM is capable of four discrete channels, I suspect that it will fall into the matrix camp. There are two reasons for this. First converting FM to four discrete channels involves a long FCC investigation, expensive transmitter modifications, as well as getting playback equipment into the field. The second reason is that the largest program input to FM stations is the disc, and it would make no sense whatever to use a discrete system to transmit four de-matrixed channels! The ultimate direction of FM would of course depend on the nature of the input material to the stations. If the popular esthetic judgement should, after prolonged listening, lean toward four discrete channels, and if tape should depose disc as king, then FM would follow suit with one of its discrete possibilities. But this is not likely to happen in the near future.

The general enthusiasm for a matrix system is based upon its minimal investment on the part of the consumer and the fact that it will not require a double inventory at the retail level. For the record manufacturers there are virtually no changes required. Any studio could tool up for it in a day with equipment already on hand. Only time will tell whether or not the matrix approach is good enough.

Two final comments should be made concerning Scheiber recordings. The performance in normal two-channel stereo is more than satisfactory; under usual listening conditions two of the input channels will appear panned in slightly from the left loudspeaker, and the other two will appear slightly panned in from the right loudspeaker. Under precise listening conditions (a dead room with the listener located on the axis of symmetry), the listener will hear the two rear channels panned slightly *outside* the loudspeaker array. Things are not as happy in mono, and this must be of concern to record companies. With the Scheiber system, the two rear channels are down from the front pair by an 8-dB level in the mono mode. With the Dynaco (Hafler) matrix one channel disappears altogether while the other three are within 3 dB of each other. There are means of alleviating these ills at both the recording and playback ends of the chain.

As a recording engineer, I am deeply concerned about the uses which will be made of the new medium. In classical recording there is already a viable recording philosophy, and it is aimed simply at capturing more accurately than was possible with earlier systems the natural acoustical ambience of a concert hall. But classical music accounts only for 5 per cent of the record sales in this country; what of the remaining 95 per cent? At this point we have given our pop producers two additional channels to mix down to; but we haven't really given them any new creative tools. Most of the pop four-channel material currently making the rounds shows this rather pathetically. In many cases producers have simply mixed down to four essentially monophonic channels instead of the usual left-center-right that we are accustomed to in normal two-channel stereo. Others have resorted to pointless panning of tracks around the room. If we can provide the right tools, producers will be able to create ambience as they need it; they may even use several perspectives at once. New techniques are available for simulating movings sources with uncanny realism when the effect is needed. Just as the evolution of pop music in the 1960's was largely shaped by the available recording technology, so pop music of the 70's (maybe classical as well) will be molded in part by four-channel technology. One's imagination goes wild thinking of the impact which the formidable tools of electronic music, along with four-channel technology, will have in the evolution of music in this decade. ■

ELECTRICAL SPECIFICATIONS

Frequency Response Range, Typical: 15-15,000 Hz ± 2 db (10 octaves) 10-19,000 Hz ± 5 db. An individual, machine-run calibration curve accompanies each headset. This curve uses standard 3-1/2 log-cycle chart paper, and reads from 20 to 20,000 Hz only.

Sensitivity: 90 db SPL at 1kHz ± 1 db referred to 0.0002 dynes/cm^2 with 1 volt at the input. Variations from calibration furnished are less than 1/2 db at 25°C.

Total Harmonic Distortion: Less than 1/5 of 1% at 110 db SPL.

Isolation From External Noise: 40 db average through fluid-filled cushions provided as an integral part of the headset.

Power Handling Capability: Maximum continuous program material should not exceed 10 volts (12 watts) as read by an ac VTVM (Ballantine meter 310B or equal) with average indicating circuitry and rms calibrated scale; provides for transient peaks 14 db beyond the continuous level of 10 volts.

Source Impedance: Designed to work from 4-16 ohm amplifier outputs. At higher impedances response at the extremes of the frequency range will progressively reduce; e.g., 50 ohms causes a loss of 5 db at 30 and 10,000 Hz.

External Power Requirements: None, except when used for precise low level signal measurement, when external ac line can be selected by a front panel switch on the E-9 Energizer (1/16 amp, 117 VAC, 50-60 Hz normally; 234 VAC with internal strap for foreign use).

PHYSICAL SPECIFICATIONS

Size of Cup: 4-1/4" h x 3-3/4" w x 1-1/4" d.

Cushions: Fluid filled for high ambient noise isolation.

Headband: Extendable, stainless steel bands with self-adjusting pivoting yokes; conforms to any head size.

Headband Cover: Formed of wide, soft molded-rubber with 1/2" polyethylene sponge cushion on underside.

Boom Mount for Microphone: Knurled, anodized, aluminum knob on left cup with threaded shaft and 2 compressible rubber washers; accepts all standard booms.

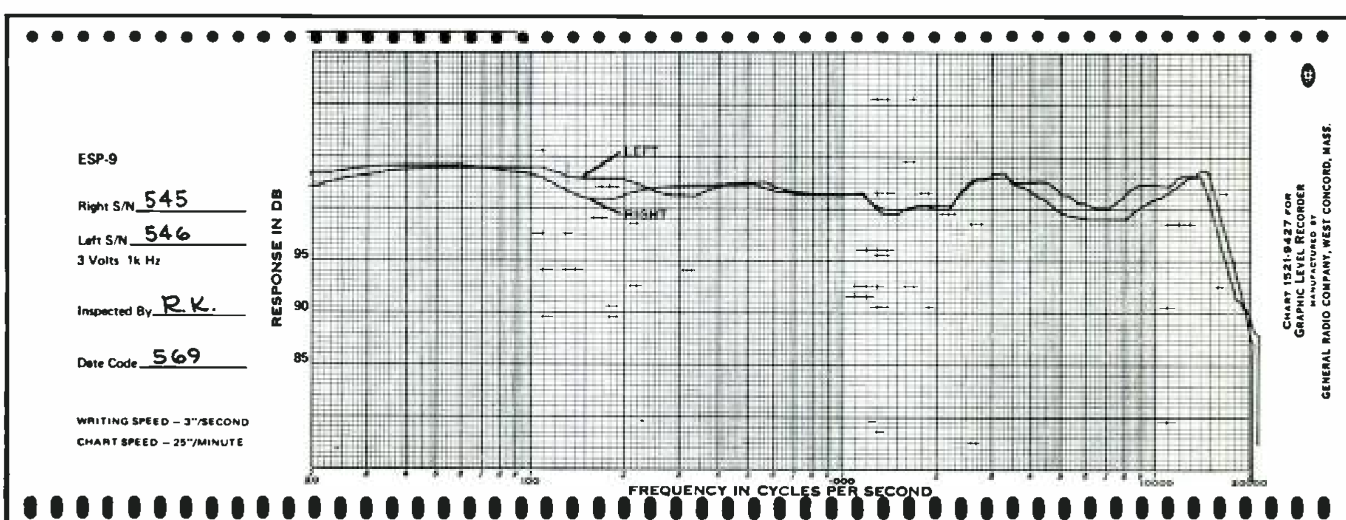
Headset Cable: Flexible, polyvinyl, 5 conductor, shielded, 6' long, black, with 5 prong plug keyed to E-9 Energizer receptacle.

Weight of Headset Only: 19 ounces

E-9 Energizer: Contains 2 coupling transformers, self-energizing circuitry, speaker/headphone transfer key-switch and ac pilot light on black anodized front panel. Also contains ac power transformer, ac on-off switch, ac line fuse, and speaker terminals. Size is 4-1/2" h x 3-3/4" w x 6-1/4" d; weight 3 pounds. Has 6' 4 conductor input cable terminated with 4 spade lugs to connect to amplifier output terminals.

Accessory Provided: 6' ac line-cord P/N 41-0235 for optional use, with plug on one end and plug-receptacle on the other.

Model ESP-9 Studio Monitor: Electrostatic Stereophones, complete with E-9 Energizer, ac line-cord, machine-run calibrated response curve and instructions; Shipping weight 6 pounds; Price **$150.00**



MACHINE RUN RESPONSE CURVE OF THE MOOEL ESP-9

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KOSS ELECTRONICS INC / MILWAUKEE, WISCONSIN
2227 NORTH THIRTY FIRST STREET / ZIP 53208

New Studios for Armed Forces Radio

ZACHARY H. JAQUETT

Armed Forces Radio was faced with special problems when they decided to build new broadcast origination studios in Washington. An acoustical-consulting and manufacturing firm was called in to help.

No one serving in the Armed Forces of the United States need go about his duties unaware of the day-to-day current of world events. Such is the mission of the American Forces Radio and Television Service, an agency of the Office of Information for the Armed Forces of the Defense Department.

The agency utilizes a broad information program through various broadcast facilities located at several points throughout the country. One of these outlets for keeping military personnel in daily contact with the world scene is AFRTS-Washington. This branch was established in 1965 to improve the flow of general and seat-of-government news through internal Department of Defense media. Its purpose is to supply timely and accurate news and informational reports, direct from the nation's capitol and from all over the globe, programmed over a 24-hour broadcast day and beamed to those thousands of places where American servicemen are stationed. To accomplish its mission, AFRTS-Washington utilizes shortwave, direct-program circuits, and teletype.

The first of these media provides live-voice broadcasts and is the only armed-forces radio facility available to military personnel in isolated outposts and aboard ships at sea.

In addition to shortwave, direct-program circuits connect AFRTS-Washington to many AFRTS networks and stations in Europe, the Pacific, and the Far East. Some of these circuits provide two-way communications, thus enabling return capability for administrative and operational information from the overseas station to the Washington office. Direct-communications circuits are used to provide optimum reliability eliminating propagation variances.

The direct teletype facilitates communication between AFRTS-W and most AFRTS networks and stations overseas. News and information, such as program-schedule changes, pertinent to the agency's operations are transmitted by teletype.

The studios of AFRTS-W are located in a new 12-story conventional office building leased by the Federal Government from private owners. When AFRTS-W prepared to occupy the structure in Arlington, Virginia, not far from the Pentagon, it had to modify the space to meet specialized requirements for broadcasting studios.

Contact was made by the chief engineer of AFRTS-W with Industrial Acoustics Company, Inc., of New York, a firm involved in designing and building acoustical noise-suppression equipment. AFRTS-W supplied the studio layout and the acoustical parameters for these chambers—conference studio, conference control room, announcers studio, auxiliary control room, main studio, main control room, sports studio, teletype room, plus two sound locks.

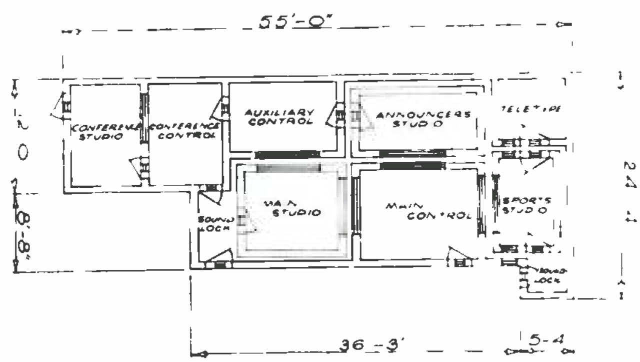

Figure 1. The layout of studios and control at AFRTS-Washington. All of the acoustic walls were constructed, as indicated in the text, of pre-manufactured wall panels.

Zachary H. Jaquett represents Industrial Acoustics Company, the manufacturers of the studio modules described in this article.

Consideration for conventional studio construction was ruled out because of floor load capacity, time for construction, and excessive floor space required. Further, conventional construction would not provide the required mobility, if it would become necessary to relocate the studios.

Selected as the outfitting materials best able to do the job were modular acoustical components made up of pre-engineered and acoustically rated noise-control panels. From these units, called the *IAC Moduline System*, a broadcast facility was designed as a building-within-a-building answering these important criteria.

1. All sections could be assembled or disassembled in minimal time.
2. The over-all weight of the eight Moduline-System chambers would be well below the load capacity of the office building.
3. Adequate working space would be provided.
4. Reliable acoustic performance was assured.

In deciding upon the Moduline-System panels, the AFRTS chief engineer was guided by two principal factors—sound isolation and reverberation time. In the acoustical environment desired for the broadcast facility, it was specified that the noise levels within the complex should not exceed NC-20, and the reverberation time would be 0.34 seconds within the sports, announcers, and main broadcast studios. To be certain before installation that the Moduline panels would meet these specifications, IAC made an octave-band analysis of noise levels at the site. From this investigation, they recommended that the walls of the above studios be of double-wall construction and single-wall for the remaining chambers. Also it was proposed to vary panel construction so the studios could be calibrated to achieve the required reverberation time.

Now for a look at the way the Moduline panels forming the roof and walls of the structure are put together to achieve their acoustical properties. All of them are four-inch-thick, variable-absorption units whose face sheets are either 22-gauge cold-rolled steel perforated with openings 3/32-in. diameter, or solid 16-gauge steel. The assembled panels are steel reinforced and filled with sound-retarding and sound-absorbing fills. These are inert, noncombustible, mildew-resistant, and verminproof. Each face sheet is welded and riveted to the panel assembly to acoustically compress and hold the filler in place.

The floor panels, whose design weight is 20 pounds per square foot, have an 11-gauge hot-rolled steel wearing surface and a 16-gauge steel back sheet. Each panel is structurally reinforced and welded to form a rugged assembly. Not affixed directly to the floor of the office building, the studios' floor rests on properly loaded vibration-isolator rails which have a natural frequency of less than seven hertz.

And since weight was a basic factor for the studio structure so as not to place loading stress on the office building, the panels checked in with an average poundage for the floor units of 18 lbs. per sq. ft. and 7 lbs. per sq. ft. for the wall, ceiling, and door panels.

In putting the panels together to form the broadcast complex, care was taken to spot weld each 16-gauge-steel panel joiner every two inches along its length. This procedure was carried out to prevent noise leakage when acoustically and structurally joining the panels. Each joiner is formed to create a labyrinth that will allow no direct passage of noise.

Furthermore, since the acoustical properties of the studios could be compromised by inadequately designed entrances, the company supplied $2\frac{1}{2}$ inch thick soundproof doors outfitted with cam-seal hinges. The advantage of these is that they supply a sure compression seal between the door and

Figure 2. A fish-eye view through an acoustical door into a portion of one of the studios.

its frame. When the door is opened, the cam action of the hinge smoothly lifts the door leaf, releasing the seal; but when it is closed, this hinge automatically lowers the compressing bottom seal tightly against the floor. This provides a positive acoustic seal every time the door closes, eliminating the need for unsafe raised sills, drag seals, and unreliable threshold closures.

To complete the acoustical picture, IAC also did away with noise from the air-conditioning through installation in the system of its *Quiet-Duct* and *Tranquil-Aire* silencers.

Since the studio complex has been in operation, its staff has had ample opportunity to find out how the enclosures are working out. The announcers have been impressed with the lack of distortion in voice transmission; it is not too live, but neither is it dead, the level being just right for broadcasting purposes. This is especially significant since all the studios are rectangular, there being no skewed walls to assist in the maintenance of a low reverberation factor. And once inside the broadcast area, the casual observer becomes quickly aware of the absence of surface noise and of the disappearance of the whine and vibration of low-flying jets coming in for landings at nearby National Airport.

Thus with these modern, acoustically-perfected facilities, American Forces Radio and Television Service-Washington continues its daily objective of keeping military personnel up-to-date. ■

Figure 3. The main studio as it is in use. Main control is on the right with the sports studio just visible through it. On the left, the view is into an auxiliary control room.

Optical Sound Track Processing

J. W. DORNER

In film sound, the best efforts of the audio engineer may be undone in the processing of the negative or positive films. This article details what controls the audio man may impose on the processing staff to maintain the quality he wants. In understanding the processing requirements—also detailed—film-sound people may be able to work more closely toward the ideal goal.

Sound is very often neglected in the production of film short subjects because people tend to think that picture quality is the most sensitive gauge by which the factors affecting the over-all quality of a process from negative to release print can be judged. Yet, much finer detail is recorded on the sound track and the ultimate in quality control is required in order to retain the original quality of the sound which is to be recorded on film.

The highest frequencies recorded on 16-mm film form an image of little spikes which are spaced only one thousandth of one inch apart. If one would project thissoundtrack under the same conditions as used for a 6 foot-wide picture, these little spikes will still be as close together as ⅜-inch on the screen. How often does one look for such detail in the picture? Processing becomes very critical if one does not want to lose part or all if the information recorded with such precision.

We all are familiar with the halo effect of lights in a night scene, which is caused by secondary exposure within the film itself from light bouncing off the film's backing. Much has been done by film manufacturers to reduce this effect. However, the high light intensity from a point-like source will always cause some reflections within the film, resulting in undesired secondary exposure. Imagine the light intensity required in producing a sound track, if you realize that the exposure time in recording a negative is as short as *one three thousandths* of a second. Secondary exposure and image growth in developing will cause a slight deformation of the original sound wave as laid down by the recording light beam in the optical recorder. Fortunately this can be counteracted by controlling the image growth that occurs in making a print, thus the shape of the original trace can be restored. However, each positive material behaves differently and only by performing a series of elaborate tests can the combination of exposures, printing lights, and developing times be determined to assure an undistorted reproduction of the fine detail recorded on the film.

The outcome of any such test holds true only for the laboratory and the material used in establishing optimum processing conditions. Therefore, if a sound track of best possible quality is sought, all recording, processing, and printing should be in the hands of one laboratory. It is essential to know prior to the recording of the negative whether release printing is to be done on black and white, or positive color or reversal color as only then is a perfect cancellation possible of the image deformations caused by the various processes.

IT HAS LONG BEEN RECOGNIZED that the quality of a variable-area soundtrack is interdependent with the image contrast on the film. To determine the optimum combination of negative and print densities for minimum image distortion (and undistorted sound) as caused by halation and fill-in, the cross modulation test has become an accepted routine in the motion-picture industry. A comparison of the test results obtained over extended periods showed considerable variations in the balance densities. Different methods of controlling the negative-developing process indicated that most consistent results are obtainable by processing the negative to a selected contrast between two steps on the sensitometric exposure strip, rather than to a prescribed gradient (gamma). The densities of the steps selected are representative of the track density and the amount of halation, thereby serving as an indicator for the gradation in the boundary area from the exposed to the unexposed portions of the soundtrack. Interchangeability and consistent quality for all soundtracks will be possible if a narrow and identical acceptance bandwidth for all photoreceptors used in the densitometry and the reproduction of photographic soundtracks is adopted and used throughout the motion-picture industry.

It may seem superfluous to write about quality control for variable-area sound recording, because if one leafs through back issues of the relavent literature one can find various papers on the subject of variable area sound recording, each one suggesting ways for better controlling the distortion products introduced by the photographic process.[1,2,3] Most papers place the emphasis on a single—but in each case different—factor, with the ultimate aim of finding the best and most reliable control procedure for producing a consistently good sound track.

Publicized data are almost always based on findings obtained from tests which were performed under extremely well controlled conditions, and can seldom be maintained in an average operating set up. But even if this were possible, there are short term variations affecting the daily production runs and they may cause the sound track quality to drift outside any acceptable tolerance limits. There must be numerous people who have performed a complete cross-modulation analysis at one time or another that indicated the excellent results obtainable under a given set of conditions, yet their production tracks may not have come off with the excellent quality as could have been expected from their tests.

Of the many variables involved from the recording of a negative, through processing and printing up to the reproduction of a print, negative developing is generally recognized as one of the most critical stages, requiring a high degree of attention and precise control, if a once optimized condition is to be maintained. However, this is not economical for very short subjects and it is felt that there is a great need for a simple method, which assures that variable-area sound negatives can be produced with uniform quality despite any minor fluctuations that may be inherent in the chemical process.

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NEGATIVE QUALITY AND CONTRAST

Ideally, the image of a variable area sound track should have a sharply defined boundary, with no transition through varying shades of gray, from the opaque track to the clear film base. To approximate such a condition as closely as possible, the manufacturer recommends to develop the negative material to a high gamma of at least 3.0 or more[1,5] and more often than not, this recommendation is religiously observed with little attention to other information revealed by the sensitometric curve. A large majority of the people involved with sound on film will certainly be satisfied with the negative developing, if a gamma figure within a tolerance from 3.0 to 3.2 or even 3.4 can be read from the H&D curve. In some instances the gradient of the so-called *straight portion* is even slightly tilted on the graph paper by the person analyzing the sensitometric data, so as to come up with the requested gamma figure. This practice can easily be followed when working with high-contrast stock which has often only three density steps with equal incremental increase between the toe and shoulder breaks. Is this really accurate enough to assure a consistent negative quality over any period of time?

Surely not, because the author has seen 35-mm variable-area negatives on Eastman 5375 with a perfectly acceptable maximum gradient slope, yet even the widest range of print densities did not produce one single acceptable print! In fact, the cross-smodulation analysis resulted in such unusual readings that the plotting of the familiar cusp-curves was not at all possible. This may have been a rare exception, but it indicated clearly to the author how meaningless a gamma figure can be.

Depending on the composition of a developer, processing time may vary to some degree between different formulas without appreciably altering the contrast gradient of the film, but there may be a decided difference in the film speed thus obtainable. Data sheets for sound recording films suggest a certain range of track densities for good over-all performance, causing some people to think that all one has to aim for is this suggested density, with little attention to the exposure and processing conditions under which it was achieved. Was it a low exposure and a long developing time—or was it the other way round?

Lewin[2] when investigating the effect of developing time on the quality of a negative track came to the important conclusion that the changes in negative quality are much greater than might be expected from any variation in the actual gamma. In practical work, developing time is frequently adjusted to arrive at a specified track density to compensate for exposure variations. Now then, if the effect of developing time is such a marked one, perhaps it would be advantageous to adjust exposure so as to suit a once established processing time which yielded a negative of good quality. Yet, in doing this, we are most likely adjusting for chemistry changes, which means in other words, that even though the chemical reaction time is now kept to a fixed value, there must be some changes in film speed and the resulting gamma. Thus, we are back at the start, trapped in a vicious circle in this search for an accurate and reliable procedure to assure a more consistent negative quality.

Data collected by the author from cross-modulation tests performed over a period of several months indicated great fluctuations in the optimum negative density for a cross-modulation product of better than minus 30 dB based on an

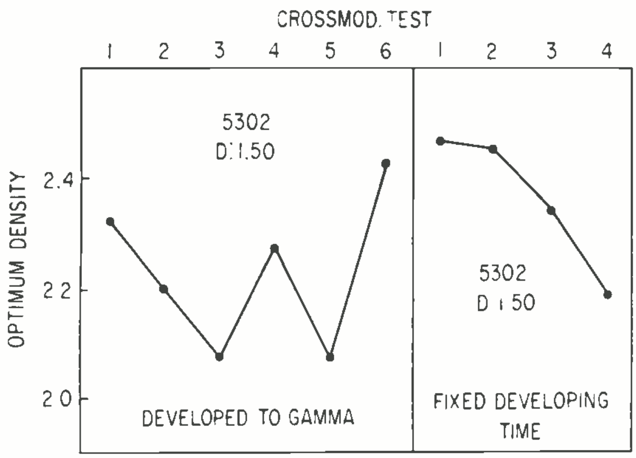

Figure 1. The method of negative developing and its effect on negative densities for maximum cross-modulation cancellation.

optimized positive density of 1.50 when processing to gamma only (FIGURE 1). In this test series, it was left to the laboratory's discretion to develop the negative as they considered best for the particular emulsion involved. As the graph clearly shows, results were very inconsistent and raised many doubts in the sound department's mind as to the quality and consistency of the daily work. Based on Lewin's findings, it was decided to change to a fixed developing time, which resulted in a marked reduction in the fluctuations of optimum densities over a series of tests (FIGURE 1). Of course, there was the occasional problem with exposure, making it necessary to decide whether to re-do the track, or to adjust the developing time. With this still being a rather unsatisfactory state of affairs, one had to ask oneself the question: Why do the optimum densities vary so much? Could it be inconsistency of the emulsions' characteristics? If optimum densities vary from test to test, then there is every possibility that they may even vary from day to day. Surely, modern manufacturing methods must be capable of turning out an emulsion which does not exhibit such erratic fluctuations. This leaves only one other possibility, namely the slight gamma differences that occur, even though the developing time is maintained at a fixed time. Assuming that this is so, then the H&D curve must contain the information, which should make more consistent sound track possible.

THE FOUR-STEP CONTRAST

The sensitometric exposure is made from an accurately

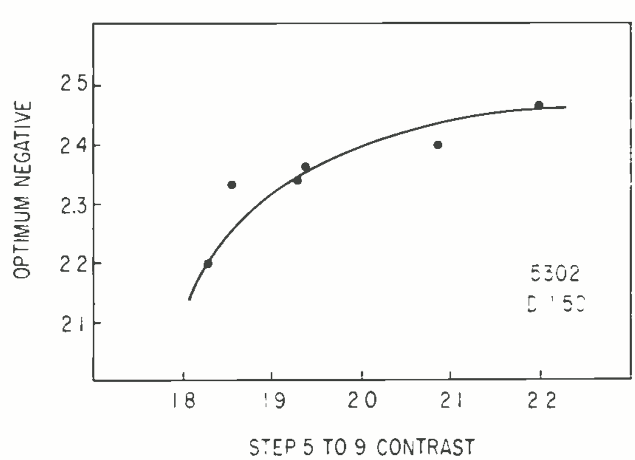

Figure 2. The relationship between Step 5—Step 9 contrast and optimum negative density.

calibrated light source and covers a wide range of light intensities or exposure times. It includes an exposure approximating that which will produce near sound track density, but there is also contained an exposure which can be taken as representative for the undesired halation, that distorts the sound image on the film. Albin, in his proposal shows some microdensitometer readings of different degrees of halation and he says: "These gradients are directly related to the gamma of an H&D curve made with an I - B densitometer but they are not necessarily equal." However, he concludes ". . . .we can safely assume that the gradient and I - B sensitometer gamma are directly proportional." In other words, we have an excellent indicator of the amount of image growth by examining the H&D curve which looks into the valleys between the spikes resulting from high-frequency sound as a magnifying glass, revealing what happens in that area where halation hurts most.

If we assume that the optimum negative density will be in the range from 2.30 to 2.50, we find that four steps down from that density, the reading on the H&D curve is approximately 0.40 to 0.50 which—for all intents and purposes—could be considered as being representative for densities produced by secondary exposure. The contrast between these steps is about 1.90 to 2.00 in density and it should be a measure for the quality of the negative track. Since a higher contrast will produce a sharper image, one must obviously be able to work with higher negative densities as the contrast increases. FIGURE 2 shows this relationship in a graphic presentation—which, of course, is valid only for the recorder, film stock, and laboratory utilized in this test series. As the graph indicates, there is even a saturation point approached as contrast increases, suggesting that a contrast of 2.00—which can easily be achieved—may be a good operating point, because this would also minimize optimum density shifts, in case some contrast variation occurs. Further tests (as well as the daily work) were then processed to this specified contrast (Step 5/0.50—Step 9/2.50) and surprisingly enough—although hopefully expected—the optimum negative density did not vary by more than a few density points from test to test and over extended periods (See FIGURE 3). These minor variations are almost within the accuracy of the densitometer and they can most likely be attributed to the fact that the prescribed densities for steps five and nine cannot always be achieved with the desired precision, thus resulting in a slightly higher or lower contrast.

EXPOSURE CONTROL

The stabilization, as indicated by the almost negligible changes in optimum density, is such a marked one, that it would seem safe to conclude that a negative of consistent quality is obtainable when processing is carried out in the manner described above. In fact, by adopting this procedure, developing attained any accuracy which made it doubtful whether or not the exposure in the sound recorder could be controlled precisely enough to avoid changing the processing conditions to accommodate exposure drifts. Any problems that could arise in that area were easily overcome by installing a solar-cell in the vicinity of the exposure lamp.[7] It samples the light and produces a voltage proportionate to the radiant energy, thus allowing precise compensation for any changes in lamp efficiency due to envelope darkening or variations in temperature. Only one very minor drawback remains: the disagreement between the spectral response of the solar-cell and the spectral sensitivity of the film, which necessitates several exposure tests during the early life of a lamp, until a stable relation between the light reading and the actual exposure on the film is reached. Once this is the case, the exposure in the recorder can be maintained with such a degree of accuracy, that any density shift of the sound track will be accompanied by a similar change in the upper control step density. This is of great benefit in the liaison between sound department and laboratory, as it will eliminate any argument as to who is at fault if a specified track density cannot be produced. Any changes in the relation between the two control steps under a given developing time are a signal for the laboratory personnel to investigate the chemistry of the developer, thereby eliminating another area of possible argument. While it is nothing new that sensitometry is being used for solution control, there may be situations where things are being allowed to drift too far. before being detected, yet with specified densities for two control steps, solution control has become automatic and is not so much a separate task any more.

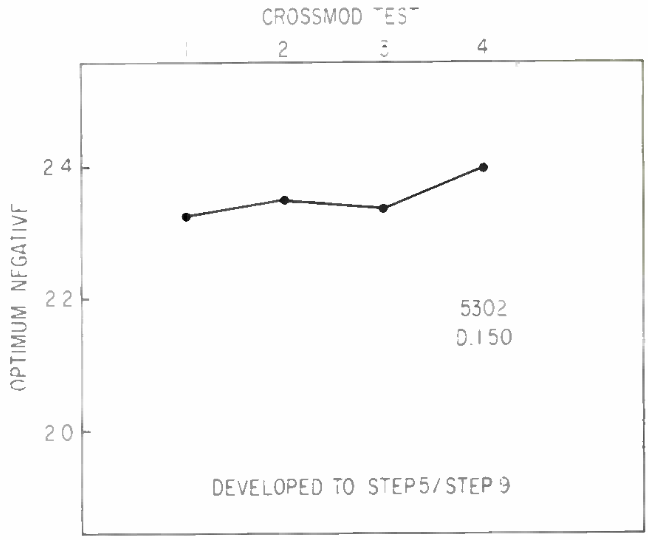

Figure 3. Stabilization of Optimum Negative Density by developing to prescribed Step 5—Step 9 densities on the sensitometric exposure strip.

EFFECT OF PHOTORECEPTOR'S SPECTRAL RESPONSE ON POSITIVE-TRACK QUALITY

The minor drawback mentioned earlier, when describing the solar cell as a light sampling device, touches upon a problem which seems to exist in many areas of optical sound recording and reproduction where photo-receptors are involved. It is unfortunate that the peak sensitivity of light sensitive devices as used in sensitometry often differs greatly from the one in the projector, where we find everything from the well-known infrared (S1) response to blue sensitive receptors (S4) as well as those which see the whole visible spectrum, including devices which read far beyond visible light. We can strive for highest accuracy in processing, yet the sound track thus produced will reproduce well only on a limited number of projectors and we shall see why.

In the literature the question of the *green track* crops up once in a while and since the writer had collected a large number of negatives and prints, this provided a good chance to determine whether or not there is any appreciable change in the quality of sound track due to aging. First results showed alarming differences over relatively short periods, suggesting that a three-month old negative would require a considerably different print density for good cross-modulation cancellation, but it was soon discovered that certain characteristics of the optical reproducers involved in this analysis could be blamed for these drastic differences. Figure 4 shows the cross-modulation distortion readings from one and the same black and white print on two different projectors. Yet, two prints taken three months apart from the same negative, performed practically identically when they were played on the same reproducer, and even when printing at half-speed, the difference was only a very insignificant one, providing they were all produced to the same density.

We are left with the question of why such confusing results were obtained from one and the same print when playing it on different projectors. The matter became even more confused when it was discovered that silver-dye tracks played equally well on the two systems involved, while black-and-white tracks showed such a marked difference.

There were several things different between these reproducers. They were in their scanning optics as well as in the type of photoreceptors used. One system allowed focusing for *A* as well as for *B* wind, whereas the other one worked with a fixed focus setting. Any effect which may have been caused by the compromise focus of the latter one could quickly be ruled out by analyzing the print on the system with the adjustable focus arrangement, because the 400 Hz component of any cross-modulation print never changed, even with the scanning optics turned completely out of focus. This was a surprise at first, as relevant literature emphasizes the need for good high-frequency peformance of the reproducer if the test results are to be valid. However, it does not seem to depend so much on whether or not the high frequencies, their sum and differences or side bands are being reproduced, because the system is only required to scan the changes in average transmission at the rate of the high-frequency amplitude variations when analyzing a cross-modulation test. A very good illustration of the changes in average transmission and the new frequency created by it is given in Baker

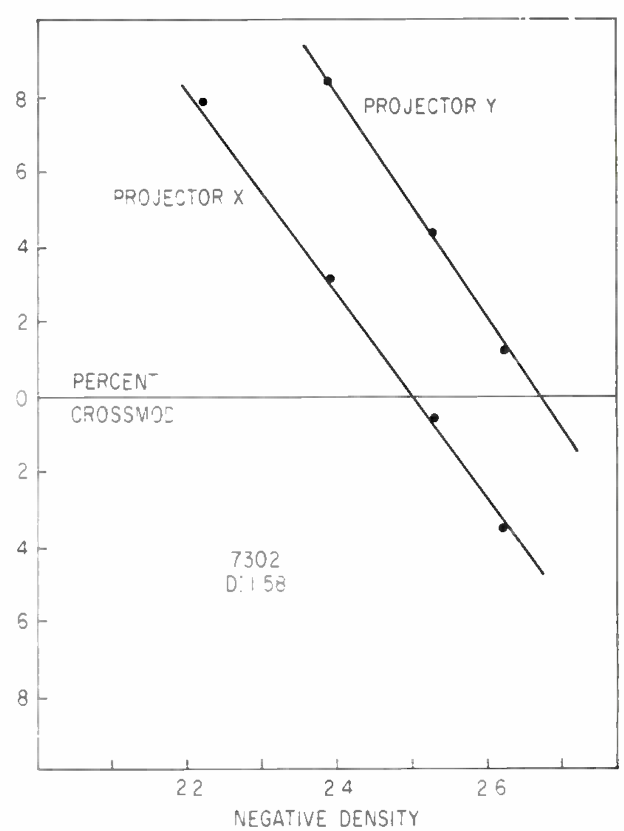

Figure 4. Cross-Modulation. The performance of a print on two projectors with photoreceptors of different spectral responses.

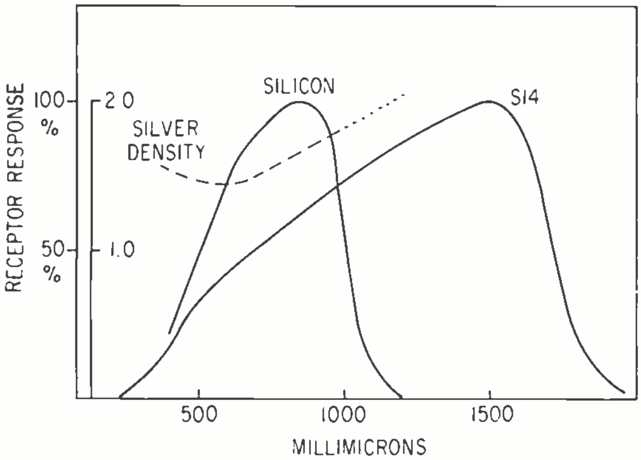

Figure 5. Spectral response characteristics of germanium P-N junction and silicon cell. spectral density of a (Black and White) silver deposit.

and Robinson's paper[1]. It is not difficult to visualize that any densitometer could detect these variations, providing the aperture dimensions were such as to read across the full width of the sound track, while covering not more than either one *on* or *off* period of the high-frequency record. In other words, in detecting the low-frequency component in the cross-modulation signal, the sound reproducer integrates the variations in the transmission characteristics of the track, and for this accurate focus is not necessary. Viewing the reproducing system for a moment as a densitometer which evaluates the image on the film under dynamic conditions, it must be assumed that the same laws apply as for static densitometry, namely, that the effective density of the photographic image will vary with the spectral response of the photoreceptor. Lovick has shown how the density of silver deposits changes as a function of light wave length. A density of 1.50 at approximately 500 millimicrons approaches 1.90 if the response of the receptor were to peak at 1,000 millimicrons and it will probably increase well beyond 2.00 at even longer wave lengths. A density change of this magnitude has the most severe effect on cross-modulation distortion, particularly on the 16mm medium.

If one and the same print can produce such widely differing results (as shown in FIGURE 4) the answer must lie in the spectral response characteristics of the various receptors employed. A closer examination of projector Y revealed that it used a germanium P-N junction as the sensing device, which has a spectral response known as S14, peaking in the region of 1,500 millimicron where the silver deposit may perhaps have an effective density of as high as 2.40 (See FIGURE 5). This would provide an explanation why the cross-modulation product differed as much from the one on projector X, working with a photovoltaic silicon receptor. Now we could also form a theory why a silver-dye track does not behave in the same manner: the dye portions of the track are still contributing to the density as seen by the silicon cell and as the dye densities diminish towards the infrared, the increasing silver density is taking over, thus keeping the effect of the image growth probably the same on both systems. Projector Y was then modified and equipped with a silicon cell which corrected and aligned its cross-modulation performance with that of type X projector, despite the different optical system.

It would appear from the results of these tests that the acceptance bandwidth of the photoreceptor has a greater influence on the quality of the reproduced sound than previously realized.

CONCLUSIONS

The success of any efforts in finding the balance dentsiies for VA-sound tracks will invariably remain confined to the combination of labortory and sound reproducing equipment under which such tests were performed, as long as recommended standard procedures do not exist.

To achieve full interchangeability of either the negative track for printing elsewhere, or the finished positive for playing on any projector without quality deterioration, standardization in the following two areas would be required.

FOR THE PRINT

The response of photoreceptors used in densitometers as well as in projectors should be defined with the same exactness as has been done in the American Standard PH 22.117-1960 for the densitometry of color sound tracks. A universal positive track density could then be specified for each printing stock, as any discrepancy between densitometers (and reproducers) would thus be virtually eliminated.

FOR THE NEGATIVE

The negative density will not require precise specification, because a variety of factors can influence the results at which cross-modulation balance occurs, if printing is done to the universal positive density. However, it will have to be strongly emphasized that accuracy and consistency can only be assured with extremely precise processing control by working to the four-step contrast method. Preferential control steps and densities should be recommended.

It is realized that there are other areas in this complex process of producing an optical sound track where quality deterioration can occur. These would have to be subject to detailed investigation and exchange of information through an over-all standardizing body. Notwithstanding this, it is the author's firm opinion that much could be gained by arriving at a recommendation as outlined above, as this would greatly increase the chances for a more consistent sound quality in the field, from a track which played once excellently in the studio's review theatre.

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The Sync Track

JOHN M. WORAM

●At the recent Consumer Electronics Show in New York City, many exhibitors featured the new four-channel sound. Lately, there has been no shortage of commentary on four-channel (or, if you prefer, quadraphonic, tetrasonic, surround stereo, wrap-around sound, ad infinitum), but with the generally run-down condition of the economy, it was surprising to find so many manufacturers actively involved in the four-channel market. With just a little help from the economy, the four-channel scene may develop faster than anyone had anticipated. And, once a four-channel disc becomes a practical reality. . . . . .

If four-channel sound is to be worth the additional expense, it will have to offer something more than just another set of speakers. No doubt, we will go through a period of ping-ping-pong-pong until the novelty wears off and recording personnel learn how to take advantage of the new medium.

Before too much more time passes, a standard speaker placement will have to be specified. Presumably, the choice will be made between rear or side speakers. Most of the four-channel demonstrations have so far located the additional speakers to the rear, yet there are some who favor a to-the-side placement. For more details on speaker placement, see James Cunningham's article, *Tetraphonic Sound* in the December 1969 issue of **db.**

Regardless of eventual speaker placement, recording engineers will develop some new techniques as the state of the four-channel art develops. To over-simplify, there are two basic approaches to four-channel recording—classical and pop. The classical approach is the more straightforward, with the additional speakers transmitting the reflected sounds that one would hear in the concert hall. However, it's simple in theory only, as the actual placement of the ambience microphones can be a very illusive art. The pop approach opens a whole new world, and if four channel does indeed catch on, its capabilities will no doubt significantly influence the direction of the rock scene. The psychedelic possibilities are obvious, and probably endless too.

PREPARING A FOUR-CHANNEL MASTER TAPE

Presumably, if your studio has a separate tape mastering console, it is presently equipped for mixing 4-,8-, or 16-track tapes down to 2-track masters. If you can spare two echo send lines, you can probably rig up a fairly good four-channel system with just a few patch cords in the following manner:

Determine the tracks that are to be placed on one of the new speakers, which I'll call left rear for this example.

With a patch cord, lift the output of the associated mixers.

Put the associated echo switch to post position, and turn the echo send full on. Now, the mixer will feed signal into the echo line, but not to the program buss.

Do the same thing with any other mixer that is to feed the left rear speaker.

Divert the output of the echo combiner, so that it feeds to the left rear track, rather than to an echo chamber.

Using a different echo line and combiner, repeat the process for those tracks to be fed to the right rear.

Of course, you also need additional monitor facilities. For the time being you might just plug into the tape machine output, and flip the selector switches on the tape machine to input.

This is certainly not the ultimate in four-channel mixing, but it will get you started in a hurry, and it means you don't have to tie up your studio's recording console for making your simpler four-channel masters. It would probably be more than adequate for mastering the classical repertoire where—once balances are established—there isn't much need for complex panning or special effects.

When involved panning is required, a joy-stick arrangement seems to be a practical solution. The physical position of the joy-stick indicates exactly where you are, and it is possible to pan in any direction quickly and easily. Automated Processes, Inc. has a Quadrasonic Stereo Panner (Model 480) available, using the joy-stick principle. The joy-stick is ideal for sound sources that are to be moved during mastering, but for static placements, a modified version of the common stereo pan-pot may be sufficient. Instead of feeding the two pan-pot outputs to left and right, they are each fed to a series of four pushbuttons, which are in turn fed to the four tracks. It is now possible to pan between any two channels by depressing the appropriate button on each side of the pan pot.

FOUR-CHANNEL ECHO SYSTEMS

With imaginative use of reverb, four-channel sound can really come into its own. Too often, echo chambers (artificial or natural) are regarded as one, input one-output devices. However-most EMT's have two outputs, and if you have a natural (room) echo chamber, you can add additional microphones within the chamber for two (or more perhaps) outputs. In the August 1969 Journal of the Audio Engineering Society, there is an interesting paper by Benjamin B. Bauer: *Some Techniques Toward Better Stereophonic Perspective.* The principles discussed in this paper could easily be extended to four-channel work. Also worth looking up are these A.E.S. Convention preprints, both of which illustrate echo networks which could be easily modified for four channel work.

M. Fouque & E. Redlich *Space Information in Stereo-phonie—Ideas on a New Recording Method for Records*, unnumbered preprint, Oct. 1962

G. Steinke *Special Echo-Mixer for a Sound Recording Control Console*, Oct. 1964 preprint #357

If there is sufficient reader interest, the subject of four-channel echo systems could be treated in detail in a subsequent column.

MICROPHONE PLACEMENT

Even on a rock date, there are a few variations on the usual close-up mic placements that may prove worthwhile. Assuming your studio isn't *completely* dead, try placing two microphones as far as possible from, say, a few electric guitars, while also recording them in the usual manner. The extra microphones are fed to two spare tracks which are later fed to the rear speakers. It may sound great. Then on the other hand, it may be wretched. If it is, erase the tracks and forget where you read this.

On a brass sweetening session, I've gotten some great results (at least, *I* think they were great) by miking the entire brass section onto two tracks, with the mics placed about midway between a typical up-tight pickup and a standard classical distant pickup. Then, additional cardioid mics have been set up in the same general working area, but facing *away* from the instruments. These mics may be fed through a high-speed tape delay and used for rear-channel information. Of course, this type of pickup requires a studio with good-to-excellent natural acoustics. Failing that, it's doubtful whether these mics—fed to an echo chamber—would be as effective as using an echo feed from the regular microphones, to create rear-channel information.

There's still a lot of trial and error experimentation to live through until the recording industry becomes completely familiar with the possibilities of four-channel sound. However, you can rest assured that in the not too distant future, soon after four channel has become something of a standard, someone will get up and say, "you know, I've been working on this new system, and it really sounds fantastic. Of course, you need 8 speakers, but. . ."

Sound with Images

MARTIN DICKSTEIN

PROJECTION SYSTEM EVALUATION GUIDE

For the simplest type of projection "system" containing a 16-mm film projector and a 35-mm slide projector, the customer might just buy the two projectors himself and assume that they will do the job required. Many times he is correct. In a small meeting room, the projectors are set on a rolling table or on the conference-room table and the material is projected on a screen or on the wall.

However, when he is wrong, or when the system is more complex and he does not want to take a chance on buying the wrong equipment, he will ask for assistance. Should the system be very involved, he will probably call on the services of a professional consultant. When he needs help (and calls on the audio man who designed and helped install the sound and intercom systems) there are a few quick *guides* (note: these are guides only) to assist in the evaluation of the existing or future projection system to determine if acceptable image brightness levels will be achieved.

First, a list of the variable factors affecting image brightness, with a notation of the charts on which to refer.

SOME VARIABLE FACTORS DETERMINING IMAGE BRIGHTNESS

Factor	For Slides Fig.	For 16mm Film Fig.
Area of Image (w x h)	1	7
Glass and Mirror	2	2
Front Proj. Screen Material	3	3
Rear Screen Material	4	4
Voltage & Lamp	5	10
Projector Lens	6	8
Shutter	—	9

For slides, the aim is to reach the image/ambient ratio determined by the material to be presented in the slides with the minimum ratio being 5.

For films, recommendation is for an image brightness of 16 ftL., plus or minus 2 (if the presentation is in a viewing room), and not to exceed 18ftL.

The following charts are guides only as the factors are approximate and slightly rounded off to aid in computations. However, the numerals are close enough to permit a good judgement on projection system capability.

Figure 1. Screen brightness (in foot-lamberts) for slides. All brightness values are average and may vary for different projectors, different manufacturers, etc.

Image Area (Sq. ft.) (w' x h')	Typical Non-Arc Projector	Typical Arc Projector
6 (3 x 2)	105	335
24 (6 x 4)	25	85
54 (9 x 6)	12	40
96 (12 x 8)	6.5	21.5
150 (15 x 10)	4	14
216 (18 x 12)	3	9.5

Type	Factor
Glass-mounted slides	0.85
Single pane of glass (2 surfaces)	0.9
Mirror (front or rear alum. surf.)	0.9

Figure 2. Glass and mirrors. All glass and mirrors are assumed clean of marks and dirt, and without much dust.

Screen Material	Factor
Matte	0.85
Ideal Matte	1.00
Non-metallic Lenticular	1.50
Metallic Lenticular	2.00
Beaded	2.80
Ektalite (Kodak)	12.00

Figure 3. Front projection screen material.

Screen Material	Factor
Highly Diffusing	0.75
Average	2.25
Highly Directional	4.00

Figure 4. Rear projection screen material.

Applied Lamp Socket Voltage	Selector Switch Setting (On those projectors having one)	Factor DEK Lamp (500 w.)	DAH Lamp (500 w.)	DEL Lamp (500 w.)	DEA Lamp (300 w.)
110	Hi	0.75	0.45	0.90	0.45
	Lo	0.50	0.30	0.60	0.35
115	Hi	0.85	0.50	1.00	0.50
	Lo	0.60	0.35	0.70	0.40
120	Hi	1.00*	0.60	1.20	0.60
	Lo	0.70	0.40	0.80	0.50
125	Hi	1.15	0.70	1.40	0.70
	Lo	0.80	0.50	1.00	0.55

Figure 5. Line voltages and lamp characteristics, for slides. *Lamp, setting and voltage used as basis for determining figures in Figure 1.

Projector Lens	Factor Typical Non-Arc Projector	Typical Arc Projector
2½" f/3.5	0.60	0.45
3" f/3.5	0.75	0.55
4" f/3.5	1.00	0.85
4" f/2.8	1.15	0.95
5" f/3.5	1.00*	0.90
5" f/2.8	1.25	1.00
7" f/3.5	1.05	1.00
4-6" Zoom f/3.5	0.90	0.85

Figure 6. Projection Lenses for slides *Used to determine Figure 1.

Image Area Sq. Ft.	(w x h) ft.	Typical Non-Arc Projector	Typical Arc Projector
12	(4 x 3)	35	140
48	(8 x 6)	8.5	35
108	(12 x 9)	4	15
192	(16 x 12)	2	8.5

Figure 7. Image brightness (in foot-lamberts) for 16mm. All brightness values are average and may vary for different projectors, different manufacturers, etc.

Projection Lenses		Factor Typical Non-Arc Projector	Typical Arc Projector
1½-inch	f/2.0	0.5	0.7
2 -inch	f/1.6	1.00*	1.00*
3 -inch	f/2.0	0.8	0.7
4 -inch	f/2.5	0.5	0.5

Figure 8. Projection lenses for 16mm. Used to determine Figure 7.

Throughout the slide charts, the assumption is made that the slides to be projected are of the 35-mm double-frame type. Other formats or copies of the original slides introduce further factors to be taken into consideration. Also, other formats usually require changes in image area, projection distance and lens, etc. Note also, values given for screens are for average values within the angle of viewing according to the manufacturers.

One example of how the factors might be used:

Slide image size required: 12' x 8' (96 sq. ft.)—(factor of 6.5 for non-arc and 21.5 for arc projectors.)
Non-image light on screen—25 ftL.
Screen material (front proj.)—matte. (Factor of 0.85)
Desired screen brightness ratio (if slide is in 100:1 category) is 25 ftL. (0.25 x 100)
Lens being considered—5" f/2.8. (Factor of 1.25 for non-arc and 1.00 or arc projectors.)
Single pane of glass in port window—factor of 0.9.
Slides—mostly cardboard mounted—factor of 1.0.

Question: Arc or non-arc projector?
Answer: Start with arc projector and determine brightness: 21.5 x 1.0 x 0.9 x0.85 equals 16.45 ftL. (below the desired brightness value.)
Suggestion: Change screen to non-metallic lenticular for factor of 1.5 instead of 0.85. Final value now: 29.0 ftL.

This computation shows that the desired value can be reached as shown. The non-arc projector would not have proven satisfactory for brightness, with the conditions originally outlined.

As this level is well above the required brightness, the lens can be changed to a 5-in. f/3.5 (factor-0.90) for a brightness of 26 ftL., still above the desired value. If desired, glass-mounted slides could be used (with the f/2.8 lens) for a brightness of 24.7 ftL (just under the desired value) and worth a try. (Factor for glass-mounted slides is 0.85.)

Similar computations can be made for film systems, but it must be emphasized that the resulting value is to be used as a guide only. ■

Shutter	Factor
2-Blade	1.00*
3-Blade	0.70

Figure 9. Type of shutter for 16 mm. *Used to derive Figure 7.

Applied Lamp Socket Voltage	DDB Lamp (750 w.)	Factor DFD Lamp (1000 w.)	DHT Lamp (1200 w.)
110	0.55	0.75	0.90
115	0.65	0.85	1.00
120	0.75	1.00*	1.20
125	0.95	1.15	1.40

Figure 10. Line Voltage and Lamps for 16mm. *Used to derive Figure 7.

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People, Places, Happenings

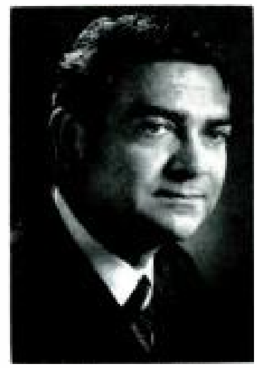
• The third **Conference on Magnetic Recording** as sponsored by the **Hungarian Optical, Acoustical, and Filmtechnical Society** will be held September 8-12 in Budapest, Hungary. The Conference will cover the theoretical and practical problems of magnetic recording on moving magnetic media (of all kinds). The preferred language of the Conference is English, but papers in Russian, German, and Hungarian will also be given. Abstracts of the papers will be published in all these languages. Registration fees for non members of HOAFS are $20 ($10 for members). Contact the HOAFS in **Budapest, V., Szabadsag ter 17. Hungary** for details.

• In a recent announcement **Ronald L. Braho** has been appointed marketing manager of professional sound for the communications systems division of **DuKane Corporation.** His responsibilities will include market research, competitive analysis, and support of sales in both direct sales and training. He comes to DuKane from a sound engineering and contracting firm located in Las Vegas. He will be headquartered at the company's general offices in St. Charles, Illinois.

• **Advent Corporation** becomes the first to announce the marketing of a high-performance cassette tape utilizing **DuPont's Crolyn** formulation. Although the tape will be first made in the cassette format, it is expected to appear in reels in other widths. Crolyn formulation uses chromium dioxide as the magnetic medium.

• Construction is in the early stages on a new building for **Sound 80 Recording Studios** of Minneapolis, Minn. The building will house both the recording studios systems division and creative services division of the company. The company is now in leased space. The new facility will include three sound studios, a mixdown studio, an electronic music studio, tape duplicating facilities, and a construction center. There will be 16-track recording capability in the main studio, with 8-track equipment in a smaller studio. The third will be primarily for production recording with announcers or vocalists.

• The photo shows **JBL president William H. Thomas** receiving a certificate of commendation from the **Mayor of Los Angeles, Sam Yorty** (on the right). The commendation was given for JBL's participation in the Los Angeles Municipal Arts Department **Junior Arts Center's Sound Tunnel** project at Barnsdall Park. In presenting the award, the mayor noted that JBL's contribution of over 200 loudspeakers had made the project possible. The sound tunnel is a 40-foot long anechoic environment in which the speakers have been implanted in the floors, walls, and ceiling. Program material is fed to the speakers by 100 amplifiers and a bank of digitally-controlled computors which can feed up to 30 simultaneous signals through the tunnel.



• **Ansel Kleiman** has been promoted to group vice-president of the **Telex Corporation.** This is a move from his position as a vice-president of the parent Telex Corporation and president of the Telex Communications Division. In this corporate consolidating position, he will assume the division presidency of the Telex Home Entertainment group, in addition to his responsibilities over the Communications group. Mr. Kleiman has been a v.-p. with Te ex since 1965 and was earlier in managerial positions with other companies.

• **Sony-Superscope** president **Joseph Tushinsky** has announced the initiation of a new program for the marketing and distribution of professional audio products. The new Superscope special applications products division will be managed by **Richard Fowle.** The initial products to be merchandised will come from both the **Sony** and **Marantz** lines and have been specifically designed for this division. A full line including microphones, amplification and mixing equipment as well as a portable battery operated two-track recorder for in-the-field mastering are initially offered.

• **Jack Ames** has been appointed as director of marketing for **Otari of America, Ltd.,** in an announcement by **Jack A. Sain,** company president. Prior to joining Otari, Mr. Ames was director of marketing for **Telepro Industries,** manufacturers of cartridge and cassettes. He was also a co-founder of **Liberty Records** in 1955 and was their general manager and executive v.p.

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