db

THE SOUND ENGINEERING MAGAZINE

DECEMBER 1968 75c

Calibrated Monitoring Systems

db Visits Dubbings Electronics

Picture Gallery - - AES Convention

[redacted]


AMPEX

Coming Next Month

• Calibrated Monitoring Systems, part 2 will conclude Don Davis' article on the application of controlled equalization compensation to studio monitors. Next month he will consider loudspeaker efficiency by frequency as it applies to the methods used to "tune" a playback system. The total procedure will be summarized.

Edward Tatnall Canby writes under the title Better T.V. Audio. In a most provocative way he will answer two questions: should t.v. audio be improved and should we adapt multiplex stereo to t.v. The answers are not as obvious as you might think!

And there will be our regular columnists, George Alexandrovich, Norman H. Crowhurst, Martin Dickstein, and John A. McCulloch.

Coming next month in **db**, The Sound Engineering Magazine.

About the Cover

• The console on our cover may well be one of the largest ever. It was shown at the recent New York AES Convention. It's a product of Wiegand Audio Laboratories for MiGinley Studios of Short Hills, N.J. It is somewhat over eight-feet long, has eighteen input positions and twelve output tracks. There is a separate sixteen-track remix section that goes down to four tracks out. This permits the producer to preview the final program. The two sections could be used independently by different simultaneous operators. An article on this console is in preparation and will appear in the near future.

← Circle 75 on Reader Service Card

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THE SOUND ENGINEERING MAGAZINE

DECEMBER 1968 • Volume 2, Number 11

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EDITORIAL BOARD OF REVIEW

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Letters

Correspondence seems to be continuing unabated on the subject of nomenclature standards.

The Editor:
I just finished reading Mr. Tucker's letter in the September issue. I have assumed that the term *hertz* would slowly be adopted by the broadcast and recording industry with little difficulty. It would seem, however, that this will not take place without a great hassle. I take this opportunity to voice my opinion.

We are engaged in a technical field. Most of us consider ourselves either by background or title to be engineers. If we are to be professional in our field it is mandatory that we be able to communicate in precise terms, both quantitatively and qualitatively. All standards in nomenclature are established as a means to allow nonambiguous communication between members of a profession.

It seems to me that our choice is simply to either act as professionals and welcome precision in our terminology, or to reject professionalism entirely, laugh at precision, and prevent our terminology from keeping pace with our technology. We should be happy that terms have been standardized to the point that they have meaning for all.

James Gundlach
Chief Engineer, Television
State University College
Oneonta, N. Y.

The Editor:
One can't argue with the sentiment that I read between the lines of Mr. Tucker's letter in the September issue. He is saying: "I know what the term *cycles* means, and I use it all the time and I like it; please don't change it."

But when he says "A cycle is a cycle . . .a hertz is an asininity," one might just as well argue that the *ohm* is also meaningless and redundant because it is merely a *volt-per-ampere*; and the *siemens* (excuse me, *mho*) is merely an *ampere-per-volt*.

Or let's go the whole hog. The basic electrical units of the International System of Units (ISO) are the *meter* (length), the *kilogram* (mass), the *second* (time), and the *ampere* (electrical current). All of the other named electrical units are meaningless and redundant. Thus, we have absolutely no need for the arbitrary name *volt*—it is much more meaningful to call it by its real name: *the meter-squared-kilogram-per-ampere-second-cubed.* Now isn't that handy! "The powerline potential difference is 120-meter-squared-kilograms-per-ampere-second-cubed." Now I really know what it is. None of those silly *volts*!

The same argument, and the same mess of units, follows for *watts, farads, henries*, etc. So be careful what you say: the use of people-names is purely a convenience for those who use them. The major justification for giving the unit for frequency the name *hertz* is that many people have found it more convenient than *cycles-per-second* and more precise than *cycles.* (We'll ignore here the point that a cycle is not really a frequency unit, because that was discussed in the June issue's LETTERS.

John G. McKnight
Staff Engineer
Consumer & Educational Products Group
Ampex Corporation
Los Gatos, Calif.

The Editor:
I've been following the letter-column comments on poor t.v. audio during the past months. One point seems overlooked.

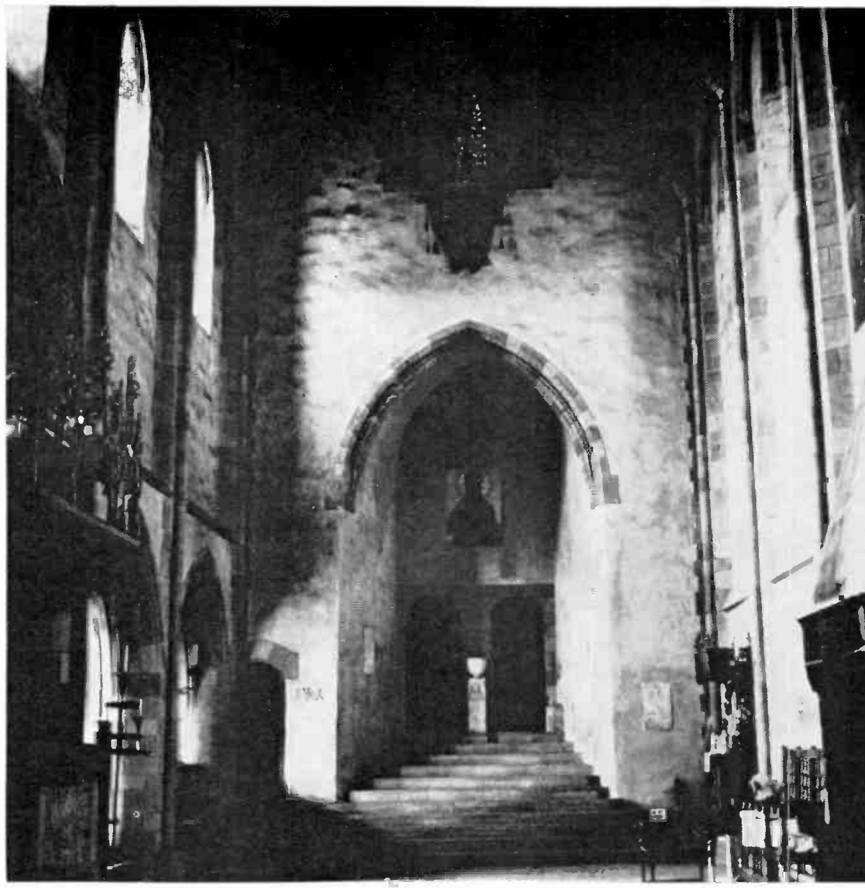
Almost all t.v. networking is done via AT&T long lines using wire pairs (and repeater amplifiers) equalized for flat frequency response only to about 5000 Hz. AT&T simply doesn't have very many inter-city lines equalized beyond 5000 Hz. And they rarely get an order for, say, a 15,000 Hz line.

Thus, you could build a t.v. receiver with a decent audio system and a nice speaker in a good enclosure. You'd have to charge more for the set—a point your competitor would make a lot of. But you would have a clean receiver with 15,000-Hz response.

Since the t.v. station is required by the FCC to be flat (in audio) out to 15,000 Hz, you would get clean audio from them—on local, live shows. On network shows you would still get 5000-Hz audio. On film shows—especially old movies—you would get about the same. How many local live shows are there on t.v.? I think this is the real hangup on the bad t.v. audio.

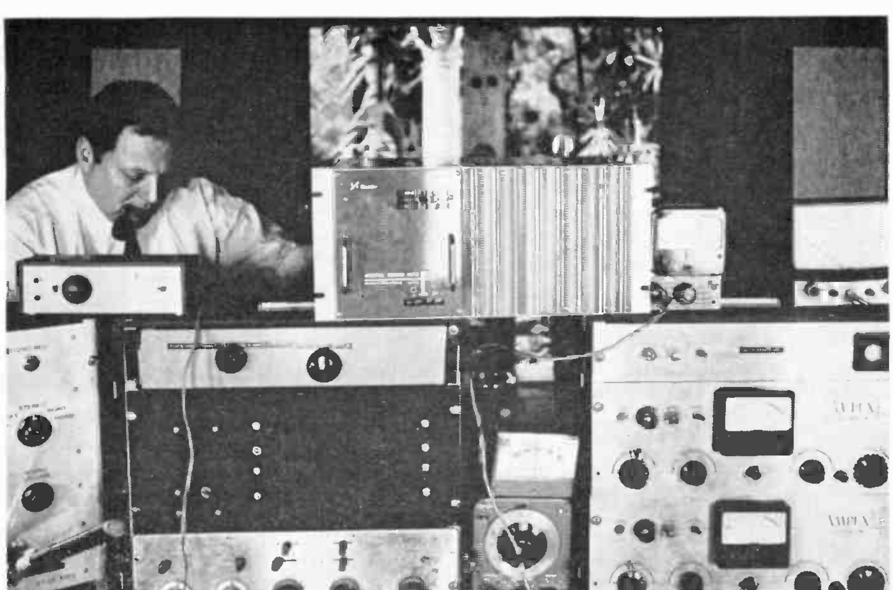
Thomas R. Haskett
New York, N. Y.

United Recording Electronics Industries of Hollywood, Calif. (Universal Audio, Teletronix, Waveforms products) issues a



The great hall of the Hammond Museum. This room is the location of the organ played by Richard Elsasser on Nonesuch H-71200 ("Yankee Organ Music") and H-71210 (Organ Symphony No 5 by Charles-Marie Widor)

AR-3a speaker systems were designed for home music reproduction. Nonesuch Records uses them as monitors at recording sessions.



Nonesuch Records recently recorded several volumes of organ music played by Richard Elsasser at the historic Hammond Museum near Gloucester, Massachusetts. To make the recording, Marc Aubort of Elite Recordings, engineering and musical supervisor, used Schoeps microphones, and Ampex 351 recorder, Dolby A301 Audio Noise Reduction apparatus, and several pieces of equipment which were custom made. To monitor the input signal and to play back the master tape, Aubort used an AR amplifier and 2 AR-3a speaker systems.

The AR-3a speaker system is priced from $225 to $250, depending on finish.

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Overseas Inquiries: Write to **AR International** at above address

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periodic newsletter of considerable merit. The following is an excerpt from issue no. 1, dated October 1968:

Have you ever connected a really good amplifier and speaker to the discriminator of your t.v. set? If you did, chances are that you immediately disconnected it again—for the receiver manufacturers know what they are doing when they put a four- or five-inch open-back speaker system in those half-G color sets. The transmitted sound (most of the time) is of unbearably poor quality.

If you try to extend the frequency range that you hear, all you get all over the low end is a peculiar conglomeration of thin bass notes and ponderous elephant stampedes. The high end carries the distortion.

T.v. sound *can* be good. The prerecorded sequences of some of the top shows come out as clean as a politician's list of kept promises. But the average is sad indeed.

In the final analysis, the blame must be placed squarely on the public. Why should the networks and broadcasters improve things if there is no griping? But the fact that there is a large listening public for standard f.m. broadcasts shows that a certain percentage of it does appreciate higher quality. It is just that everyone is so conditioned to bad t.v. sound that nobody complains.

The transmissions parameters allowed to television are much the same as for f.m. radio. The only notable point of difference is the restriction of t.v.'s f.m. modulation to ±25 kHz instead of ±75 kHz, which degrades the signal-to-noise characteristics a bit. The sound should be uniformly excellent.

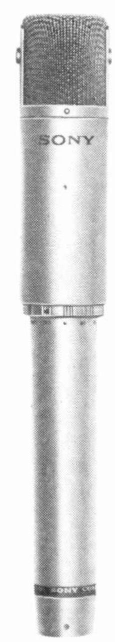
Many European t.v.-casters subjugate the picture in favor of the sound when presenting programs that feature music. Especially serious music (which, admittedly, is not telecast often here in the U.S.). They seem to have the odd idea that hearing the music produced by a performer is of more interest than camera angles that show you the hairs in his nose.

In all fairness, the sound on the commercials is pretty good.

Such a case in point was the recent CBS telecasting of a Vladimir Horowitz piano concert. Picture quality was superb—color was beautiful. Sound was awful.

Columbia Records has released a stunning-sounding disc of the concert's soundtrack. If they could get good sound, why couldn't those of us with good t.v. sound systems, have had it too? This is but one example—we do not mean to single out CBS, the problems transcend network affiliations. And the local stations, in New York at least, seem no better. Ed.

On the beam at the flick of a capsule.



1. Adjustable capsule can be positioned vertically, horizontally, or anywhere between, making the C55-FET unusually convenient for hand-held, boom, stand and suspended operation.

2. Through the use of a field-effect transistor, and the elimination of the vacuum tube, we have achieved rugged dependability and discarded bulky external cables and power supply.

3. Completely self-contained, the C55-FET has an internal 9-volt battery. We think it's a big improvement over the bulky power supply you used to have to hook up.

4. Permits use of standard Cannon XLR-3-12C output connector for battery operation. Now a condenser mike can have the same connector as any dynamic or ribbon mike, providing complete compatibility among all your studio equipment.

5. Exceptional sensitivity and superb cardioid performance characteristics.

6. By the use of a field-effect transistor, Sony eliminates overload problems commonly associated with tube-type mikes caused by grid blocking.

7. The C55-FET's low current drain permits at least 800 hours of battery life. A pilot light indicates battery condition at the flick of a switch.

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These are just a few of the reasons that Sony professional microphones are becoming increasingly popular with knowledgeable studio engineers and sound experts. For more reasons or more C55-FET information, please write to Harold Watson, Sony/Superscope, 8150 Vineland Avenue, Sun Valley, California 91352.

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The Audio Engineer's Handbook

GEORGE ALEXANDROVICH

• Last month we reviewed a few methods of switching audio using solid-state devices and light-sensitive cells. Although our attention has been focused on the switching *element*, we should not forget that actual switching must be initiated by an ordinary mechanical device.

As isolation of the action and faults of these conventional switches, buffer circuits are employed between the actuator and the switching elements. In transistor switches the transistor is isolated from switching transients through the use of r-c decoupling networks. These serve to smooth out the sharp turn-on and turn-off transients caused by the mechanical switch.

One of the particular advantages of light-sensitive cells is that this sort of smoothing action is an integral part of the cell and the light source driving it. Each time current is sent through the bulb there is a small delay before the bulb starts emitting light. There is further delay in the cell until sufficient photons have been set in motion to lower the resistance of the cell. If any transient occurs when the current is turned on (lasting for microseconds) it is filtered out by the combined delay of the cell and bulb (several milliseconds).

Another advantage of the ldr is that it is continuously variable in its resistance (thus introducing no distortion). This is unlike conventional transistors and field-effect devices whose resistance can only be varied over limited range.

This property of the ldr has already been demonstrated in automatic switches for street lighting, in photographic automatic-exposure circuits and light meters, in computors, as flame detectors, color differentiators, etc.

The ldr cell made its debut in the early fifties. One of the first applications was as an automatic brightness control for t.v. sets. First attempts to use ldr's in audio started in the early sixties. The limited practical experience with these devices took its toll. The reliability expected from the cell and the light source was not realized. Along with defects of the cells (poor bonding agents for fastening the contacts to the body of the wafer) and the inadequate reliability of bulbs (long filaments not suited for low-level illumination) there were equipment failures galore. The results undermined the faith in circuits using these cells.

After several years of painful advancement—relying mostly on the respective manufacturers of bulbs and cells to come up with something better—product reliability came to be something to be expected from such circuits. Compressors, limiters, attenuators, and controls have been designed and marketed. The application of cadmium-sulfide cells in these circuits is the subject of this month's column.

Any circuit designed with such a cell should comply with the basic requirements for the operation of the cell. I have outlined these requirements before— low impedance with sufficient illumination for the attenuator circuits, and fast light sources for the compressor and limiter devices. When these

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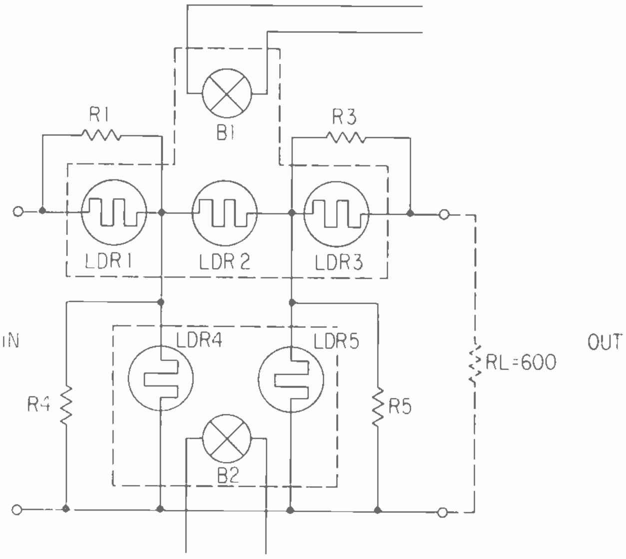

Figure 1. A constant-impedance attenuator. R_1 and R_3.. 560Ω; R_4 and R_5. 4700Ω.

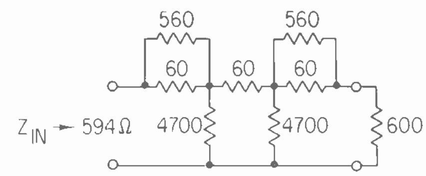

Figure 2. The equivalent circuit of an attenuator without added attenuation. Insertion loss is 3 dB.

ground rules are complied with, you can expect the desired results.

It's easy to develop highly complex circuits when designing with ldr's; they readily lend themselves to over design. Such circuits have already appeared: they make use of transistor amplifiers, driver and ldr cells with several light sources and two transformers—all in a simple attenuator circuit. The other side of the coin has attempts to take pennies out of a design by trying to have one cell and one light do what only tandem units can do. Neither approach can be considered the correct one.

I believe that a circuit should only do what it is expected to do. If an attenuator with constant impedance for 600 ohms, minimal insertion loss, and capability for infinite attenuation is needed, the simplest approach should be used that does the job.

If an attenuator is required to trim the gain of an amplifier for remote-control purposes, a single ldr can do the job in the feedback loop of the amplifier. In this case, impedance, response time, and applied level are not factors. Quite the contrary, the use of a cell in the feedback assures us of wide enough a control range along with minimal distortion (identical to a non-ldr circuit) and small power requirements for the light source.

Figure 1 shows a constant-impedance attenuator: The attenuator consists of five ldr's driven with two light sources located in two light-tight compartments. When light B_1 is on, B_2 is off. Fully illuminated ldr's 1,2, and 3 are in the direct signal path offering resistance of several hundred ohms between the input and output terminals. The resistors R_4,R_5, and R_L are shunted across making the circuit look like 600 ohms for the source. Figure 2 shows the resistances of the circuit when bulb B_1 is on; Figure 3, when B_2 is on. Note that the impedance remains close to 600 ohms. Since the center ldr 2 is in total darkness, its resistance rises to a minimum of 10 megohms. A signal entering the attenuator is reduced 20 dB by the pad R_1 and R_4. Further signal reduction is in the pad ldr 2 and R_5. The voltage ratio of this pad is approximately 200,000/1 giving us a signal reduction of 106 dB. The final pad is formed by R_3 and the load resistance, giving a 6 dB loss. Totalling all of the losses gives an attenuation of 132 dB. So for all practical purposes signal fed into the attenuator at 0 dBm with a system noise of the amplifying chain at 70 dB below signal, will have attenuation at 52 dB below the noise level.

At any point of illumination, *off*, *partly on*, or *full on* the input and output impedance is kept constant, but the attenuation changes from a nominal insertion loss of 3 dB to any value down to −132 dB. By using transistors to control current to the lights with the single potentiometer, illumination can be precisely adjusted for the desired value of attenuation by using a non-linear element.

As is seen from this example, light cells can do the job that once could only be done by switches and resistors.

Attenuators using ldr devices have two outstanding advantages. First, there is the ability for the remote control of gain without moving parts. Second, there is the ability to control a multitude of channels simultaneously.

With different ldr's located around a common light source, attenuation for several channels can be adjusted to track within 0.5 dB over the entire attenuation range. In this era of multitrack recording, this ability for fading all channels at once is important.

Compression circuits can benefit from ldr's. Several units now on the market employ ldr units as a compression element. Attack time can be speeded with the use of fast (thin element) bulbs or fluorescent light sources—with the cell operated in a high-impedance circuit.

Attack times of some compressors are quoted at speeds of as low as 100 microseconds. But what is truly important for compressors (not limiters) is the over-all gain control or average audio level. A compressor is normally designed to compress the dynamic range—not protect the circuit from excessively high transient peaks (by clipping them). A compressor does offer definite overload protection. It does this in the form of general gain reduction rather than with specific limiting of transient peaks. It has been noted by several audio experts that excessively short attack times can be as quality damaging as excessively long times. If the time is incorrect, the compressor will not average the levels but will either sample peaks and change the loudness of the program (if speed is too fast) or only react too late on sustained notes (if too slow).

Compression amplifiers can be made frequency selective (as can the ldr circuits). Thus, high-, low-, and presence-compressor circuits can be constructed. The same techniques lend themselves to expanders, automatic loudness controls, and equalizers. There is no end to the number of applications for light-dependent elements.

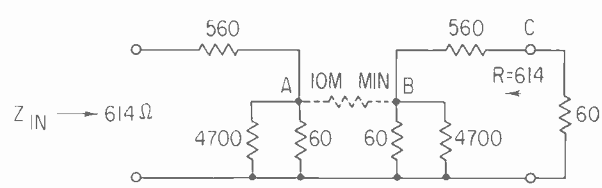

Figure 3. With maximum attenuation the equivilant circuit looks like this. Signal attenuation will be infinite.

Meet the mixer with the mostest–

priced from only $300 with 6 inputs and 3 controllable outputs—that's E-600 by Electrodyne.

It's ready to go in any application from high-quality recording through CATV to sound reinforcement. Rack-mounted or completely self-contained in its own handsome Teak cabinet. E-600 comes with 6 mike or line inputs, each with speech/music select switch. All three outputs—Line, Headphone, and Auxiliary—have their own independent level controls.

Thanks to integrated-circuit, plug-in electronics and Electrodyne design, frequency response is 20 Hz to 20 kHz ±1 db. Maximum output is +30 dbm; distortion less than 0.5%.

There's an illuminated VU meter with multiplier, along with a built-in oscillator for level set. The fully-regulated power supply can be paralleled for fail-safe battery operation, and the headphone jack can be paralleled for an additional output.

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The Feedback Loop

JOHN A. McCULLOCH

• In the June issue I requested your assistance in determining the present usage of the XLR type of connector, and the general practice in its wiring. At this time I have had about thirty replies to the request, and the breakdown of the replies confirms my own investigation into the matter.

About forty per cent of the response utilized pin *two* as the hot lead point. Thus sixty per cent seemed to favor pin *three* as hot. But. . . if we should breakdown the number of microphones involved and devise a percentage on that basis, then fully seventy-five percent were wired with pin *two* hot.

Let me pause for a moment and state what is meant by my use of the designation *hot*. Positive pressure on the diaphragm will produce a positive voltage at the pin termed hot. Some reaction has been along the vein that, since the microphone is usually a balanced line the phasing does not matter. But, should the phase angle between two microphones vary (due to acoustical or electrical causes) and the microphone pickup be combined, then degredation of the information picked up by both microphones (leakage) will result. Sometimes electrically phased microphones can help, but more often, unknown, they can and do harm the quality of the pickup.

It is interesting to note that most of the manufacturers of microphones adhere to the use of pin *two* as hot. It seems that the recorder and console equipment manufacturers prefer pin *three*.

What to do? If we have to decide upon an absolute standard some facts must be considered.

First, more and more of the modern highly-sophisticated microphones are being sealed for acoustic reasons. Others, due to assembly practice, cannot have their connectors removed without great danger of breaking the leads.

Second, since the final output of the microphone is usually balanced it would not matter in a single system which side was finally referred to ground.

Third, microphones are usually the only objects concerned in which different manufacturers' products are used together simultaneously, and mixed into a common signal path. (Recorders are usually concerned only with their particular tape/track which will be replayed upon a device whose outputs are similarly phased. Mixers are generally thoroughly checked for correct phase from input to output, and if used in combination are generally the same make.)

Therefore, which of the methods of wiring the XLR (should a standard be issued) require the least changeover, or none at all? I feel the pin *two* hot should be preferred. Present recorders, consoles, or other equipment accepting a balanced-line input, would not have to be modified to continue working. Your method of color-coding cables could remain the same, but you would be assured that all microphones would arrive in-phase and ready for work.

Up to this point an argument can be stated this way for either pin. The final argument for pin *two* hot is that the great majority of microphones presently used are shipped from the factory wired pin *two* hot.

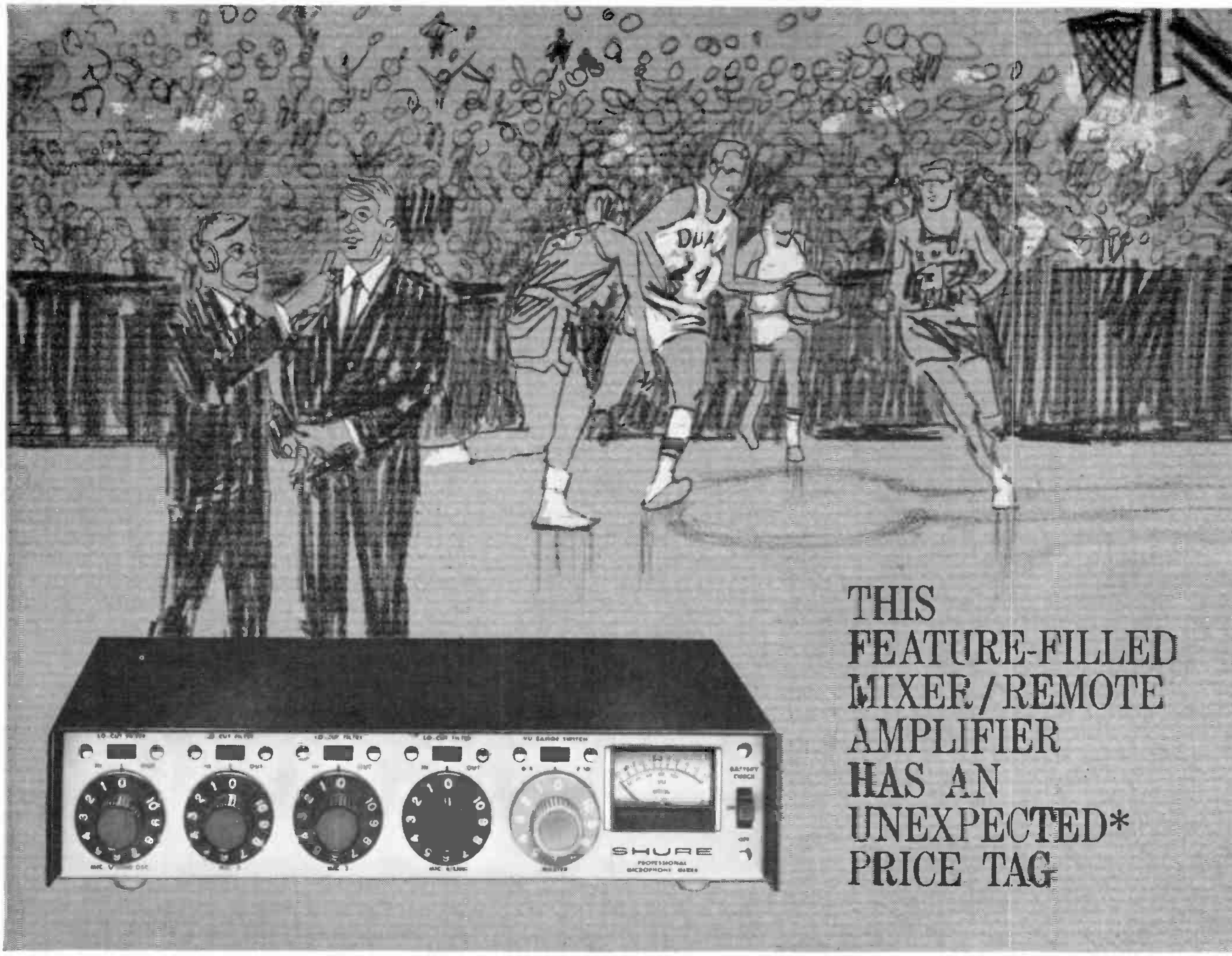
It is also my contention that all equipment should carry a designation somewhere in its technical descriptive literature similar to the following:

Positive (pressure/voltage) on the diaphragm/input pin #—) will produce a positive (voltage/pressure) at the (output pin #— cone).

Then we all could be positive in our phasing connections without resorting to time-consuming tests.

To those of you who wrote in your preferences, many thanks. To others, if you disagree with strong reason to my conclusion, let me hear from you. And so, until a standard is issued from the EIA, I am going to recommend the following wiring procedure for the XLR type connector.

Pin one — ground . . pin two — hot signal . . pin three — low signal.



*Unexpected? Judge for yourself: the Shure M67 Professional Mixer/Remote Amplifier has virtually *every* important feature asked for by broadcasting and recording studios and sound installers. It can be used as a complete, compact console for studio, remote, or original installation use—or as an "add-on" mixer for expanding existing facilities and providing additional microphone inputs with tape recorders and VTR's.

It features 600 ohm line and microphone level outputs; an illuminated VU meter calibrated for +4 and +10 dbm; four low impedance microphone inputs, each with individual low-frequency filter inputs—plus one input convertible for line bridging, or 600 ohms; built-in tone oscillator to provide calibration signal to set signal levels; extremely low noise and RF susceptibility (noise level −126dbv); female Cannon connectors, two-level headphone monitor jack and many other features.

Its distortion level is UNDER 1% from 20 to 20k Hz at +10 dbm out! Clipping level is +18 dbm. It also features a unique *noiseless* automatic switchover to battery if the AC line fails. (Battery pack optional.)

It is not much larger than two cartons of cigarettes (11⅜" x 7½" x 2½") and weighs just 4 lbs. 4 oz.

Incredibly, this truly professional, compact, complete mixer/remote amplifier has a professional net price of only $147.00!

Write Professional Products, Shure Brothers, Inc.
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SHURE M67 *new*

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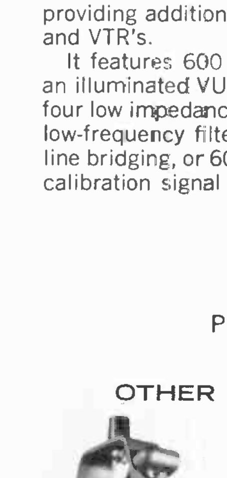
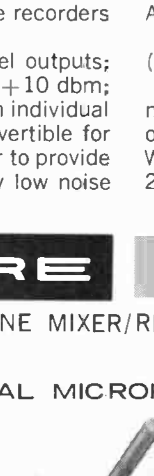
OTHER SHURE PROFESSIONAL MICROPHONES . . . FOR SUPERIOR AUDIO



MODEL SM5
CARDIOID BOOM DYNAMIC

Because its cardioid directional pattern is uniquely uniform with frequency and symmetrical about its axis, the SM5 is singularly independent of the effects of environment. Even in extreme shooting situations (such as with tight sets, low ceilings, hard walls, low microphone angles, traffic or air-conditioner noise and rumble and changing distance) the SM5 minimizes sound coloration and ambient noise pickup.



MODEL SM76
¾" OMNIDIRECTIONAL DYNAMIC

Ideal for interviews and audience participation, yet unusually smooth wide range response (40-20 KC) for critical music reproduction. Instantly detachable from stand. Steel case with Cannon connector.

MODEL SM50
OMNIDIRECTIONAL DYNAMIC

Self-windscreened and pop-free for news, sports, remotes, and interviews. Also ideal for many studio and control room applications. Comfortably balanced for hand or stand use. Natural response.

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Theory and Practice

NORMAN H. CROWHURST

• After building a multivibrator for something else, I had the idea of making one that I could use as an electronic siren, for injecting as an effect on various programs where appropriate. This meant that the frequency must be made variable, as well as the intensity or amplitude.

In theory it's easy. To change the frequency of a multivibrator, you can change either the capacitors, C, or the base feed resistor, R2, (Figure 1). Neither of those can easily be made electronically variable. But the supply voltage to the top end of the base resistors can easily be varied (Figure 2). I set it up with a potentiometer to explore the effect: a circuit to produce the required voltage to simulate a siren could come later.

Calculating the basic frequency of a multivibrator proved quite easy. When a transistor suddenly conducts—and that is sudden, because it's expedited by the combined current gain of both transistors, acting as a two-stage ampilfier—the base of the other one is suddenly pushed to opposite polarity (positive for pnp, negative for npn as shown), by a voltage equal to supply voltage.

Now comes the timing part. That base is cut off until the voltage reaches zero, when the transistor starts to conduct and the other flip is triggered (Figure 3). The voltage across the base resistor starts at twice the supply voltage and drops as the current through it charges the coupling capacitor, the other side of which is held at zero voltage by the other transistor being saturated.

Looking up the value of x that makes $e^{-x} = 0.5$, or $\log_e 0.5 = -x$, from natural log tables I found it is 0.693. Within the tolerance of values I would be using, I could take this as 0.7. So if the base resistors are 10K and the coupling capacitor is 0.1 mFd, the time constant is 1 millisecond, so that 0.7 of it is 0.7 millisecond. This is the half period, so the whole period will be 1.4 milliseconds. The corresponding frequency is 1000/1.4 = 714 Hz.

Making up the circuit, I found this accurate, well within the tolerances of parts used. Collector resistors are 1k, to make sure each transistor saturates, because current gain is far more than 10, the ratio between base and collector resistors.

Now I unhooked the base resistors and provided them with a variable voltage (Figure 2). The first question was how far could this control vary frequency. The theory is still fairly simple, with natural log tables handy. Suppose the voltage to the base resistors is a fraction a of the supply voltage: then the voltage across the base resistor at the beginning of a half period is $(1 + a)$, and at the end of the half period it is a (Figure 4).

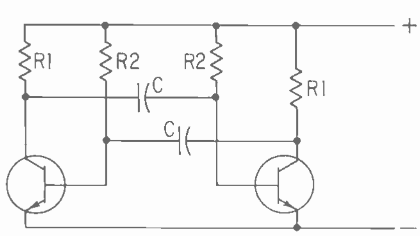

Figure 1. If you want to change the frequency of a multivibrator you can change either the capacitors C, or the base-feed resistors R2. But neither of these are readily made electronically variable.

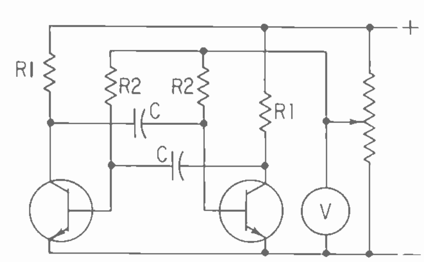

Figure 2. A potentiometer can be used to electronically vary the top-end supply voltage for purposes of exploration.

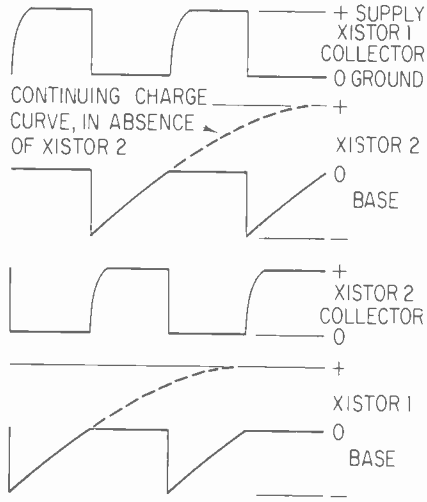

Figure 3. The triggering action in the multivibrator circuit shown in Figure 2.

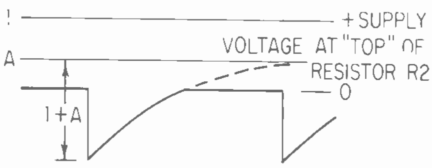

Figure 4. The voltage across the base resistor at the beginning of a half period is (1 + a). At the end of the half period it is a.

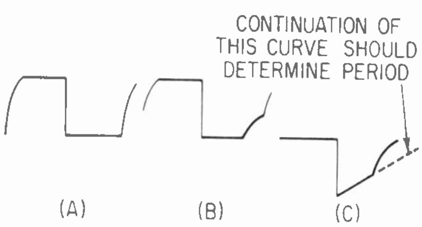

Figure 5. While the frequency went down with reduced base-supply voltage, the collector waveform looked like (A). When the frequency started going up again, it looked like (B). Voltage starts to rise before the end of the conducting half period. (C) This couples through the capacitor, making the other side reach zero (and conduction) sooner than it should.

So the half-period is the value of x that satisfies the expression, $\log_e a/(1 + a) = -x$. Making $a = 1/5$, or 0.2, $a/(1 + a) = 1/6$, and $x = 1.792$. Say 1.8 in round figures. Using the same values, the half period is 1.8 milliseconds, the period 3.6 milliseconds, and the frequency 1000/3.6 = 278 Hz.

Make $a = 1/10$, or 0.1, $a/(1 + a) = 1/11$, or 0.091, and $x = 2.398$. Say 2.4. Now the half period is 2.4 milliseconds, the period 4.8 milliseconds and the frequency 1000/4.8 = 208 Hz.

To check this out, I ran the voltage down to these values. The 1/5th point I just about made, according to prediction, but then, instead of going on down, the frequency started going up again.

What happened? I'd coupled the collector of one transistor, through some high-value resistors, to an amplifier so I could listen to the output. So now I got the 'scope to look at it.

While the frequency kept going down with reduced base-supply voltage, as it should, the collector waveform looked like Figure 5(A). But when frequency started going up again, the collector waveform looked like Figure 5(B). Now I saw what was happening.

The *on* transistor is supposed to be saturated during its part of the period. When it's first switched, it saturates, because of the charge on the capacitor. But the steady current through the bias resistor, with the lower voltage I was now using, is not enough to saturate it.

So the voltage on its collector doesn't stay at zero for the whole conducting half-period. Toward the end, it starts to rise. This couples through the capacitor, so the other side reaches zero and conduction, sooner than it should. Checking the waveform at the base confirmed this as in Figure 5(C).

The only way to make the transistors continue to saturate with lower base-supply voltages is to use a lower value base resistor, or a higher value collector resistor: to make the ratio less than the 10:1 I started with. Right now, I was concerned with seeing the change, so I changed the collector values, leaving the base circuit values the same. I changed the 1k resistors for 2.2k resistors.

I switched on with the base supply voltage all the way up. I had in mind checking that the frequency at this position wasn't changed by changing the collector resistors. But it didn't oscillate. Instinctively, I turned the voltage pot down and it started to oscillate at a lower frequency, and now I could get down to 1/10th of supply voltage, and the frequency was close to the predicted 208 Hz.

I turned the voltage back up to the top, and it didn't quit oscillating. The frequency was a little higher than the 714 it had been with 1k collector resistors. So I switched the supply voltage off and repeated my check. Again, it wouldn't start with the voltage at the top. As my siren circuit wouldn't want to start at the top, but at the bottom, I suppose I needn't have worried any further, but I wanted to know why it wouldn't start at the top.

Out of curiosity, I changed the collector resistors again, to 4.7. Now I could turn down to about 1/25 (0.04) of supply voltage before the oscillator started going up in frequency, and the frequency went down to about 155 Hz, which my calculations predicted. But when I turned it up, the oscillator quit just before it got to full supply voltage, and didn't start again, until I turned the voltage down to about 4/10 of supply.

Examining the waveform, I found that something I noticed with the first waveform I looked at got worse: the rounding or slowing of the front edge of the rise gradually robbed the amplitude (Figure 6) until it quit oscillating, fairly suddenly. I realized that my calculation assumed the voltage on the collector (at the moment each transistor starts to conduct) has reached full supply value. And I also realized that this isn't true when you go too far in the direction I had just gone.

At my starting point, the rise time is 1/10th of the base-circuit time constant, or 100 microseconds. In 700 microseconds (0.7 millisecond), the voltage reached is 0.9991 $(1 - e^{-7})$ of supply, which is close enough not to bother with. But with 4.7k, the time constant is 0.47 milliseconds. So in 0.7 milliseconds the voltage across the collector resistor drops to $e^{-0.7/0.47} = e = ^{-1.49} =$ 0.225 of supply, and the voltage at the collector is only $1 - 0.225 = 0.775$ of supply.

That would not stop oscillation, but it changes frequency, as well as amplitude. The reverse voltage switched onto the base is no longer equal to supply, but goes only to 0.775 of it. So the fraction used to calculate half-period is not ½, but $1/1.775 = 0.562$, which makes the half period calculate to 0.576 millisecond, instead of 0.7. So now the voltage reached will not be 0.775, because the time is shorter.

Going round the calculation again, in 0.576 milliseconds the voltage across the collector resistor drops to $e^{-0.576/0.47} = e^{-1.22} = 0.288$ and the voltage at the collector is only $1 - 0.288 = 0.712$ of supply. So the fraction is now $1/1.712 = 0.586$, which makes the half period 0.534 milliseconds, instead of 0.576, and we go round again.

There may be a shorter mathematical approach than this round-and-round, to determine the ultimate frequency and amplitude, and where this cumulative action builds up to cut off oscillation altogether. We have gone far enough to show that it happens. Now the question comes: what stops the oscillator from *starting up*, until a certain point is reached, although it will continue oscillating above that, once started?

This I realized was something I should have asked myself earlier. I checked the condition when the oscillation wouldn't start, because I suspected (the first time) that I may have blown a transistor. But I hadn't. And both the transistors were conducting at once, to saturation. Now I asked myself the opposite question: how would an oscillation ever get started, under this condition?

The fact is it did, and the answer isn't too difficult to reason out, although

Figure 6. When supply voltage was turned back up the oscillator quit just before it got to full supply voltage and didn't start again until voltage was turned down to about 4/10 of supply. The amplitude of the waveform was gradually being robbed at the front edge.

finding data to determine the exact point may be difficult. When both transistors are really saturated, gain disappears. The circuit only works, or at least it only starts, when the saturation is incomplete enough so that the current gain is more than 1.

In later experimenting, I found that different transistors, with the same current gain, would start at different current ratios between base and collector current. This is a function of how the collector curves converge at saturation. A transistor that saturates relatively suddenly will not start until the ratio of currents is higher, and the frequency range over which such a transistor will be self-starting is thereby reduced.

I wasn't getting much nearer to my siren circuit, although I'd proved I could make a variable frequency oscillator suitable for the purpose this way. But I was finding out some interesting facts about multivibrator circuit design. And I'll tell you more about that in the next issue.

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Sound with Images

MARTIN DICKSTEIN

• One of the important audio/visual art forms is called *Son et Lumiere* in the original French or *Sound and Light*. This method of aural and visual display and the philosophy associated with it began in France and was first demonstrated for the public in 1952. France with its medieval castles was the perfect starting point for this form of exhibition and it became popular very quickly. The idea spread to other locations in France, then to Belgium, England, Portugal, Italy, Greece, Egypt, Lebanon, Switzerland, Sweden, Austria, Holland, and to the western hemisphere in Mexico and the United States.

In the original concept, an historic site was chosen as the star of the show; a script including voices, music, and sound effects was written and recorded on multi-channel tape. Lights of different types, intensities, colors and characteristics were carefully hidden from the audience's view although the lights were located to illuminate the site from different angles and sides. Loudspeakers were judiciously placed to create a total effect of realism in conjunction with the lighting, and a large staff of technicians was hired to manipulate and operate the complex equipment required to put on the performance.

Through the variations of the colors, the changes of brightness, and the suddenness of changes in general illumination, a static structure was brought to life. Through the recorded voices of professional actors and extremely realistic sound effects, with music specially created to enhance the dramatic impact, the audience was taken on a tour of the historical background of the site. Adding to the overall realism, the show is given at night to take advantage of the darkness in which to unfold the drama.

The first programs were written with few lighting changes. These changes were made by technicians operating hand levers as is done in theaters. As the shows were changed to include more and faster lighting variations, the staff of technicians increased while the number of available lighting circuits dwindled. It became obvious that a simple matter of economics would prevent the programs from being made too complex.

As all the original equipment contained wire connections made by soldering, it was a time-consuming project to make any changes in the show. Thus, a program was produced only once for its maximum audience-draw and was played until the box-office take became too low. The show was then changed to re-draw the old audience and to attract new visitors.

Automation took place first in the the audio portion of the program. The introduction of the fifty-step telephone selector switch made switching speakers a snap and introduced an expensive but flexible means of automation. As the connections were still made by soldering, making changes in the show remained difficult.

The next change in the equipment related to automation was the introduction of the punched card. Now, the changes in the show need no longer be made in the wiring connections. When a show needed "doctoring," the change was made in the punched card. At first, the cards were not really punched as we know them today. The cards contained rows of ridges and furrows running the length of the card. The microswitches rode the tops of the ridges. When a switch was to be activated, the ridge was broken at that point so that when the card moved forward sufficiently, the desired switch fell into the depression and its circuit was put into operation. It can still be seen that when a card was punched incorrectly it was doubtful that the error could be corrected without redoing the entire card.

It was the introduction of the punched card with its added flexibility, the development and use of thyratrons and motor-driven potentiometers, and the changeover from metallic contacts to audio-frequency tones on the tape for triggering the card movement, which permitted greatest artistic achievement in the dramatic development of the *Sound and Light* program. (Thyratrons were used in Europe because of the 220-volt source available in almost all locations.)

To the best of our present knowledge, each of the presentations was given outdoors with the exterior of the site being illuminated. This was in accordance with the regulations at most historical sites that none of the equipment be located where it can be seen.

Using an auditorium big as the outdoors really created problems in the dynamic lighting effects that could be devised to light one side of the structure as well as the location for the various fixtures. The lights had to be able to move for a travelling-light effect in relation to the travelling-sound, change colors gently and gradually (or harshly and suddenly) in keeping with the mood of the program material, fade in and out slowly or quickly and go on and off suddenly. Devising the proper effects, getting the colors just right, balancing the various components into a final dramatically-acceptable blend—and then devising methods to hide the fixtures in the desired locations—took a good deal of sweat and tears. The same was true, of course, in the audio effects. The speakers, usually sound columns, had to be located strategically for travelling sound, stereophony, dramatic balance, proper orientation with respect to the public and the apparent location of the sound effect or voice. They had to be hidden from the sight of the public, both during the performance (for dramatic effect) and during the day when the public was allowed to wander through the historic monument or shrine.

The heart of the system was the multi-track tape. Five tracks were used for voices, sound effects and music; three tracks were used for recording control signals for the dimming circuits; one track was used for recording tone signals for the punched-card programmer controlling on/off lights; and the last track was used for the tone signals which stepped the punched cards for switching speaker circuits. The actual sound tracks were recorded in a professional studio on professional equipment. Blending and mixing was then done onto the ten-track tape ready for the presentation.

This tape was then taken to the site where the control signals were recorded for lighting effects. A specially-designed multi-frequency encoder allowed the artistic director to "write" his lighting score while actually watching the lighting effects taking place. A matching decoder and frequency-sensitive circuits sorted the signals for the proper lighting. Those familiar with the technique of recording control tones on a tape while trying to match a previously-recorded audio track are aware of the ticklish timing problems involved.

The first performance of this relatively new audio/visual medium in the western hemisphere took place on July 4, 1962 at Independence Hall in Philadelphia. Others followed at other historic sites thereafter with one in Mexico created at the Pyramids of the Sun and Moon at Teotihuacan near Mexico City.

One of the most recent was opened on July 4, 1966 at an exact replica of the original Independence Hall specially constructed in Southern California for the purpose of presenting a sight-and-sound tour of the early history of our country relating to the signing of the Declaration of Independence.

In this presentation, the audience is seated in the room in which the events took place. The stage is set by tables and chairs, appropriate props, doors, windows, and effects in precise keeping with the original occurances.

A specially-modified fourteen-track tape is utilized. Thirteen of them are used for voice and sound effects with the last one for two-tone control of house lighting and table candles. The precise balance of sound levels from 56 speakers carefully hidden throughout the stage area creates an effective illusion of the return of the original characters of this most important dramatic and historic event. A modern innovation in this presentation is the incorporation of an automatic recycling process for the tape machine to rewind and recue after each performance is completed. By taking place indoors, the show can be repeated regularly throughout the entire day with only enough time required between shows for a change of audience.

Today, the sound-and-light medium of educational entertainment can be expensive, requiring bulky permanent installation. But the development of smaller lamp fixtures and sound equipment, smaller control units for the audio and visual equipment, and innovations in the artistic aspects and requirements might make possible in the near future, a host of road shows which can provide interesting and educational entertainment almost anywhere that a historical site can be found. Perhaps exact replicas of shrines can be made portable to take history from any original location to any place else.

Editorial

THE PROBLEMS OF THE GHETTO have been widely discussed. While there is general agreement that business and industrial fields have a responsibility to help the disadvantaged, there have been few attempts at direct action.

A recent article that crossed our desk discussed an effort in the photographic industry to take ghetto youngsters and train them for jobs as lab technicians. (Industrial Photography, Ghetto Children by Ralph Baum of Modernage Photo Services, N.Y.C.) Apparently several firms, notably Western Electric Company's photographic department, have offered concentrated courses in lab work. They are using Western Electric's beautifully-equipped photo lab in the evenings (when activity is at an ebb) as a classroom. The youngsters (16-18) are receiving thorough training in the use of equipment. Further, an expertly conceived course (taught by WE's personnel) was specifically designed for individuals with little education and poor language skills.

The graduates of this course are finding work. If they persist at their employment they will never again be unemployment or welfare statistics.

At the same time, the New York City Neighborhood Job Corps has undertaken similar courses. They are badly understaffed, under-financed, and under-coordinated but the extreme personal dedication of some instructors has helped to overcome these handicaps.

This all relates, of course, to the professional audio field. We all know of the shortage of skilled technicians in broadcasting and recording. There are few courses training individuals for such jobs; those that exist do not come within reach of the ghetto student.

We need courses geared to bring these students to where we need them. This can best be done through local educational facilities on a high-school level (but *not* as part of the high-school curriculum).

The educational facility usually can find the necessary financing, but it is woefully ill-equipped to *teach* professional audio. That is where the audio pro must come in. Most recording studios and broadcasting stations have some time of the day when activity is low enough to permit an hour or two of use as a classroom. Certainly, they have the personnel that can function as teachers. The audio man, working with a professional educator must define these practical courses.

Perhaps the most important ingredient is the industry's real desire to help. It is not enough to be motivated by fear — fear does not produce meaningful education. Directing the unemployed to where business needs them is a basic premise of American capitalism. It has always proved good for business . . . now it is vital for the welfare of the entire country. Who out there is ready to help . . . *now?*

L.Z.

Calibrated Monitoring Systems

PART ONE

DON DAVIS

It is not necessary to accept the monitor speaker as the weak link of the recording system. The author describes a method that equalizes the acoustic output of the monitors so that they become a calibrated part of the studio.

Don Davis is manager, Acousta-Voicing—, Altec Lansing.

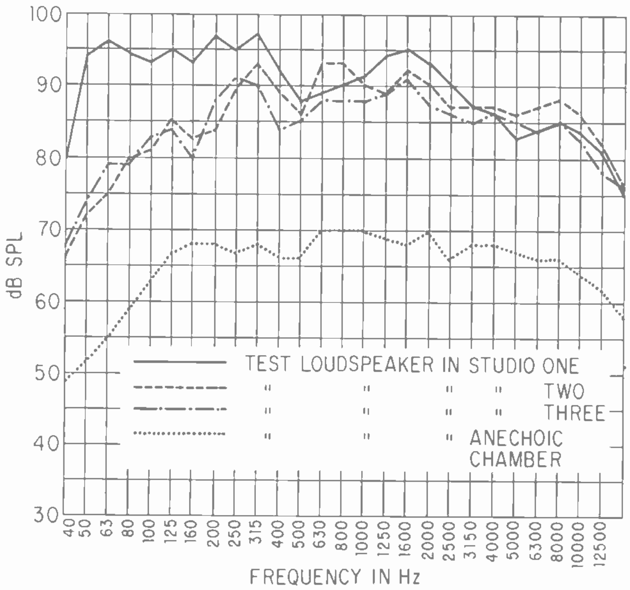


Figure 1. Each of these response curves was taken with the same test loudspeaker, but measured in different rooms. The anechoic chamber measurement was made in the large chamber in the Altec Lansing engineering laboratory.

RECORDING, BROADCAST, AND MOTION-PICTURE sound engineers rely on aural monitoring systems as a guide in mixing, balancing, and dubbing program material. They lavish time and money on achieving remarkably low variations in the electrical frequency response of their apparatus (+0.5 dB or less). Their electrical distortion is usually controlled to less than 0.5 per cent thd and much soul-searching is carried out in the reduction of esoteric amounts of intermodulation distortion, scrape flutter, phase shift, etc. Maximum dynamic range is sought as equivalent input noises are reduced to their theoretical limits, and heat is generated by powerful output amplifiers of 50, 100, 200 and even more watts.

But when they come to that terminating resistor used during the electrical tests, they pack up their instruments and sit down to listen with their ears to a totally uncalibrated and untested loudspeaker-room combination. They are quick to reply that they have isolated the exterior noises that would intrude and have in general followed all the collected folklore of previous studios as well as an acoustical consultant's best advice. In the case of the loudspeaker, they have the manufacturer's sales sheet showing a "typical" curve for the unit *as measured in an anechoic chamber*. All this is fine and as it should be. Care with details helps the whole to perform more efficiently. But, what do you suppose are the *real results that they hear at their ears*? FIGURE 1 shows what can happen to all this planning. FIGURE 2 illustrates test equipment used and its connection to the playback sound system.

Why These Curves Look As They Do

Most of us have had the experience of hearing and seeing a window pane or wall panel vibrate as the result of a loud sound. If attentive, you will observe that the vibration seems to produce a sound of definite pitch. Actually, the window or panel is vibrating at one of its natural frequencies (modes). The closer to this natural frequency the exciting sound source approaches, the greater will be the response of the window or panel.

A room also has its preferred modes. It is easily demonstrated that a source with calibrated uniform amplitude will be heard alternately exciting and failing to excite these normal room modes as the frequency of the uniform amplitude source is changed. In some frequency regions these modes pile up. Sounds having a frequency within a region where there is a concentration of modes are emphasized by the resonant action of the room.

Many an early church father learned to pitch his voice to take advantage of such concentrations to allow himself to be heard in large reverberant spaces. Many chants were originally developed to excite such phenomenon. (Better acoustical control might be one of the reasons assigned to the decline of chanting.)

When a loudspeaker is designed, great care is exerted to minimize the inherent resonances of the driver mechanism. So too, in professional studio design, care is taken to achieve as uniform a distribution of normal room modes as possible. Since there are no perfect speakers and the application of present-day knowledge of room design fails to achieve even distribution of normal room modes at low frequencies in small rooms, the possibility of having natural resonances in the loudspeaker excite natural resonances in the room is an omnipresent probability.

Corrective Measures Taken

Robert Metcalf, chief engineer at Century Recordings, Saugus, Calif., was unhappy with the compromise required when he stepped from the live program in the studio into the monitored program in the control room. He felt even greater dissatisfaction with the distressing differences in the reproduction of identical program material as played back over good monitor loudspeakers in the cutting rooms (FIGURE 3) and the quality-control rooms (FIGURE 4) at Century Recording. It was apparent that they sounded differently in each of the rooms. What disturbed him most was the question, "which room was right and which room is out of balance with the original sound?".

Still another problem at hand was the fact that he had purchased two new professional monitor speakers (FIGURE 5) and they did not sound as balanced in his cutting room as the supposedly inferior older units. In the course of having these units checked at the manufacturer to make sure they did indeed perform as specified, Mr. Metcalf heard of some sound-reinforcement experimental work the manufacturer was engaged in.

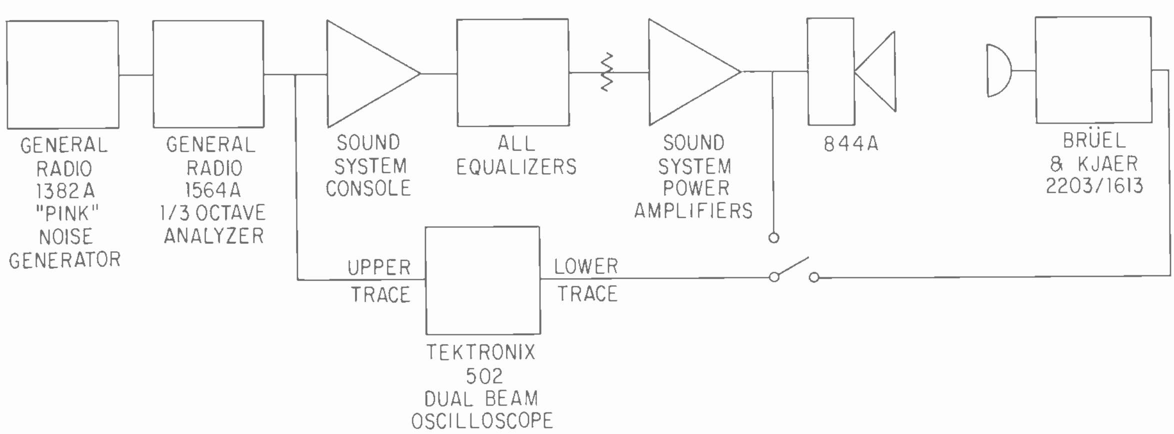


Figure 2. The type and interconnection of test equipment such as was used in Figure 1 Standing-wave effects have been averaged at low frequencies. Repeatability of these measurements is high—variations of curves made by several engineers was ±1 dB in any of the spaces measured.

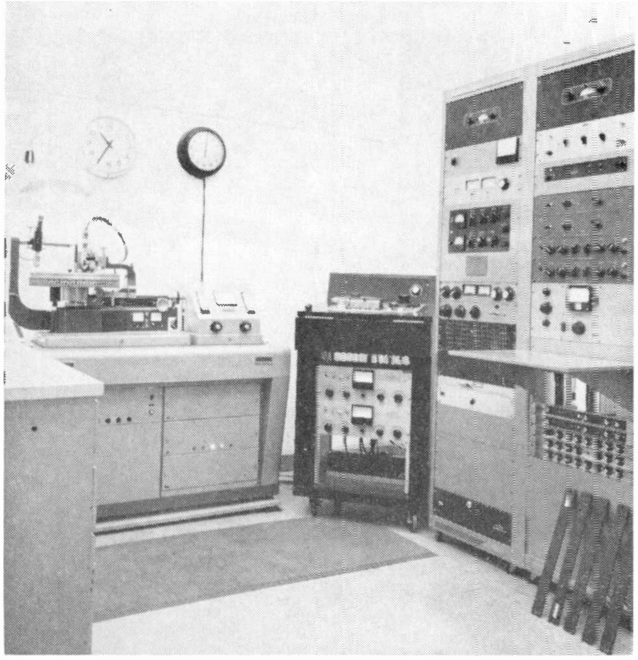


Figure 3. A view at Century Recording where the engineer stands in the cutting room. The sound-level meter was placed at ear level in front of the cutting lathe.

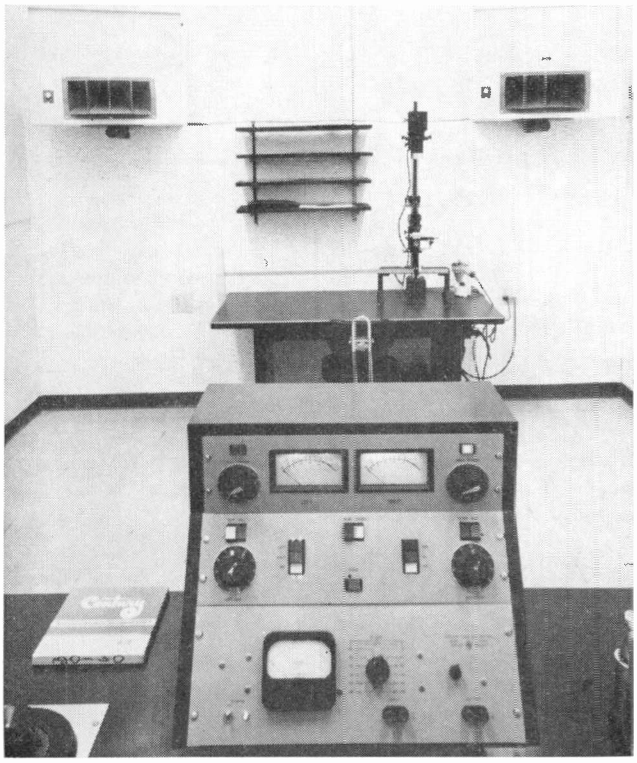


Figure 4. The view from the operator's seat at the console in Century's quality-control room. The two speakers (Altec 844A) are aimed at the operator's head. The sound-level meter was placed at the camera position for equalization measurements.

He at once contacted the engineer in charge of the project and asked if it was applicable to his studios. The factory engineer stated that he felt so, but had never tried to tune a loudspeaker in a small studio. Arrangements were made and a couple of engineers from Altec Lansing's new *Acousta-Voicing* division arrived with a ranch wagon load of acoustical test equipment and a set of Altec *Acousta-Voice*™ filters.

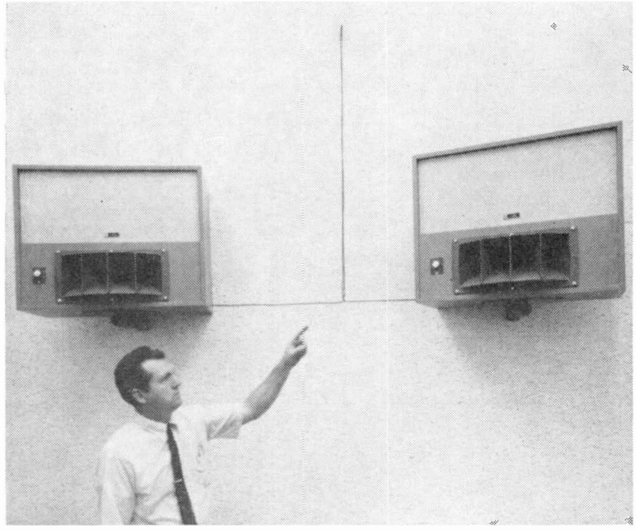


Figure 5. Robert Metcalf, chief engineer at Century Recording, points at the special, rigid mountings he had fabricated for his studio monitor speakers.

(Up to this point, all work with these units had been directed at solving the feedback problem in reinforcement systems, and with startling success.)

The chart in Figure 6 (black line) shows how the system and room interacted at the engineer's ears. This is the same type speaker as shown in Figure 1. The dash-dot-dash line in Figure 6 illustrates the frequency response at the engineer's ears after *Acousta-Voicing*. The dotted line is inverse of the *Acousta-Voicing* filters' electrical response. It can easily be seen that Mr. Metcalf's ears were correct in telling him that the bass response in this room was lacking and it was just this complaint that he had voiced about the new monitor speakers. As it turned out, the older monitor units had bass *accentuation* which helped in this particular room, but were not right in the quality-control studio.

This series of test measurements on the new monitor *versus* the old monitor illustrated the relationship of their room frequency-response curves to the difficulty in tuning them to a uniform frequency response in this acoustically troublesome cutting room. It was found that both units could be brought to the same acoustic response uniformity but that the older, not as smooth unit, required more filters for its successful tuning. The additional filters cost more than the difference between the cost of the old and new monitor system. It is easy to see that equipping all the studios with the new monitor units was far less costly than trying to *Acousta-Voice* the old units.

Quality Improvement

Musical instruments heard on the *Acousta-Voiced* system are now in clear, sharp relationship to each other; balance between bass and treble is highly realistic and all stridency is removed. As each over-emphasized tone is brought into equality with all the normal responding tones in the room, the sound quality is vastly improved. When all channels of a stereophonic system are heard at the listener's ears with an acoustic accuracy of $\pm$ 1dB, the spatial realism becomes *eerie*. The ability to move a sound smoothly across channels with a pan pot becomes as exact as focusing a fine lens on a precision camera.

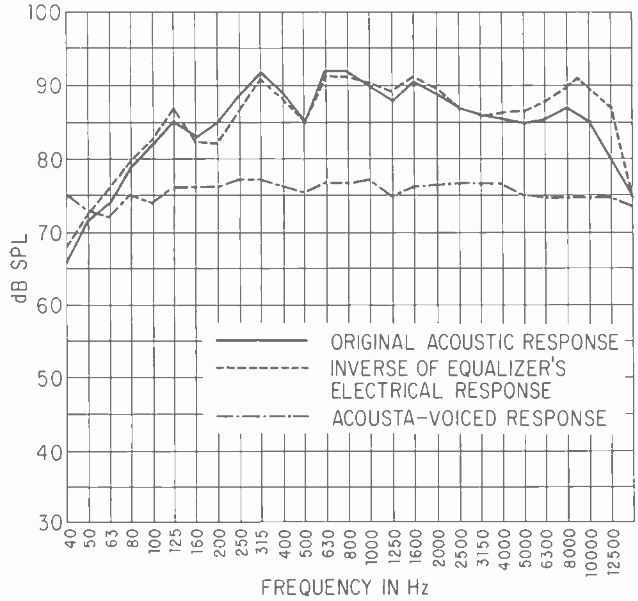


Figure 6. The result of equalizing one of the six channels. The continuous black line is the original acoustic response. The dash-dot-dash line is the Acousta-Voiced response at the engineer's ears. The dotted line reveals how closely the equalizer's inverse electrical response matches the original acoustic response.

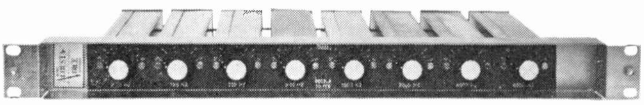


Figure 7. The Altec Lansing model 9018A broadband equalizer. Note that each section is a separate module.

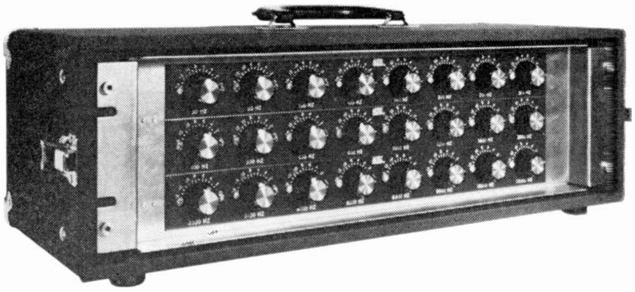


Figure 8. The Altec Lansing model 9014A narrowband equalizer in its portable carrying case.

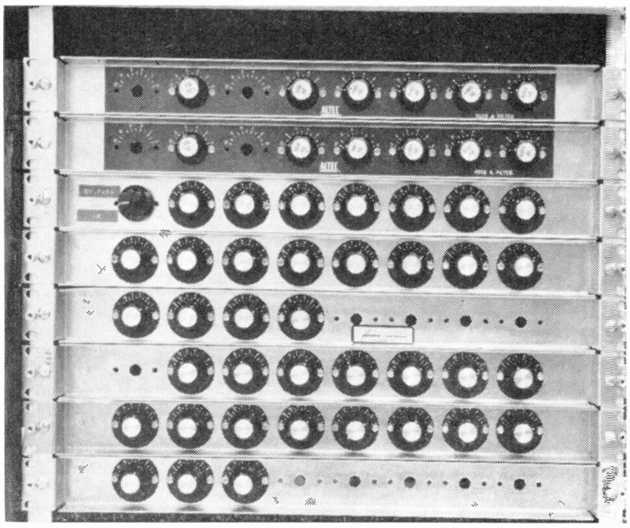


Figure 9. The actual filter sections used at one of Century's rooms. The top two chassis have the needed filters from a 9018A equalizer. The bottom six contain the necessary filters from a pair of 9014A equalizers. The filtering system is tuned for a two-channel audio system. Note the A-B comparator switch to allow comparisons with or without equalization.

Aural Acuity Demonstrated

One very interesting side light to this tuning that reveals the degree of aural acuity that professional recording personnel can develop occurred in the following manner: In each studio were two channels, hence two sets of equalizers. (See FIGURES 7, 8 and 9 for various views of the equalizers used.) To allow "before" and "after" comparisons of the effects of the tuning, a comparator was supplied which switched the equalizers out and substituted a bridged T attenuator set for the same insertion loss as the equalizers.

One switch developed a malfunction due to the heavy demands on it for demonstration purposes. Mr. Metcalf called Hannon Engineering, the sound contractor who supplied the *Acousta-Voicing*. They had a technician in the area who made the repair. Mr. Metcalf called back the next day to say that the system didn't sound right since the technician worked on the switch. Hannon sent an *Acousta-Voicing* engineer to Century. He found that the technician had inadvertently switched channels so that the left equalizer fed the right-channel amplifier and the right equalizer fed the left amplifier. The difference between the two channels was not major, just a few dB difference here and there. Mr. Metcalf, however, having become acquainted with really uniform sound found these few dB quite noticeable.

This is striking evidence that the historic disregard of the large variations we have endured (FIGURE 1) is perhaps a conditioned situation and that, in actuality, it is possible to really use and benefit from ±dB control of the acoustic response of the system.

One Caution

One caution should be tendered at this point. If the monitor loudspeakers are not heavy-duty two-way professional systems, they are more than likely not suitable for such equalization. Three-way speaker systems tend to burn out their very fragile high-frequency driver while undergoing test signals of white and pink noise.

Because of the dearth of spectral power distribution information in the past (accessible primarily to those companies that supplied equipment to the motion-picture industry and had a chance to gather such material on the live sound system by means of a real-time spectrum analysis) many manufacturers assumed an unrealistic power requirement for the higher frequencies and compromised on one to three watt high-frequency units. Additionally, three- or four-way speakers present insurmountable phasing problems if really smooth mid- and high-frequency response is to be achieved.

(To be concluded next month)

db Visits Dubbings Electronics

LARRY ZIDE

Beginning a new feature that will appear from time to time. This month our camera visited a large-scale tape duplicator that has made a specialty of cassettes.

The production of duplicate tape recordings for large-scale use has become big business. One of the oldest firms in this business is Dubbings Electronics of Long Island, N. Y.

Dubbings began operating in the early 1950's. Unlike other duplicators they chose not to take the route of the standard master/slave dubbing system. Julius A. Konins, company founder, and the late Bob Marshall of Fairchild Recording devised a special common-capstan duplicator. It is this machine, more than any other factor, that has placed Dubbings where it is today.

That original duplicator was a full-track mono machine. Soon afterward there followed duplicators that could handle the new half-track mono, two-track stereo (stacked and staggered), and four-track stereo. At the same time, methods were being achieved to use higher duplicating speeds while maintaining established quality standards. In the course of their history they have produced tapes recorded at three or four different speeds, variable-speed tapes for non-capstan machines and specialized precision test tapes.

The introduction of the Philips cassette found new use for the common-capstan duplicator. Dubbings was the first in this country to offer recorded cassettes. This early start developed a leadership that Dubbings has not relinquished. Today they are the nation's largest producer of recorded cassettes and, at the same time, have a substantial business in four- and eight-track cartridges and reel-to-reel. They also produce private-label and their own labels of recorded materials.

Recent company figures have 30 million feet of tape per week flowing through their plant. Much of this business has come to them over the last year. They have grown in this space of time from 46 production people to 176 (and additional recruitment is underway).

Figure 1. The Dubbings management team. Left to right: Julius A. Konins, v.p. and founder; Paul C. Smith, Jr., president; and Robert A. Harris, v.p.

Concurrent with this rapid growth has been the absorbtion of Dubbings by the Consolidated Electronics Industries Corp. Consolidated, a public corporation, also owns Mercury Records and has direct links to Philips in the Netherlands. This has placed at Dubbings' disposal Consolidated's industrial experience and technical know-how, and has afforded them channels for exchange of ideas and techniques through technical assistance agreements with Philips of the Netherlands via the North American Philips Company. Is it any wonder that Dubbings have become so prominent in the cassette field.

(*Turn the page*)

Figure 2. Part of the chain of common-capstan duplicators. The racks contain the master tape which may be in conventional reel-to-reel form, or an endless-loop storage bin toward the bottom of the rack.

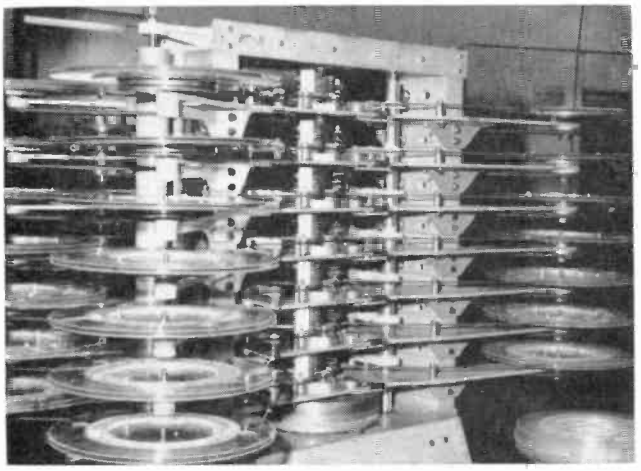


Figure 3. A single common-capstan duplicator can simultaneously run fourteen tapes. For cassettes the master speed is 120 in./sec., while the duplicator is running at 30 in./sec .All four stereo tracks are recorded in the single pass.

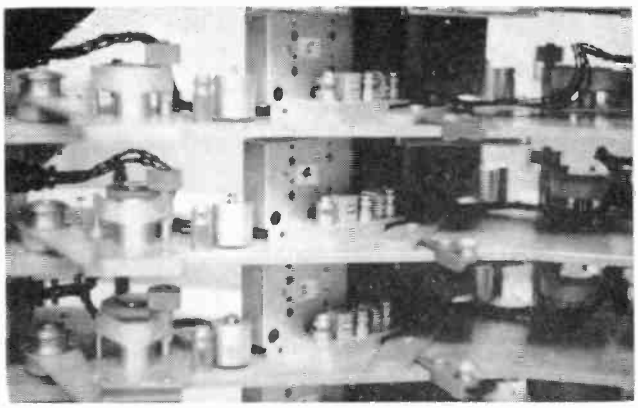


Figure 4. The head and drive assemblies on the duplicator. Because of the fast speeds (and ultra-high frequencies encountered) special recording heads with high resonant points are used.

Figure 5. The large reel containing the duplicated tape is placed on this Philips-designed automatic loading equipment. The machine attaches a leader to the tape, winds it onto a cassette hub, then cuts it at the end of the program material, placing a closing looped leader on the tape.

Figure 6. This close-up of the loading machines seen in Figure 5 shows the operator controlling the winding of a cassette hub. The two white rolls on the machine (to the left of the operator's hands) contain the leader material.

Figure 7. While the tapes are being processed, a separate production line is pre-assembling the cassette body.

Figure 8. An operator places the recorded hub into the body of the cassette and threads the looped leader onto the other hub.

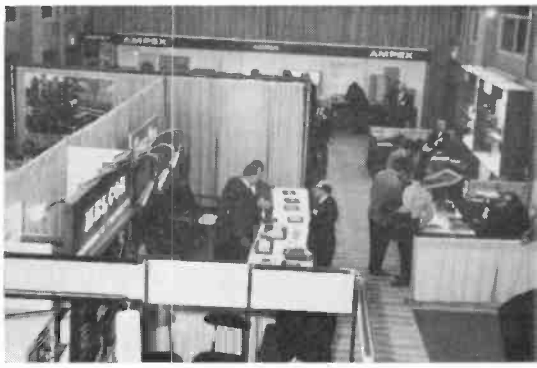
Figure 9. The loaded cassettes are checked before they are fed to a machine that will seal the cover to the body. Then other automated equipment labels the cassette, places it in a case, and wraps it.

Picture Gallery: East Coast AES Convention

ON this and the following pages are the results of our roving camera's visit to the thirty-fifth Audio Engineering Society's Convention held in New York City on October 21st through the 24th.

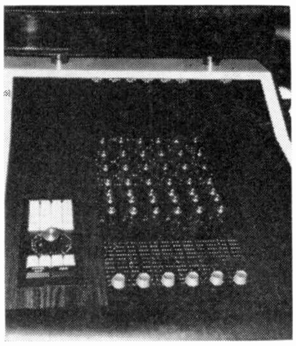
This was the biggest convention yet. Many people contributed to making it the best yet. If we can single any individual out, the tireless Jacqueline Harvey seemed to be everywhere she was needed, and seemingly she was needed everywhere.

On this page — a montage of scenes at the exhibition. Following, is a display of the *products* shown. Many, of course, premiered at this show.

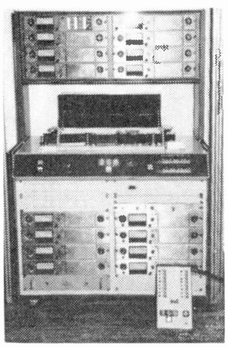


THE following illustrations highlight much of the new material shown at the AES Convention. Each product photo is keyed to the Reader Service Card at the back of this issue. Circle the appropriate number for further information to come directly from the manufacturer.

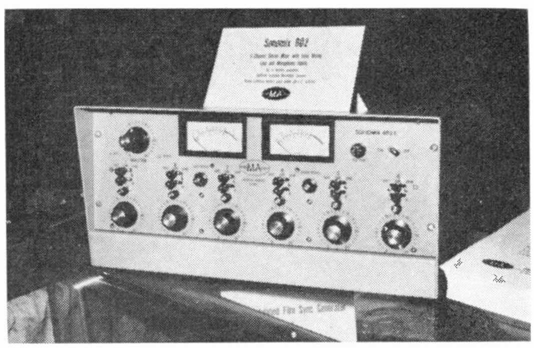
R.T. Bozak Mfg. Co. Model CAV-6-2 permits the user to adjust the balance between its two mono outputs so that the apparent location of sound sources correspond to the physical locations of microphones. There are six inputs, each divided to drive both left and right speakers. Power output is +18 dBm or 6 volts into 600 ohms. Price: $399.00. Circle 58 on Reader Service Card.



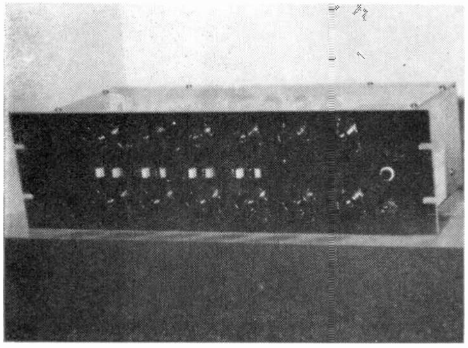
Audio Designs & Manufacturing, Inc. The Audex computerized switcher has been designed to simplify bus selection and the mixing of multi-channel consoles. Model 1 separately switches up to 16 program channels and 4 echo channels. When used with the Audex switching control it is possible to switch up to 32 Audex modules. Input positions to be programmed are selected on the control. Echo channels may be selected without cancelling a program channel. Selections are indicated on a digital readout. Model II can select up to 20 modules. Circle 70 on Reader Service Card.



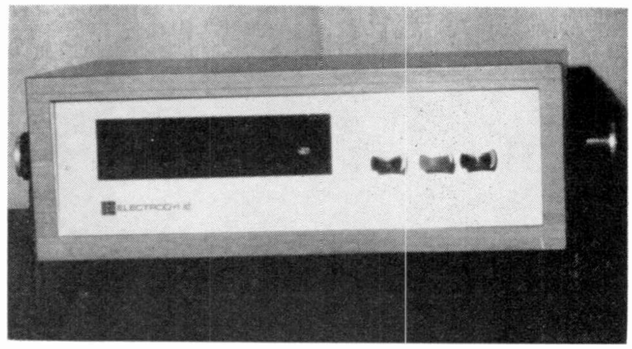
Ampex Corporation. MM-1000 series master units can be had in up to sixteen-track versions. Sel-sync selection can be made on the machine or with a remote-control accessory that also provides tape-motion control. Circle 67 on Reader Service Card.



Martin Audio Corporation. Sonomix 602 is the designation for a new stereo six-channel mixer with echo mixing, line, and mic inputs. Battery or a.c. operation. Price: $945; $1095 in case. Nicad battery equipped adds $70. Circle 52 on Reader Service Card.

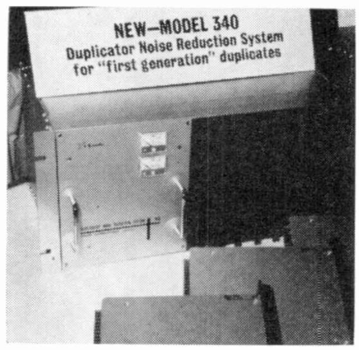


Gately Electronics. Model EM-7 echo miser is a four-channel stereo mixer designed to be a basic building block for applications needing echo addition or equalization of the signals to be mixed. Each channel has an associated echo-send control. Each of the two outputs has its own individual gain control and echo return control. Two or more mixers may be stacked with their mixing busses tied together. They may also be cascaded to permit mix downs from multiple sources. Price: complete with 6 mic preamps $725.00. Circle 64 on Reader Service Card.

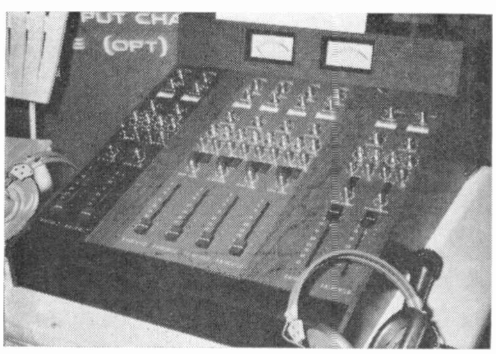


Electrodyne Corporation. An electronic stopwatch with readout numbers that will not be mistaken as by misreading of conventional clock faces. The DC-900 series has versions of this unit that count up or down, with or without preset, offers times in seconds or tenths of a second — up to an hour and forty minutes. The clock can be rigged for remote operation. Also offered is an optional tape motion sensor for high-speed direct timing of audio and video tapes. Integrated-circuit logic is used. Price: $495.00. Circle 63 on Reader Service Card.

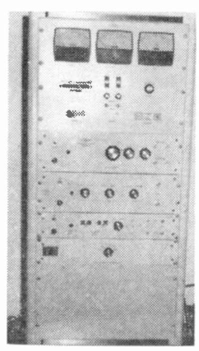
Crown. A dual-channel basic amplifier with a power response of d.c. to 20,000 Hz ±1 dB at 150 watts r.m.s. per channel at an 8-ohm load. Stability is secure with all speaker loads, capacitive loads under 1 mFd, and other capacitors if isolated by 1 ohm. There is complete protection against shorts, opens, and mismatches. At 8 ohms, typical i.m. distortion is under 0.1 per cent at any level up to full power. Price: $685. Circle 62 on Reader Service Card.



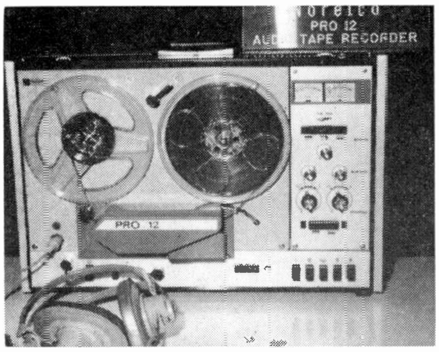
Dolby Laboratories, Inc. Model 340. Has been designed to process noise-reduction recordings at the high speeds used in commercially-recorded dubbing operations. Frequency response extends to 640 kHz, providing full 20 kHz bandwidth at up to 32-times normal speed. The end result will be an NAB recording with substantially the same noise level as if it had been made at the original recording session. Price: $1,950.00. Circle 57 on Reader Service Card.



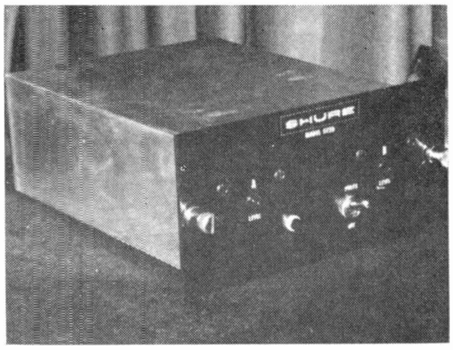
Fairchild Recording Equipment Corp. This console unit is built around individual modules with a +18 dBm output. A complete module includes built-in compression and equalization. Price is $525. The price goes down from this level as individual function is removed from the module. Circle 66 on Reader Service Card.



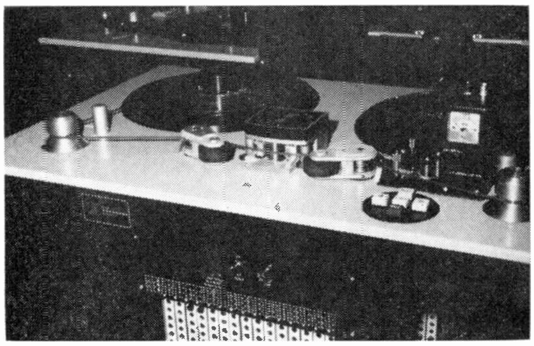
HAECO, Holzer Audio Engineering Company. CSG system. A complete package for disc recording that offers a stereo logic device which recognizes the common signal components in the channels of a stereo system and permits combining the channels externally or in the playback cartridge for full mono compatibility. Completely solid state. Available on lease. Circle 56 on Reader Service Card.



Norelco, Philips Broadcast Equipment Corp. High reliability and ruggedness are claimed for this 3¾, 7½ in./sec. recorder. All transistors and suitable for twin-track stereo or mono. There's also a quarter-track stereo version. Cue and dubbing facility, multi-play and s-o-s, three-head monitoring, headphone jack, remote-control facility. Circle 55 on Reader Service Card.

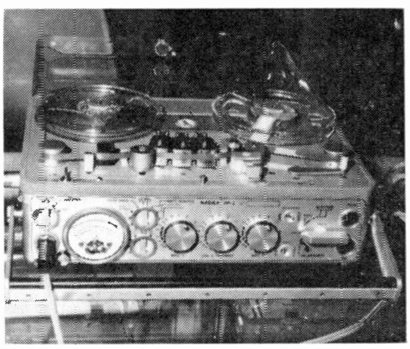


Shure Brothers, Inc. Model SE20 is a solid-state stereo transcription preamp. Precise RIAA/NAB equalization is included, noise and r.f. susceptibility are extremely low, full line capability is to +20 dBm. Distortion is under 0.5 per cent at +20 dBm 20-20,000 Hz. Input is 47,000 ohms; output is for 600- or 150-ohm balanced line. Price: $336. Circle 68 on Reader Service Card.

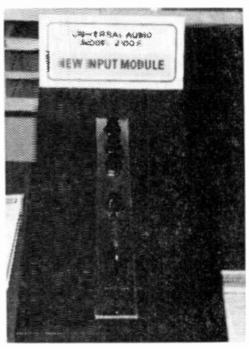


Gauss Electrophysics, Inc. Series 1200 tape slave. This model takes cassette-width tape and runs at 120, 60, or 30 in./sec. Price unmounted: $8,350.00. Circle 54 on Reader Service Card.

McMartin Industries, Inc. The LX-600 is a six-input solid-state mixer. Bass and treble boost and cut, master gain, earphone jack. Circle 53 on Reader Service Card.

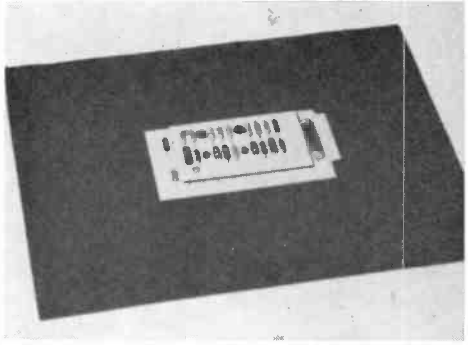


Nagra Magnetic Recorders, Inc. The IV-L model has two mic and one line input, mixing facility, manual or automatic level control, low-frequency rolloff, and unbalanced line output. This model also adds three speeds (15, 7½, and 3¾ in./sec.) and limiting and fading features. Camera-sync. facility can be added to any unit at additional cost. Price: $1,660.00. Circle 59 on Reader Service Card.



Universal Audio. Primary intent of this input module, model 2100S, is as a microphone channel in a control console. It incorporates the features of amplification, attenuation, and equalization for program and reverb. It is a self-contained plug-in unit except for power supplies. The transformer input is switch attenuatable, outputs are through a transformer and are not grounded inside the unit. Price: about $400. Circle 61 on Reader Service Card.

Tandberg of America, Inc. Pilotone 11-1-P is the latest version of the three-speed battery portable (7½, 3¾, 1⅞ in./sec.). Compatible with standard 16 and 35 mm. cameras for sound-sync. Five heads including special tachometer head for speed control. Takes up to seven-inch reels. Price: $699.00. Circle 51 on Reader Service Card.



Spectra-Sonics. Model 601 is a compressor/limiter on a plug-in card. Field-effect transistors are used to provide independent compressor and limiter action up to 30 dB at under 0.05 per cent t.h.d. (1 kHz). Attack and release times are automatically variable. Price: $111.60. Circle 60 on Reader Service Card.

(There will be more new products from the New York AES convention featured next month.)

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People, Places, Happenings

- The appointment of **John F. Vorisek** as president of **Reeves Sound Studios** has been announced by **Hazard E. Reeves,** chairman of the board of the parent organization, **Reeves Broadcasting Corp.** Reeves Sound Studios is heavily involved in motion picture and television sound recording and has recorded the sound of many major t.v. and motion-picture productions. Mr. Vorisek is a recognized authority on sound recording. He has had his own company **Sound Enterprises,** been with the foreign department of **Universal Pictures,** he edited the stereophonic sound effects for the original *This is Cinerama*. Recent credits include editorial work on major t.v. productions. For a number of years he has been a consultant for radio and t.v. affairs to **New York Governor Rockefeller's** staff.

- **The Muter Company,** Chicago, Illinois, parent of **Jensen Manufacturing Co.** and **General Magnetic Corp.,** has released operating results for the nine month period ending September 30th. Net loss from operations for this period, after adjustments, was $382,585 or 27c per share. In 1967 the net loss from operations was 13c per share for the same period. Muter Company president **Herbert J. Rowe,** stated that a small operating loss was incurred in the July-September period, but the results were a significant improvement over the first six months of 1968. An operating profit was realized in the month of September and favorable operations are anticipated for the balance of the year, although they will not be sufficient to overcome the accumulated losses of previous months.

- **Allan Liggens,** P.E. has recently joined the Vancouver Canada-based consulting acoustical engineering firm of **Barron & Strachan** as a senior engineer. He is an electrical engineering graduate of the University of British Columbia with nine subsequent years of experience in electronic circuit and system design. He will serve the firm's clients primarily in the fields of sound reinforcement, audio/visual, and industrial control systems.

- **Milton Philipson,** has been promoted to the newly-created post of special marketing manager for **Craig Corporation's** products division. He was the firm's national accounts manager. In his new assignment, Mr. Philipson will be responsible for developing in-depth marketing programs for the sale of Craig mobile and home stereo tape recorders and players in military, government, premium, accessory, and catalog areas. Before joining Craig in 1967, he was national sales manager of **Roberts Electronics.**

- At **Visual Electronics'** subsidiary **Corinthian Electronics Corp.** comes the announcement that **Leonard G. West** has joined Corinthinan as chief engineer for a.m./f.m. transmitters. In making the announcement **Peter A. Tyrrell, Jr.,** Corinthian president, noted that Mr. West is the former field marketing manager for **Varian Associates** and has, earlier, been employed in various top-level engineering capacities in firms specializing in transmitter equipment.

- **Dean L. Burdsall** has joined **General Recorded Tape, Inc.** as accounting manager of the Sunnyvale, California firm. He is responsible for internal auditing and accounting policies and procedures at the corporate level. For the past eight years he was a senior accountant for **Peat, Marwick, Mitchell & Co.,** certified public accountants of San Jose, Calif. GRT is an independent producer of recorded stereo tapes for the entertainment, educational, government, and industrial markets.

- **Saul B. Marantz,** founder of the high-fidelity products company bearing his name, has announced that he and two engineering associates have organized a new enterprise specializing in subcontracting and custom manufacture of electronic equipment and subassemblies. In addition, it is offering engineering design and development services to the industry. Mr. Marantz severed his connection with the Marantz company late last year. He stated that the new firm has future plans for the development and marketing of its own proprietary electronic products. The new firm is known as **Ferrodyne, Inc.,** and is located at 35-01 Queens Blvd., Long Island City, N.Y. 11101.

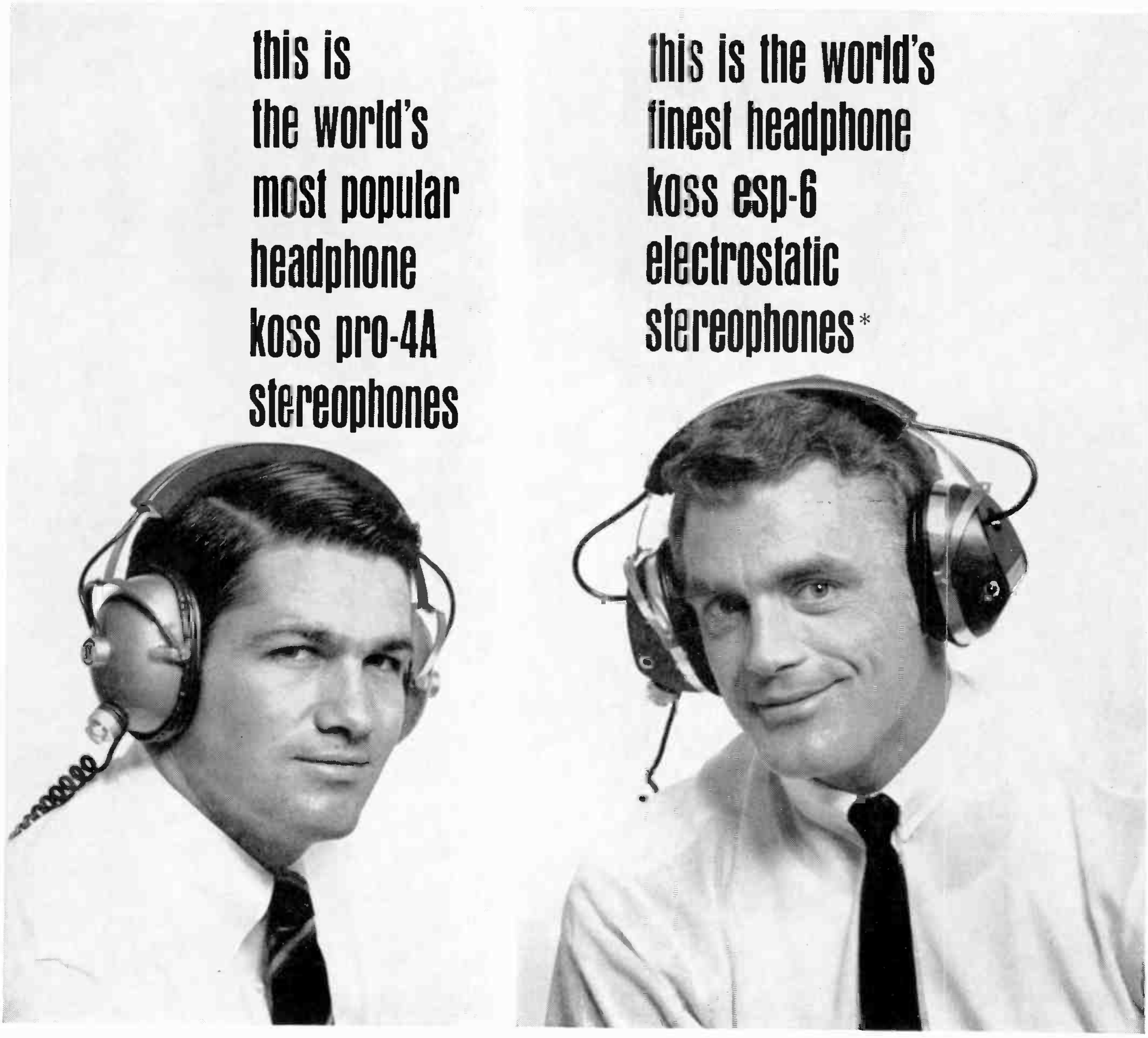
- Growth has caused personnel changes at the **DuKane Corporation.** According to an announcement by **J. McWilliams Stone, Jr.,** president of the firm, **Robert Alvarez** has joined the company in the newly-established position of materials manager. He will supervise the purchasing, customer order, and finished stock departments.

Thomas Jordan has been promoted to the title of works manager. He has been with DuKane for 33 years and was, most recently, plant superintendent.

Paul Cristiansen has joined DuKane as purchasing agent. He was last with Electric Products as manager of purchasing.

Arthur Taylor comes to the company as inventory manager. His last positions were with **General Time** and **Motorola.**

The recording industry was deeply saddened by the death of Claude L. Rie on November 16th. He was 40 years old. Mr. Rie was well known in the New York area to audio pros, particularly those with an interest in disc mastering at which he was an authority. At his death he was president of the International Recording Company with headquarters in Bronxville, N.Y. He shall be missed.



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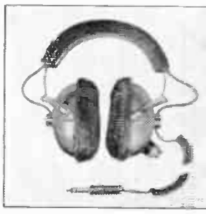

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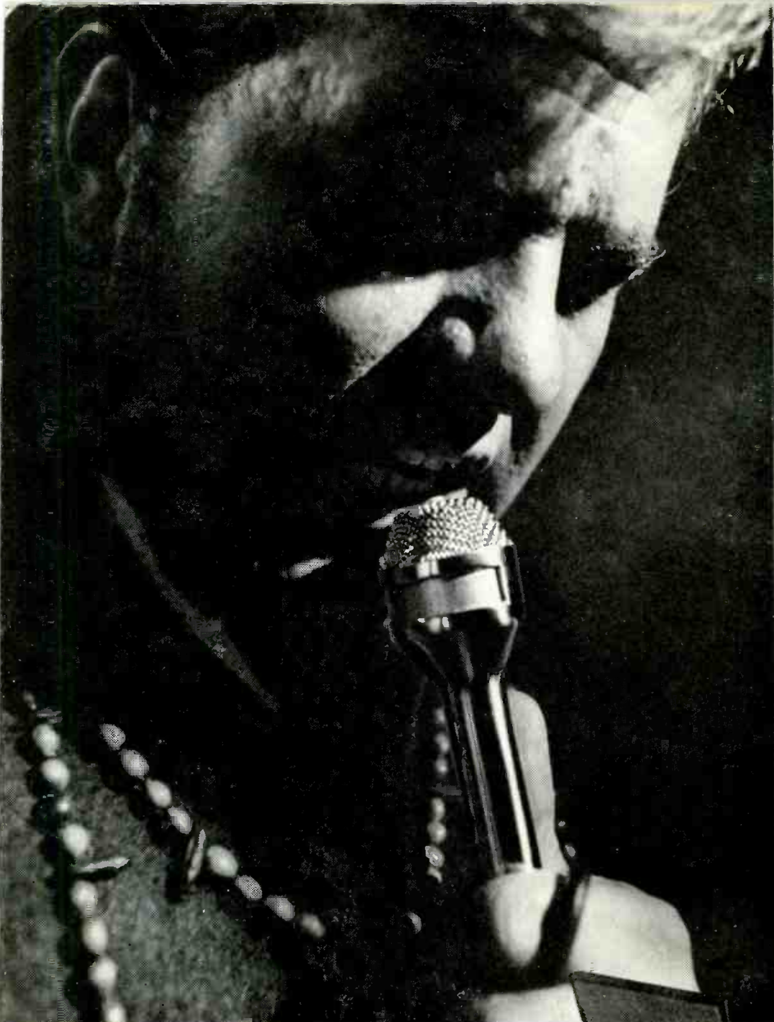

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