APRIL 1975
VOLUME 6 / NUMBER 2

INCLUDING: CONCERT SOUND REINFORCEMENT

RECORDING engineer/producer

RELATING RECORDING SCIENCE • TO RECORDING ART • TO RECORDING EQUIPMENT

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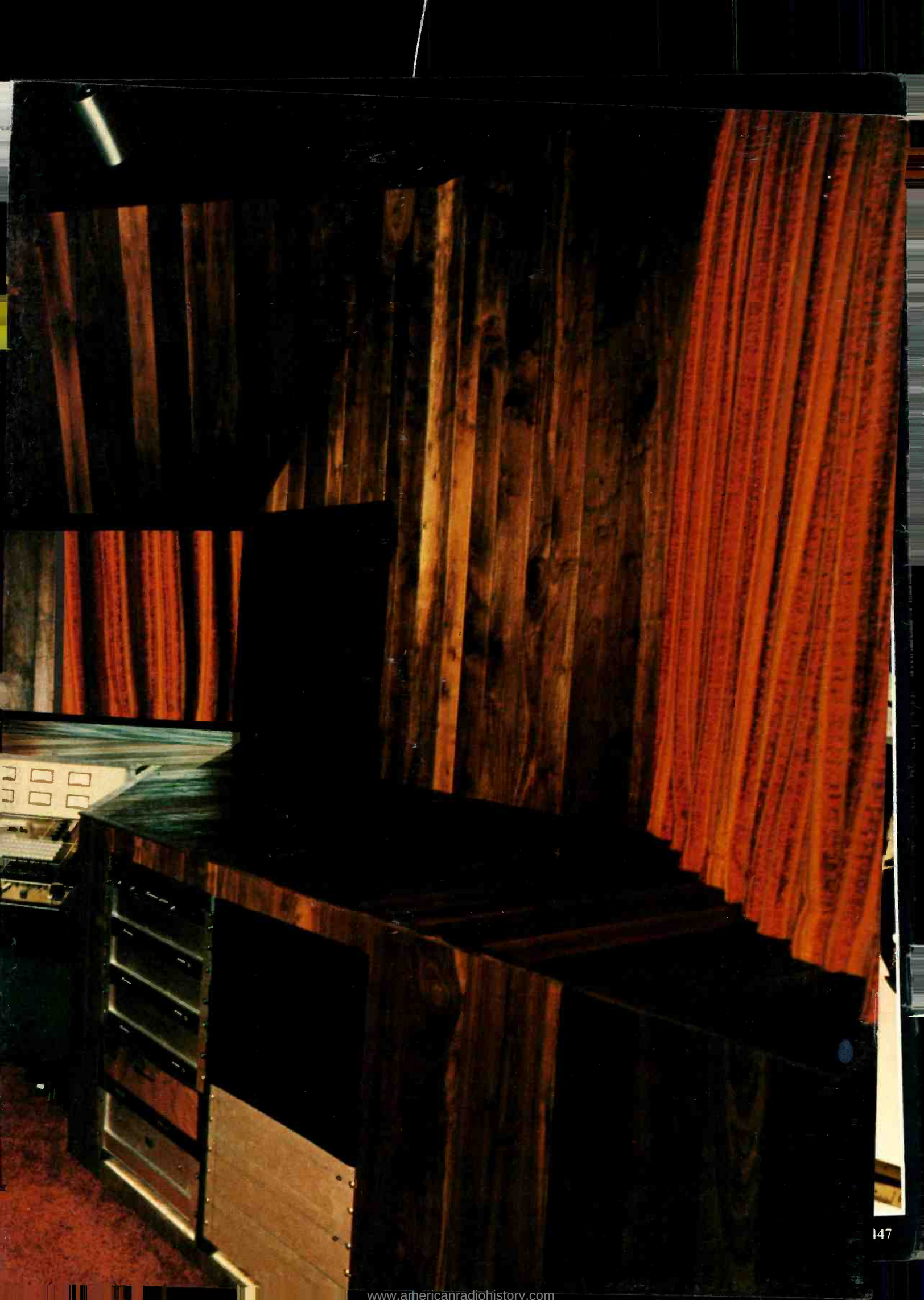

QUALITY: **SPECTRA SONICS** audio control consoles show the care and attention to detail that are the mark of the skilled American craf-sman. The internal wiring, module construction, console housing and the control display reflect the precision and distinctive craftsmanship that is characteristic of SPECTRA SONICS.

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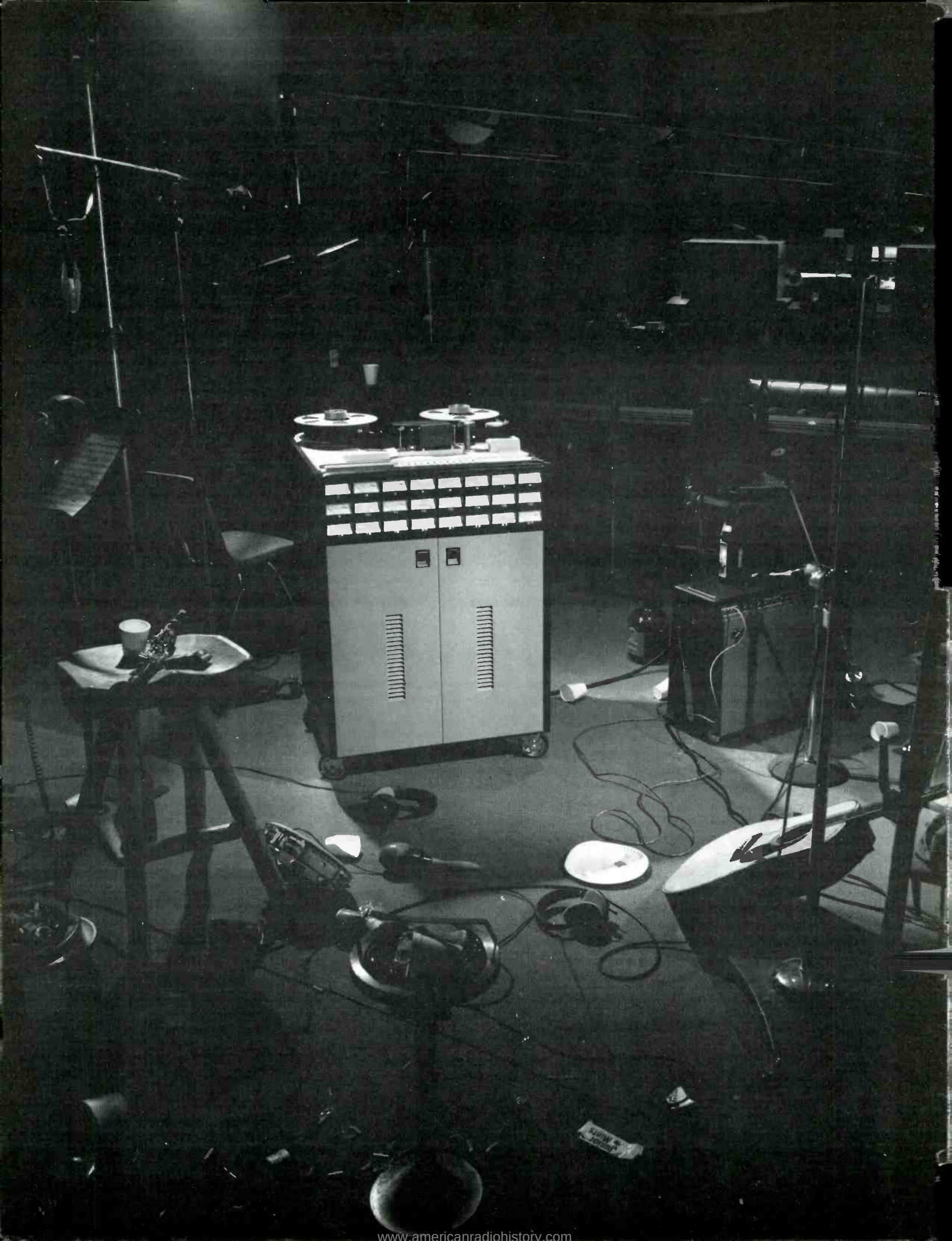

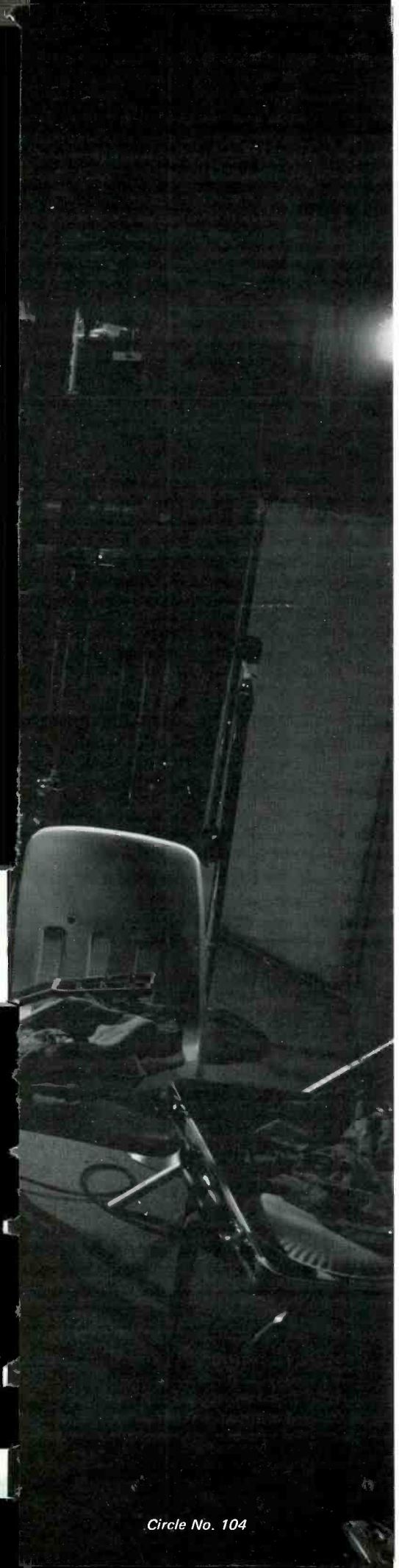

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One MM-1100 can give you 24 tracks. Two of them hooked up together with a synchronizer will give you 46 channels. That's enough to mike your setup for left-and-right running water, if you want it that way. And with all that channel capacity, you can save plenty of channels for sweetening, later additions, and a last-minute background by a hundred voice choir.

Three heads are better than one when you can change them yourself
Ampex makes 24-channel, 16-channel, and 8-channel head assemblies for the MM-1100, and changing them is as easy as turning one thumb screw and swapping units. Touch up the equalization and get on with the session. No need to worry about tape tension adjustments because that's all done automatically by the MM-1100 transport mechanism.

Never enough time, never enough tape. But we handle a 16" reel of two-inch easily
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In a delicate business like production recording, there just isn't any substitute for the Ampex MM-1100—a very rugged tape recorder.

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Audio-Video Systems Division
401 Broadway
Redwood City, California 94063
(415) 367-2011

Circle No. 104

R-e/p 9

RECORDING engineer/producer

– the magazine to exclusively serve the recording studio market . . . all those whose work involves the recording of commercially marketable sound.

– the magazine produced to relate . . . RECORDING ART to RECORDING SCIENCE . . . to RECORDING EQUIPMENT.

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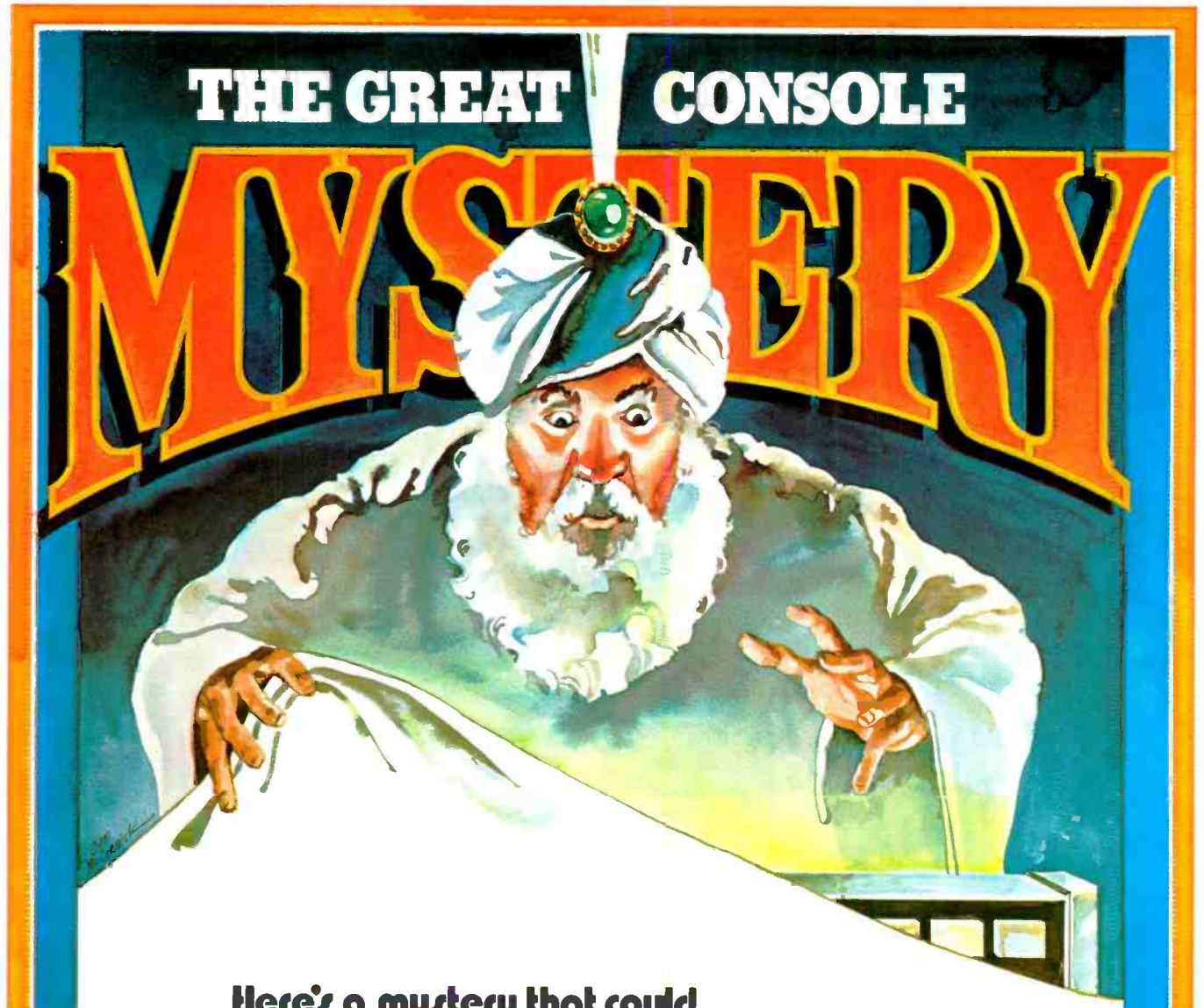

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A mystery that's worth investigating. Especially if you are making expansion plans. Thinking of bigger things.

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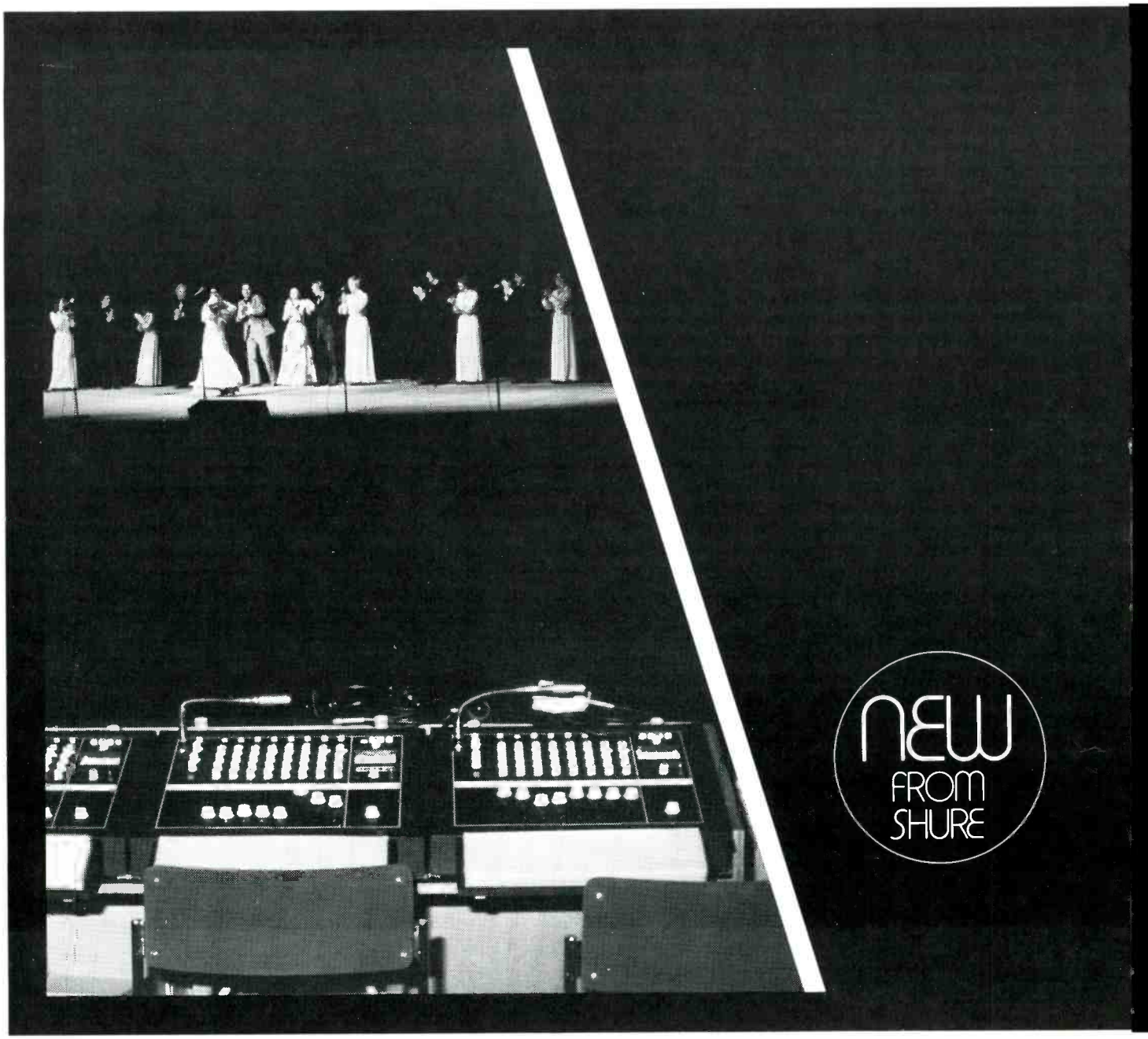
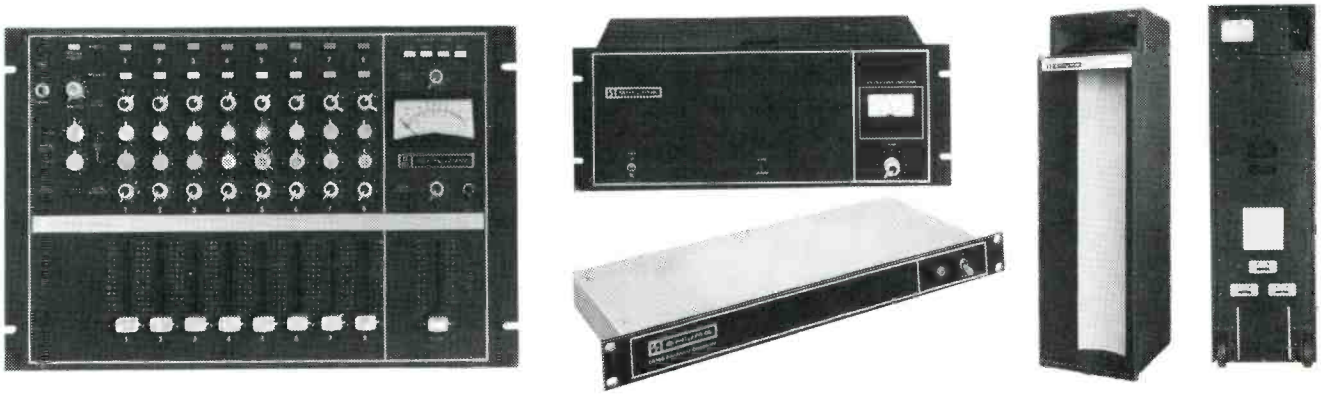


SHURE SR PROFESSIONAL SOUND REINFORCEMENT EQUIPMENT



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**Tom Moores talking about SR equipment.*

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YAMAHA

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Letters & Late News

From: Charles Ludwig
Ludwig Sound Specialties
Houston, TX

As a subscriber and reader of your magazine I find many of your articles to be of great interest and in-depth in their coverage. I find, however, that most of the articles deal with techniques in recording situations, or become analytical in nature, as relates to new products and equipment. I have not seen in any of the journals, any articles covering the subject which I am writing you about.

The purpose of this letter is to ask, and encourage you, to do an in-depth series of articles on the subject of radio interference with recording studios in particular, and audio installations as a whole. I understand, for example, that while I have suffered serious interference at my recording studio, that area churches, our City Hall, our Municipal Court Building, and even Houston's Astrodome have also experienced these problems. Very specifically, the problem of Citizens Band Radio Service causing interference needs to be looked at in these articles. In our operation we have paid the supreme price for this. The problem is of gigantic proportions and the longer I look at it, the more aware I become that I am not the only one in this business suffering from this problem.

It is not particularly hard to determine who the person is who is causing the trouble if it is a repeat situation, but it is next to impossible to do anything about it. We have had such a situation now for over one year where we have known the party or parties involved, and are way past the point where we can negotiate.

As a businessman and citizen, I first searched my own house to be sure it was in order and that it was not weaknesses on the part of our particular brands of equipment. I found that some of it, in fact, was weaknesses, however, from brand to brand the weaknesses were about the same.

Our studio equipment consists of Electrodyne, Scully, Ampex, and RCA, plus Neuman, AKG, and Sony microphones and an assortment of professional limiters, equalizers, etc. We found, without exception, all the major equipment mentioned here was susceptible; our Scully 16-track machine and our RCA 2-track machines were especially sensitive to radio interference; our Ampex model AG-440B, not as sensitive, and our Electrodyne console sensitive only in certain portions. Regardless of one's choice of brands, everything mentioned here is, at worst, good professional equipment.

We soufht relief from this problem first by contacting the Federal Communications Commission. They listened sympathetically to our complaints, they were nice, and that was about it. At the same time, we began chipping away at the problem piece by piece to eliminate or reduce it to the point where we could live with it. This proved to be in vain. In the months that have passed, we have filed eight letters of complaints of specific violations with the F.C.C., which got us nowhere at all. We finally got relief by filing lawsuits against the individual committing the violations for substantial monetary damages and injunctive relief, and against the F.C.C. to compel them in a court of law to enforce Section 47:303 of the United States Code.

Now, this is ridiculous and expensive.

The Citizens Band Radio problem in size is phenomenal and beyond the imagination of most people. I have been told by a member of the Commission that there are presently more than nine million licensed Citizens Band transceivers in the United States, and probably more than 80% of these are operated illegally. He had no idea how many there were that are not licensed. He estimates that over 50% of the violations have to do with excessive power over 4 watts D.C. in the final. Power levels ranging from 15 watts to as high as 1 kilowatt are quite common for base station operation, and power levels in the 100 to 200 watt range are common in mobile units. These transmitters are not F.C.C. type approved. Side band filtering is virtually non-existant, and the majority of these are garage-built from cheap components. Now, there is just no way average equipment can reject this kind of signal when distances are very short, say from the street to the control room.

If you choose to do a series of articles on the subject, I would like to see you cover three phases.

(a) Define the size of the Citizens Band Radio population.

(b) Define the manufacturer's responsibility to its' customers of building its' equipment it intends to market so that it is not susceptible to this kind of interference.

(c.1) In-depth coverage of ways which someone like ourself can go about installing direct, to the point effective solutions, not just theories. (Grounding in itself is not the answer.)

(c.2) Explore the possibilities of corrective solutions by the choice of building materials used in structures, such as screening the inside walls using metal buildings as compared to standard construction, etc.

ED: We would very much like to hear from others who might be similarly plagued, so that a presentation can be made to the FCC. An article is in preparation for publication in a future issue.

BYU AUDIO RECORDING TECHNOLOGY COURSE SET FOR JUNE 9-27

For the ninth consecutive year, the Brigham Young University (BYU) in Provo, Utah will offer a summer program for those interested in the audio recording and related fields. The course this summer will be similar to the highly successful format implemented last year, with several improvements. Tentative arrangements have just been completed to again have Norman Crowhurst, eminent author, lecturer, and consultant in communications, as a member of the staff for this year's program.

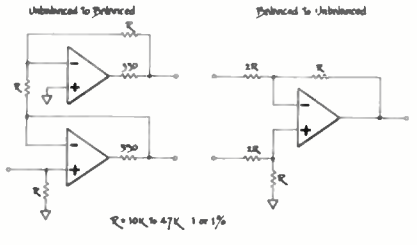
The course will again be an intensive five-day-per-week program lasting three weeks, during which a full semester's academic work will be completed. However, this year, to diversify and provide a more self-paced approach to instruction, much of the "basic" instruction will be given by means of audiovisual materials which will be supplied to each person who enrolls. In addition to this "basic" packet of materials, introductory and prerequisite materials will also be available in individual instruction packets for review before the course actually begins. The group sessions involving the entire class will thus be devoted specifically to workshops and demonstrations which cannot conveniently be handled on an individualized basis.

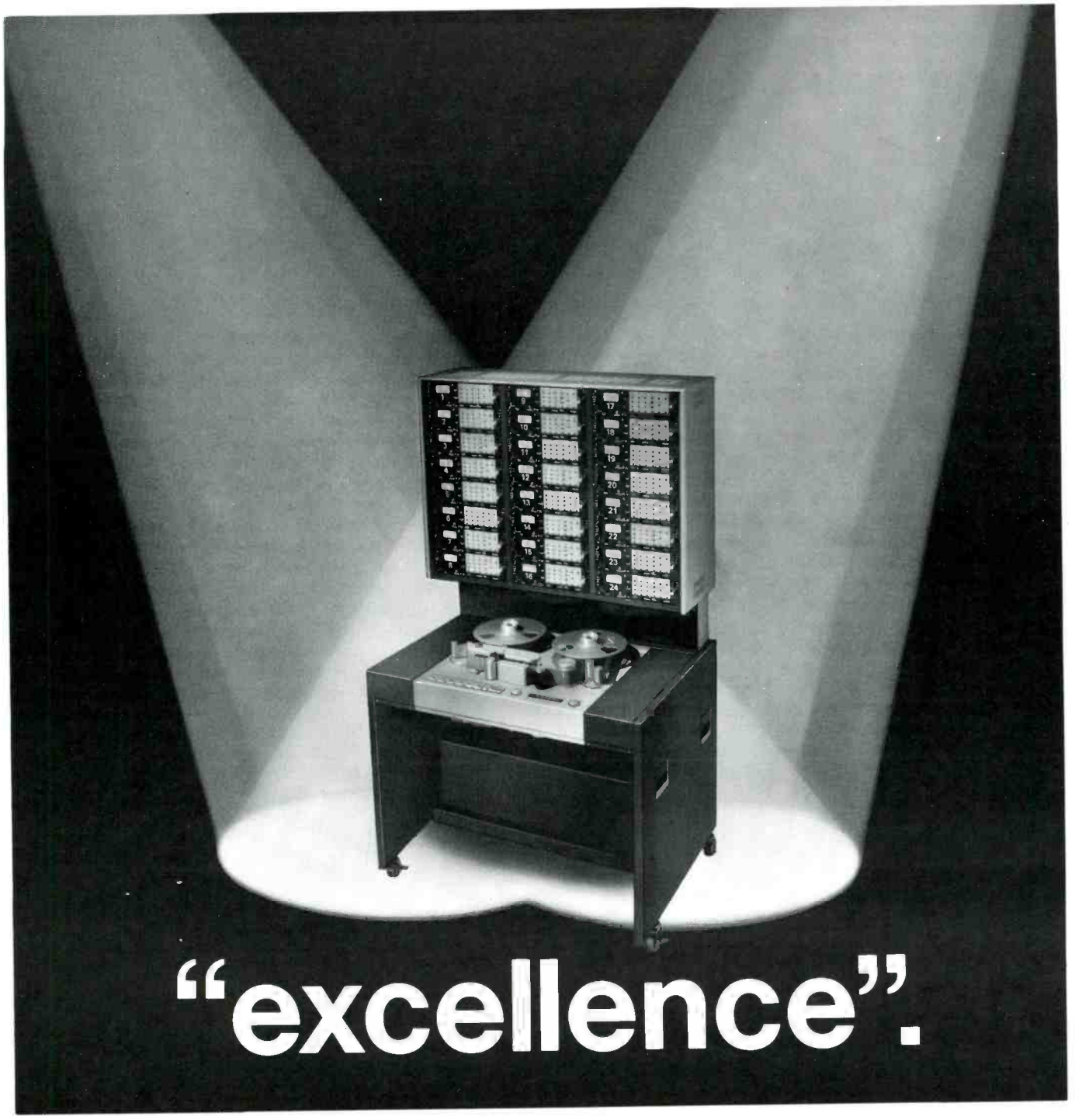
CORRECTION

Our apologies to JOHN W. CALDER whose letter concerning his experiences recording in London appeared in the February issue. Mr. Calder engineers at COOK HOUSE RECORDING STUDIOS in Minneapolis, not Smoke House, as was inadvertently typeset.

CORRECTION

The schematic diagrams for Peter DeBlanc's transformerless Balanced to Unbalanced and vice-versa circuits (Page 49, R-e/p Feb. '75) lacked labeling of the inverting and non-inverting inputs for the op amps. They should be labeled as shown.





For over twenty-five years the *Studer* name has stood for excellence. Through the years Studer tape recorders have come to be recognized throughout the world as a standard of the broadcasting and recording industries. Studer's A80/VU Recorder was developed for use in recording systems where only the highest quality is acceptable. The great care taken in the design, development and construction of the A80/VU is reflected in its unusual reliability and stability. A superior transport combined with a versatile electronics system and high-precision heads allow studio personnel to work surely. With the recorder a stable, predictable instrument, the energies of the mixer and producer can be applied totally to the creative aspects of their work. For complete information contact . . . WILLI STUDER AMERICA, INC., 3916 Broadway, Buffalo, New York 14227—phone 716 681-5450. (In Canada: WILLI STUDER CANADA, LTD., 416 423-2831)

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Circle No. 108

FROM: PAUL RAINEY
PAUL RAINEY RECORDING
VILLA PARK, ILLINOIS

(On wiring a Heathkit stop clock to the controls of a recorder.)

There are at least two electronic stop clocks on the current market as seen in the ads of several publications, but for many studios they suffer the drawbacks of excessive cost (nearly $200 for the least expensive), and excessive accuracy just not needed by most studios. I regard it a sad commentary on any part of the industry that would be so uptight as to consider tenths of a second so important, but let's let that be and proceed with something very real, practical and inexpensive.

The electronic stop clock described here is a Heathkit model GC1005 electronic clock with external adaptions to take advantage of two of its built-in characteristics to alter it to a stop clock function. The first step is to buy and build a kit, wiring it for 24 hour operation. Once it is done and in operation, unplug it while watching the readout. You will note that (1) the readout holds the time when power was cut, and (2) it takes 2 to 3 seconds for the power supply filter capacitor to discharge enough for the numerals to go out. Next, plug in the clock again. It will read all eights. Push the TIME HOLD switch to *hold* — the clock reads all zeros. Push the TIME HOLD switch to *run* — the clock starts counting from 00:00:00.

That's the way Heathkit designed it. Now let's bring out some external controls. I have interwired the clock with the START, RECORD, and STOP buttons on my mixer console so that as the master recorder is operated, the stop clock also functions, with no extra buttons to push. Here is the sequence.

Button Pushed	Recorder Function	Clock Function
START	Start	Nothing (still reads all 8's from last stop)
RECORD	Record	Resets to zero
START	Nothing	Starts counting
STOP (hold button)	Stop	Holds last count, fades out
(release)	Nothing	Displays all 8's

When you are ready for a take, push START and RECORD as you normally would in sequence. The recorder starts and the clock resets to zero. Then as the music begins, press the START button again. This does nothing to the recorder, but starts the timing. Then when the recording is finished, depress the STOP button and hold it until you note the time displayed on the clock. There is plenty of time to see it before it fades.

Figure 1

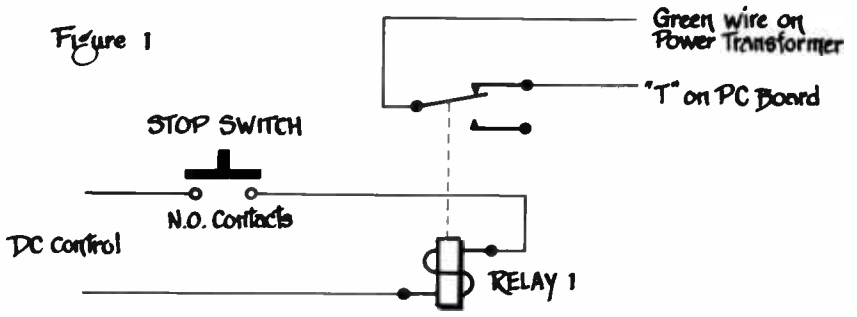
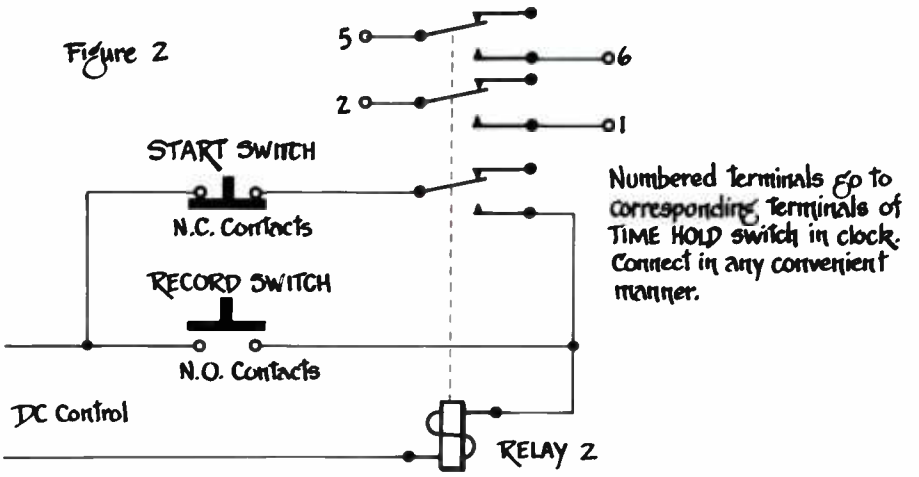


Figure 2



Releasing the STOP switch will throw up all 8's.

This sequence of operations is extremely convenient as it requires no extra buttons. One more surprising benefit; whether your channels are in safe or record this sequence still goes, which means that the same sequence of button pushing will get you a timing of a previously recorded take that you are now overdubbing, syncing, ping-ponging or just plain listening to (use whatever terminology you wish — the sequence always works).

The clock kit makes a very nice looking unit, so the package was left as is and placed on top of the mixer in an easy to see place. The connecting cable and socket were already on hand in spare parts boxes. The basic circuit, as described, requires six conductors and two relays.

Relay 1 (in Figure 1) is energized through a spare set of NO contacts on the STOP button of the recorder. NC contacts on Relay 1 open the green wire from the power transformer to point T on the circuit board. Pushing the STOP button removes 18.5 VAC from the two rectifiers supplying the IC timer, causing (1) no further counting, and (2) the 15 VDC to bleed down causing the display to extinguish. Release of the STOP button reconnects power to both rectifiers, but the characteristics of the IC are such that all 8's are displayed, showing power has been off.

The relay we found for Relay 2 was too large to mount in the clock case, so it was mounted in the console and the four wires were brought down to it from the clock. Figure 2 shows it is used as a latching relay. Pushing the RECORD button latches Relay 2 and a pair of its NO contacts parallel the operation of the TIME HOLD switch on the clock. Pushing the START button opens the latch circuit of Relay 2, and it drops out, releasing the TIME HOLD circuit, and the clock starts counting.

There you have it; one clock kit plus two relays and a little time gets you something better than any commercially available unit for simple stop clocking at considerably less cost. And, if you wire to the function buttons as suggested, it is much more convenient.

If ordinary clock operation is desired, just unplug the connections to the relays and set the clock as usual. Now that you are done and it is working, will one of you genius type people among the readers please examine the timing and reset circuits of the clock? It seems to me that after the stop button is released, there ought to be a way of making the clock resume counting the correct time-of-day as it does after you have set the alarm, and to make the display switch from time-of-day to stop-clock operation when the RECORD button is pushed. That would make it really useful. I want to build mine that way, even if it takes a second IC counting chip and a diode or transistor switchover from one IC to the other.

Choose one:
1.4 cubic feet or 12 cubic feet

Interface: A or Sentry III. Systems with different names and substantially different appearances. Yet both issue from a common technology and what we believe to be the important performance criteria.

Flat frequency response, uniform total acoustic power output, extended bass without lumps, low distortion...these goals are reflected in the actual performance of the Sentry III and Interface: A.

What, then, is gained from the large size of the Sentry III? Higher efficiency and larger dynamic range. The Sentry III offers 6 dB more efficiency and an additional 3 dB power handling capacity. Not that the Interface: A is any slouch; a pair can produce a sound pressure level of 107 dB (very loud) in an average living room. It's just that the Sentry III can reach 116 dB.

The Interface: A is a vented, equalized system with a low-frequency limit of 32 Hz. The vented Sentry III reaches 40 Hz; the optional equalizer extends its low-frequency limit to 28 Hz.

Interface: A is a home system finding professional application. Sentry III is a studio monitor well suited to home use. Either way, you will find incorporated the latest technology and outstanding performance. Let us send you full information on these systems, plus a list of dealers where they may be auditioned.

Interface: A™/Sentry® III

Electro-Voice® INC ELECTRO-VOICE, INC., Dept. 454RP, 674 Cecil Street, Buchanan, Michigan 49107 a gulton company

From time to time there emerges, under the spur of commercial exploitation, an experimental push in the development of an application of one or another physical phenomena. Hollywood, and the film industry, has always played a major part in bringing to the public demonstrations of unusual phenomena, visual as well as auditory, in the guise of (hopefully) artistically conceived and produced motion pictures.

The 1939 Walt Disney production "Fantasia" was the first commercially successful application of, among many other firsts, multitrack stereophonic sound. The system was called Fantasound, and although at the time the technology was clumsy and limited, the possibilities were recognized and efforts were continued in other studios.

Next came a similar system called Perspectasound, and soon after 3, 4, and finally 6 track magnetic sound tracks for the various wide screen formats which were developed.

Now, from MCA Universal Studios, we have another variation; SENSURROUND, a Hollywoodesque title for a sound system capable of reproducing very very low frequencies at very high sound levels.

EARTHQUAKE'S SENSURROUND

BY WAYNE YENTIS

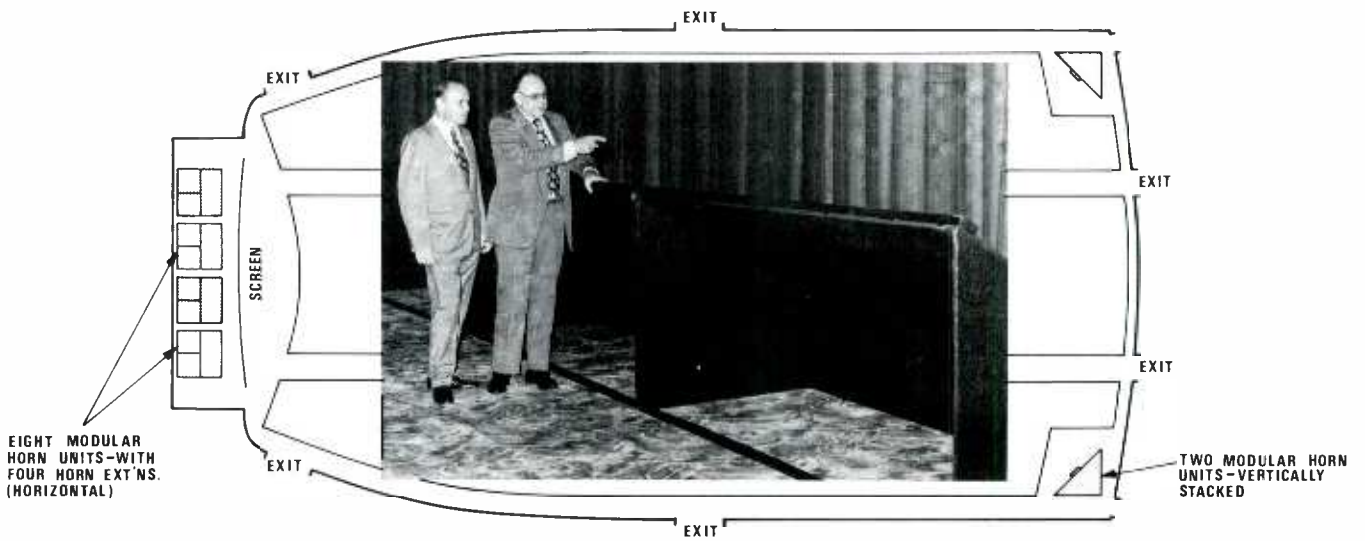
Early in 1974, Universal Pictures began preproduction on EARTHQUAKE, a major and early effort in the current rash of disaster films. Looking for that special something to get people away from their TV's and out to the theaters, the producers (Mark Robson and Jennings Lang) turned to the resources of the Universal Pictures Sound Department, headed by Richard Stumpf, and also called the former head of the department, W.O. Watson, out of retirement. Taking stock of available technology, they suggested using high intensity low frequency sound as the new element in the motion picture experience, and, after some preliminary demonstrations, the producers were sold and development began in earnest.

The low frequency sound was to simulate earthquake rumbles and vibrations, and was to be of such intensity that it could be physically felt in the body as well as heard. To accomplish this obviously required the development of a special sound system which could not only reproduce these frequencies, but which also could be effectively tailored to each of the theaters which desired to use it. The system also had to be producible in quantity, and be reasonably transportable as it was to be shipped to theaters all over the country. Further, the system had to be compatible with the three currently standard formats for film sound; 4 track magnetic sound for 35mm Cinemascope release prints, 6 track magnetic sound for 70mm release prints (for use only outside the US), and the standard 35mm single track optical sound format for the majority of smaller theaters throughout the world.

Early in the project it was realized that none of the existing standard film sound recording techniques could effectively record and reproduce frequencies below 40Hz in the theaters, so it was decided that each installation would be equipped with a low frequency noise generator. Low frequency tones recorded on a special audio track were to control the timing and intensity of the low frequency rumble from the noise generator.

Initially it was thought that the flexibility offered by a control system like this could allow the control of a variety of effects in the theater besides the earthquake rumble. Plans were suggested which included dropping dummies from

continued . . .



Universal's STUMPF and WATSON inspecting a typical modular "M" horn unit, within plan drawing of a typical theatre installation.

OPERATION UPDATE! Add sophisticated, highest-quality **ADM**® components to your console

560S VUE-SCAN

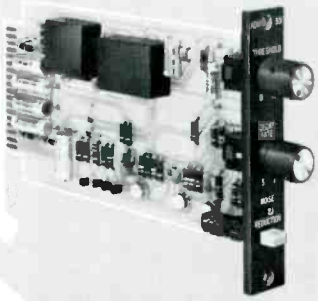
Replaces up to 28 VU meters—eliminating annoying "head swiveling". Displays each analog channel as an easy-to-read illuminated bar graph on a TV-type screen for convenient monitoring. Overmodulated condition of any channel is quickly identified.

TV TYPE MONITOR

Visually displays either the audio spectrum or 28 recording channels on the monitor screen by a simple switching network.

660 SPECTRUM ANALYZER

This Real Time Audio Spectrum Analyzer divides the frequency range of 40 Hz—20 KHz into 28 third octave filters, which are displayed as 28 illuminated bars on a TV monitor screen. Two time constants—averaging or peak and hold. Monitor-compatible with 560S VUE-SCAN.



301 NOISE SUPPRESSOR

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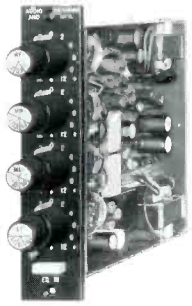
SLIDEX

Unique, noise-free linear attenuator with elements sealed against dirt. No contacts to clean.



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Unity gain equalizer, designed to provide EQ anywhere in an audio system. Four bands and reciprocal functions provide absolute control over the entire frequency spectrum.

ADM offers a selection of professionally engineered components which provide complete flexibility. With them you can update your console to meet the most exacting demands of today's recording operations and provide for future expansion. Write for Technical Bulletins detailing the features of ADM components.

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ADM's NRC Consoles are built to the most exacting quality standards. Unique plug-in modules provide exceptional flexibility for expansion.

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the ceiling in front of the screen at appropriate moments, and letting dust sift down from bags in the ceiling over the audience to give the impression the theater was crumbling. These, and other effects were tried, but in the end the final decision was that they looked too fake to be worth the effort. It was the rumble that was to be the major effect.

Loudspeaker and amplifier manufacturers were invited to participate in the early demonstrations and to submit designs for the necessary equipment. Cerwin-Vega, a manufacturer of professional and hifi sound systems, was involved early in the project and brought over to the Universal lot some of their stock L-48-DD concert bass horns (two 18 inch drivers in a folded 32Hz horn) and two prototype 28Hz corner horns.

The first demonstration runs were made using a General Radio random noise generator with a low pass filter to restrict the bandwidth to below 63Hz.

Even though these demonstrations used horns that rolled off the response below 30Hz, they confirmed that very low frequencies, reproduced at sound pressure levels of about 120dB SPL (C scale, or flat response), did indeed give the illusion of actual physical vibration and movement, as well as being psychologically effective when presented in conjunction with a picture depicting action in an earthquake situation.

Efforts were begun to design a system with a lower cutoff frequency, with a design objective of around 15Hz. Demonstrations were continued using various types of speaker configurations, but after utterly destroying about 60 drivers in a variety of enclosures it was concluded that the only means of effectively radiating the necessary power levels was through the use of horns designed for efficient coupling to the air at the low frequencies desired.

Of course, designing a horn to operate at these frequencies is no mean feat, and for a cutoff frequency of 15Hz theoretically requires the horn mouth to be about 300 square feet in area, and due to the slow taper rate necessary requires a very long horn.

Folding a horn to reduce its length usually results in degrading the frequency response, but only at frequencies well above the cutoff frequency. Because frequencies above 63Hz were to be handled by the normal theater sound system, it was considered practical to fold the horn.

The mouth area problem was reduced by taking into consideration that the required mouth area for optimum coupling depends on the angle into which the horn is required to radiate. If the horn is located in a corner, the walls and floor of the theater form boundaries in the listening space, and, in effect, restrict the angle into which the horn radiates. The principal effects of too small a mouth are some attenuation of response in the region immediately above cutoff, and a periodic fluctuation with frequency of the reflection coefficient at the horn mouth. Here the nature of the signal to be reproduced permits a reduction in mouth area, since dips in response are much less noticeable with noise than with continuous tones, especially in the narrow (2 octave) band of low frequencies of interest.

Three configurations of horns were developed for use with SENSURROUND, all using the special drivers developed for the purpose by Cerwin-Vega. Special consideration was given in all the horn designs to prevent destructively large driver excursions which occurred at and below the cutoff frequency.

The first is a relatively conventional W horn design, two drivers operating into a double folded horn. It is large and bulky, and therefore difficult to transport, but it finds application in theaters where requirements demand a low profile horn. It measures 80"W x 40"H x 60"D and weighs in at 450 pounds. See Figure 1.

The second design, called the model C horn, is a vertical corner horn for use in theaters where the height is restricted, such as under low balconies at the rear of the theater, or over an exit in the corner of the theater. It is 76"H x 60"D x 24"W, is driven by two Earthquake drivers, and because there are no internal folds as in the W Horn, must be operated into a

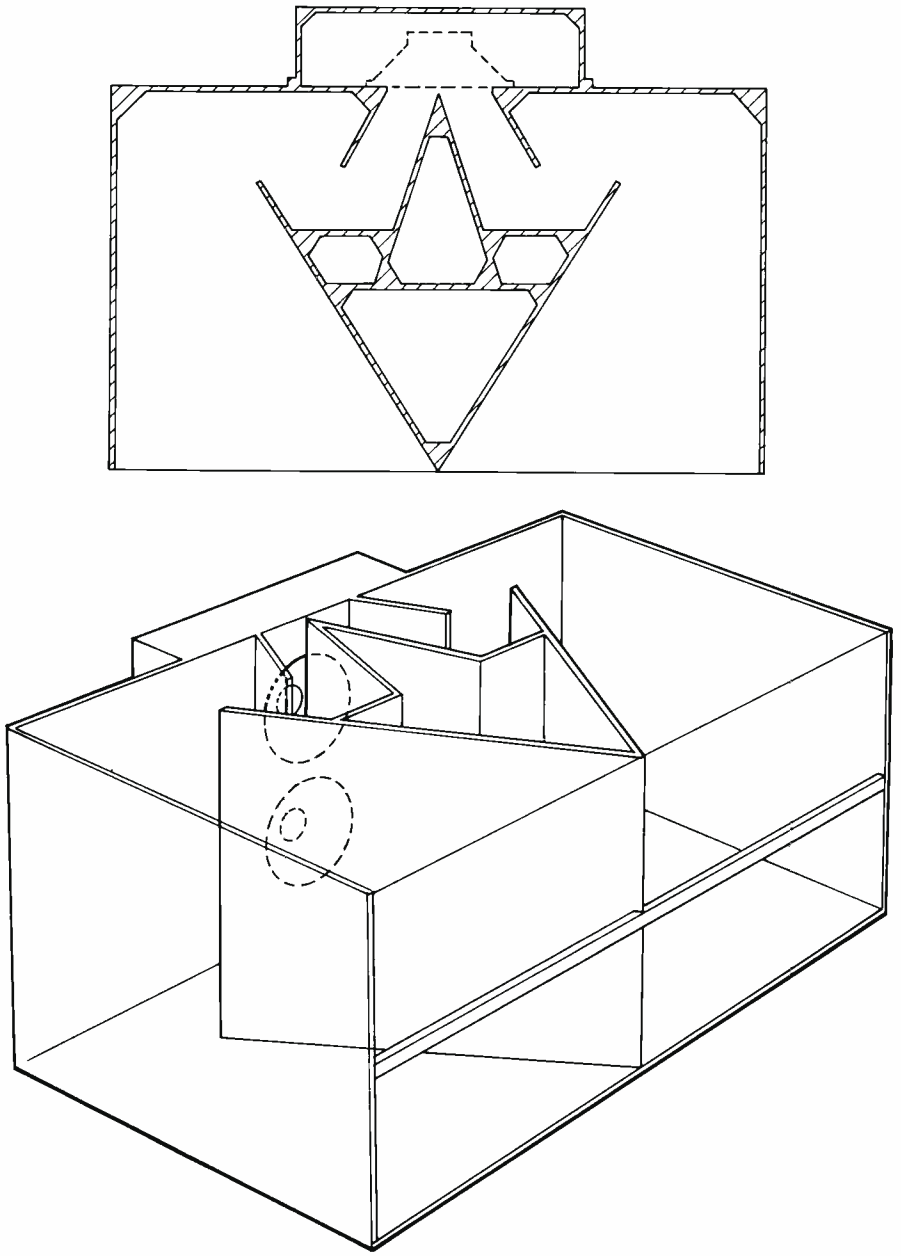

Figure 1 "W" Horn, Cross Section and X-Ray View

EARTHQUAKE . . . "SENSURROUND"

corner, as well as requiring the addition of top and bottom horn extenders. In some installations, a corner has to be specially built around the horn, if the actual corners in the theater are not usable. See Figure 2.

The third design, called the M horn, or modular horn, was the end result of design efforts to create a widely applicable and easy to manufacture and transport horn assembly. It is in a package four feet square and 20 inches wide, and was designed to operate into a corner much like a C horn, only in multiples, stacking up to 4 modules high. See Figure 3. Alternatively, where a suitable corner is not available in the theater, one can be built around the stack, as with the C horn. Or wooden mouth extenders can be attached and the units laid end to end on the floor as in Figure 4, or against a wall. This arrangement utilizes the floor or wall as part of the horn extension, and is effective but not as efficient as operating into a corner.

Each M horn contains only one Earthquake driver, and by operating the driver into a tightly constricted throat, the compression loading raises the efficiency of the system and reduces the driver excursion required for a given SPL.

High compression loading does result in increased distortion, generated by the slightly nonlinear relationship between pressure and volume in air. This distortion

. . . *continued on page 27*

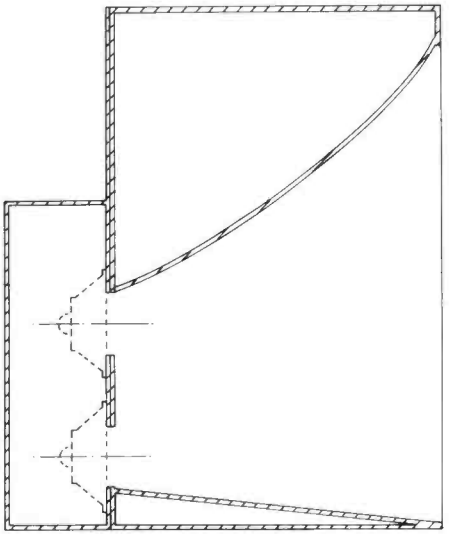

Figure 2 "C" Horn, Cross Section and X-Ray View showing Operation into a Corner

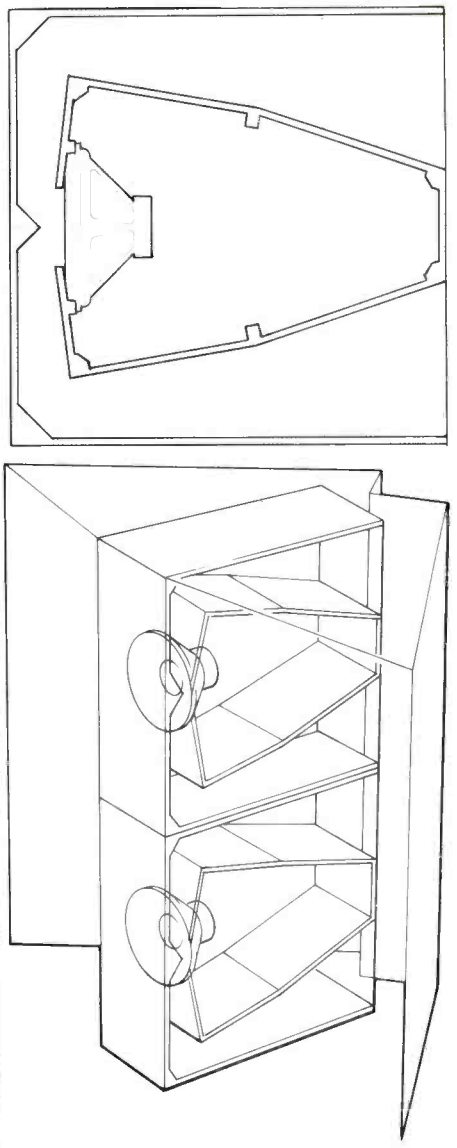

Figure 3 "M" Horn, Cross Section and X-Ray View showing Operation into a Corner

"C" Horn installed over an exit

Figure 4 - "M" Horns with Mouth Extenders, on floor in front of screen

SHURE®

INTRODUCES SR

PROFESSIONAL SOUND REINFORCEMENT EQUIPMENT

audio console

electronic crossover

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speaker systems

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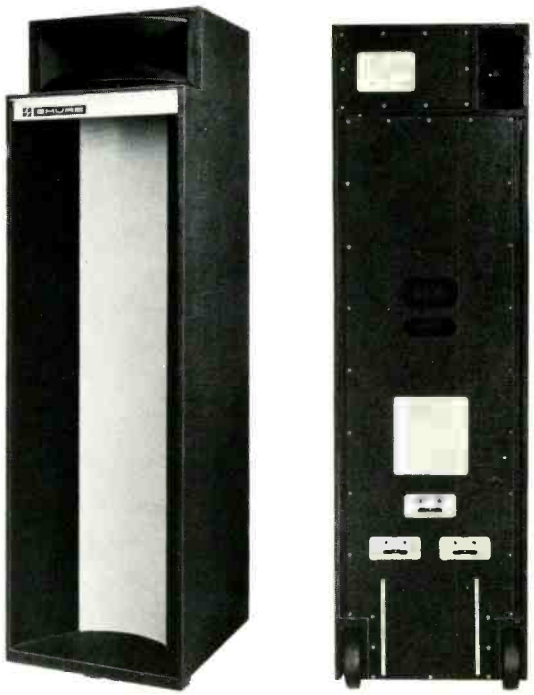
SR108 EXTENDED RANGE SPEAKER SYSTEM



The Shure SR108 is an extended range, two-way speaker system designed for high sound-pressure-level reproduction of wide frequency range program material in sound reinforcement applications. The speaker system utilizes six eight-inch cone-type speakers with a total speaker cone area of 1097 cm^2 (170 in^2) and four high frequency drivers. The SR108's pressure sensitivity has an EIA rating of 54 dB at 9.2 m (30 feet) from 1 milliwatt (equivalent to 102 dB at 1.2 m (4 feet) with a 1-watt input).

SR106 ELECTRONIC CROSSOVER

The Shure SR106 Electronic Crossover is a rack-mountable, unity gain, selectable-frequency dividing network, designed for use with two- or three-way speaker systems such as the Shure SR108 Extended Range Speaker System in high-quality sound systems. It utilizes the principle of biamplification to separate an audio console or mixer-preamplifier output into two frequency bands for distribution to separate power amplifiers. In this manner, the advantages of lower distortion, increased high-frequency power, wider dynamic range, and maximum efficiency are obtained.

The SR106 provides switch selected crossover frequencies of 500 Hz, 800 Hz and 2600 Hz. It can be used to provide two output frequency bands or, with a second SR106, three output frequency bands, with each output routed to a separate power amplifier (tri-amplification).

The SR106 is supplied with a protective power switch cover, and rack-mounting screws for mounting in standard 19 in. (483 mm) audio equipment racks or in optional carrying cases.

The SR106 measures 44.5 mm (1¾") high x 483 mm (19") wide x 216 mm (8½") deep, and weighs 3 kg (6 lb, 8.8 oz).

INPUT CAPABILITIES

Balanced bridging input accepts balanced line level or unbalanced signals through parallel three-pin female professional audio and three-circuit ¼-inch phone jack connectors.

The sound power distribution is nominally uniform over a 140° angle in the horizontal plane and a 90° angle in the vertical plane. The speaker system's bass reflex design provides extreme low-frequency enhancement, with low-frequency speakers column-mounted in a horn-loaded, front-ported enclosure. The four high-frequency drivers are coupled to a single radial horn.

The uniform response is peak-free from 40 Hz to 15,000 Hz when driven by a constant-voltage amplifier and radiating into an acoustical half-space.

INPUT CAPABILITIES

- Continuous 100-watt maximum power rating (40-volt source). 16-ohm nominal impedance in single-amplifier operation.
- Accepts up to 100 watts of program material in the low-frequency section and up to 30 watts in the high-frequency section during biamplified operation.
- Inputs consist of three pairs of parallel-wired phone jacks for (1) conventional full-range operation, (2) low-frequency biamplified operation, and (3) high-frequency biamplified operation.

CONTROLS

- Rear-panel Loudspeaker Operation switch provides four high-frequency level positions of −4, −2, 0 and +2 dB for conventional full-range operation and a BIAMP position for biamplified operation.

The SR108 Speaker Cabinet is ruggedly constructed of wood measuring 15.9 mm (⅝"). It is covered with scuff-resistant vinyl paint, has rear-panel rails and handle, two wheels and a cable storage compartment. The radial horn is made of high-density, structural urethane foam. A plug-in 15.2 m (50-foot) speaker cable is included. The SR108 measures 1730 mm high, 495 mm wide, and 517 mm deep (68⅛" x 19½" x 20⅜") and weighs 64.5 kg (142 lb), including speaker cable. Operating temperature range is from −7°C to 43°C (20°F to 110°F), with storage temperatures from −29°C to 71°C (−20°F to 160°F).

OUTPUT CAPABILITIES

- Outputs are +18 dB m, 600-ohm, balanced, line level with greater than 110 dB signal-to-noise ratio.
- Provides parallel three-pin male professional audio and three-circuit phone jack connectors for high and low frequency line level output.

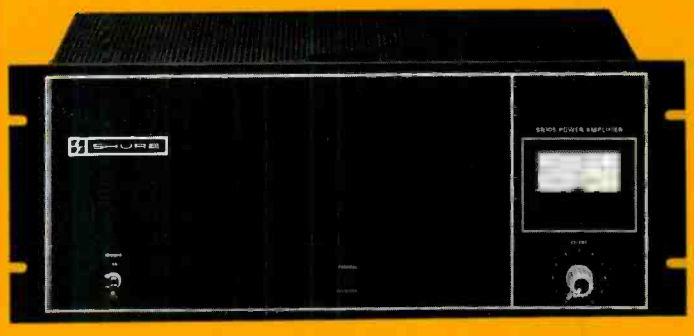
CONTROLS

- The SR106 Electronic Crossover incorporates a power On-Off switch and indicator lamp.
- A three-position Crossover Frequency switch adjusts to select crossover frequencies of 500, 800 or 2,600 Hz.

OPTIONAL ACCESSORIES

- May be used with Model A30A or, A105A Carrying Case. (See back page.)

SR105 POWER AMPLIFIER



The Shure SR105 is a high-quality, high-power, rack-mountable amplifier designed for use with the SR101 Audio Console and most other associated equipment used in professional sound reinforcement systems. It is available in two models: The Model SR105A, which provides both direct- and transformer-coupled, constant-voltage 70-volt output, and the Model SR105B, which permits direct speaker coupling only. Both units measure 178 mm (7") high x 483 mm (19") wide x 270 mm (10⅝") deep. The SR105A weighs 15.66 kg (34½ lb); the SR105B weighs 12.23 kg (27 lb).

Each SR105 amplifier is capable of delivering 200 watts rms to a four-ohm load, and may be interconnected with additional units to provide the greater sound power required by especially large coverage requirements. Harmonic and intermodulation distortion are low, and frequency response is flat, 20-20,000 Hz, ±1.5 dB.

Both the SR105A and SR105B are completely protected against short or open circuit loads, and can operate at ambient temperatures up to 42°C (110°F.) without derating.

INPUT CAPABILITIES

- Accepts line level or auxiliary level signals through a professional three-pin balanced bridging input connector and dual, unbalanced, paralleled phone jacks, for compatibility with all Shure audio control components and virtually all professional consoles.

OUTPUT CAPABILITIES

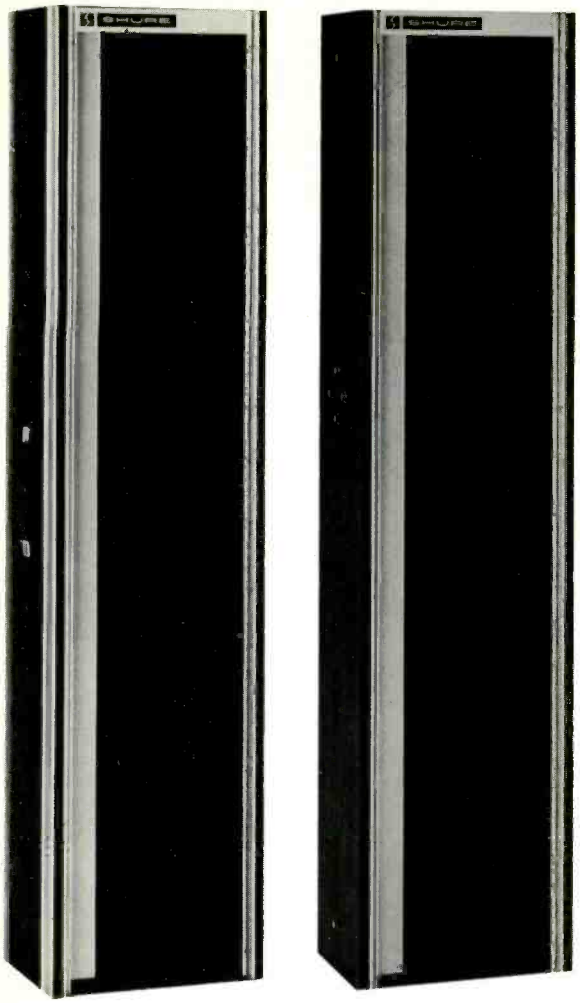
- Provides four direct-coupling output jacks for connection to speaker systems, along with a two-screw terminal strip for additional direct speaker coupling.
- Model SR105A, with 70-volt output provision, delivers 150 watts rms output to 70-volt distributed speaker systems.

CONTROLS AND METER

- Both SR105 units incorporate a power On-Off switch and a volume control.
- Accurate, front-panel output voltage meter indicates output voltage as a percentage of 100% maximum voltage.
- Thermal overload indicator lamp is provided to indicate if thermal cycling has occurred due to improper ventilation or operation.

OPTIONAL ACCESSORIES

- Both the SR105A and SR105B may be used with the Model A105A Carrying Case. (See back page.)



SR102 & SR103 SPEAKER COLUMNS

The Shure Models SR102 and SR103 Speaker Columns were developed especially for use with high-power amplifiers such as the Shure Model SR105 in system installations requiring very high quality sound reproduction. Both models offer wide frequency response (100 to 15,000 Hz), very low distortion, and *exceptional* Penetrating Power.

- Both models are 16 ohms, and are rated at 100 watts maximum program material.
- Two 10-inch and four 8-inch heavy-duty speakers, along with twin high-frequency speakers that function as both dome radiators and acoustic horn radiators, are used in both models.
- Both models employ an enclosure with a specially tuned rear port that heightens Penetrating Power by producing a highly directional characteristic (dispersion is 140° horizontally, 65° vertically).
- Both models produce optimum auditorium penetration, with maximum feedback reduction.
- Both the SR102 and the SR103 may be used with either direct (4 to 16 ohms) or constant-voltage (25 or 70 volts) amplifier outputs without matching transformers.

MODEL SR102: The SR102 is a portable speaker column, and is intended primarily for temporary installations. It measures 1522 mm (59 15/16") high x 354 mm (13 15/16") wide x 241 mm (9 1/2") deep, and weighs 32.62 kg (72 lb). Its enclosure is fitted with anodized solid aluminum siderails for protection, and with a strong retractile handle for ease in transportation. The enclosure is covered with a tough, scratch-resistant, moisture-proof vinyl fabric. Connections are made through a six-screw terminal strip, or through twin, parallel-wired phone jacks. A 15.2m (50-foot) rubber-jacketed connecting cable with locking phone plugs is supplied.

MODEL SR103: The SR103 is intended for permanent installations, and may be used outdoors for extended periods without adverse effects. All electrical components, hardware and enclosed surfaces have been designed for maximum resistance to adverse weather conditions. Adhesives used are moisture-resistant, and all trim and fastening hardware are highly corrosion-resistant. Drain holes are provided to minimize moisture accumulation. Connections are made through a six-screw terminal strip. It measures 1513 mm (59 9/16") high x 352 mm (13 7/8") wide x 240 mm (9 7/16") deep, and weighs 31.8 kg (70 lb). Designed for use with Model A103A Wall-Mount Speaker Bracket shown below.

CUSTOM-DESIGNED SR ACCESSORIES

Model A101B
Panel Lamp

A small, low-intensity light unit to illuminate SR101 Audio Console controls in dimly lit areas. Attaches quickly to front panel with screw-down connector, and draws its power from the SR101. Flexible neck affords variable positioning.

Model A102A
70-Volt Transformer

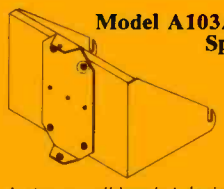
A high-efficiency, low-loss autotransformer that mounts easily on back of a speaker enclosure to provide wattage taps of 50, 25, 12, and 6 watts in 70-volt distributed systems, and impedance taps of 8 and 16 ohms to accommodate a wide variety of speakers. Five-screw 70-volt terminal strip and three-screw impedance terminal strip.

Model A101A
Carrying Case

Designed to give complete portability to the SR101 Audio Console. Tough, weather-resistant outer covering and solidly attached carrying handle. Rear panel removes to provide a padded operator armrest. Contains space for carrying A101B Panel Lamp and operator cue sheets. The A101A measures 370 mm (14 9/16") high x 518 mm (20 3/8") wide x 256 mm (10 1/16") deep overall.

Model A30A Carrying Case

Similar to A105A above with 3 1/2" (89 mm) of rack-mounting. (Not shown) "for SR106"



Model A103A Wall-Mount
Speaker Bracket

A strong wall bracket designed especially for permanently mounting the SR103 Speaker Column on almost any vertical surface. Allows the installer to aim the speaker for optimum audience coverage. Tilts as much as 19° left or right, and 18° up-tilt or 12° down-tilt, then locks in selected position.

Model A105A
Carrying Case

A sturdy, weather-resistant case designed for easy transportation of either SR105A or SR105B. Front-and-back hinged removable panels permit access to front and back of unit for operation. May also be used to create a portable, "custom" stack of smaller audio control components (such as the Shure M67 Mixer and M610 Feedback Controller with appropriate rack panel kits). The A105A measures 533 mm (21") high x 232 mm (9 1/4") wide x 311 mm (12 1/4") deep overall. Provides 178 mm (7") of rack-mounting space (such as the SR106 or controller with kits). Weighs 6 kg (13 1/2 lb).

Model A3S-T
Speaker Tilt Stand

A tubular steel stand that provides added stability for SR102 Portable Speaker Column, and permits an upward tilt of up to 30°.

Model A50XC

15.2 m (50-ft.) speaker Extension Cable. (Not shown)

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continued from page 22 . . .

gets worse as the amount of compression loading increases. The small throat area required to achieve compression loading also results in increasing the horn length for a given flare rate and mouth area. The final design of all the horns is a balance of enough loading to prevent bottoming of the driver suspension with an acceptable distortion level and compact horn length.

The driver units are relatively standard dynamic speakers, except for their large size (18" diameter) and extremely robust construction. The high power levels and low frequency requirements excluded using any commercially available speakers. Cerwin-Vega had already earned a name for speakers that can handle very high power levels, and in fact before working on the SENSURROUND project they had developed voice coils capable of dissipating 1kW continuously without burning out.

The voice coils used in the Earthquake drivers are wound on heat conducting aluminum bobbins which have a row of holes around the circumference which effectively pumps cooling air around the voice coil during operation. The coil is bonded to the aluminum bobbin by a specially formulated epoxy material.

In order to achieve critical damping at resonance (around 20Hz), the magnetic field strength and gap width were adjusted to achieve a somewhat lower field density than that possible with todays materials. A measure of loudspeaker motor efficiency is a quantity called the *Bl product*, which is obtained by multiplying the magnetic flux density (B) in the voice coil gap (in Webers/m^2) by the length (l) of voice coil wire in the gap. The result is an expression of the driving force (in Newtons) produced per ampere of voice coil current. The Earthquake voice coil is of sufficient diameter and number of turns to achieve a Bl product of 44 N/A, or about 10 lbs/ampere, compared to about 5 to 10 N/A for a typical low efficiency woofer.

But combining high power capability and the long excursions necessary at these frequencies was another problem. It wasn't difficult to design the voice coil to accommodate a long swing, but cones, surrounds and spiders needed considerable improvement. After much experimentation a combination of cone and suspension materials and polymer treatments was found that yielded adequate excursion capability and an acceptably low failure rate. Universal has specified that the driver must be capable of 1½ inch peak to peak excursions during the 20Hz, 300 watt free air burn-in during final testing of the driver units.

Universal turned to several manufacturers to supply the high power amplifiers. At the time work was beginning on the SENSURROUND project there were very few amplifiers available that could deliver more than 300 watts. Cerwin-Vega was in limited production with their A3000 stereo amplifiers, at 700 watts per channel into 4 ohms, and BGW was building their model F50 at 300 watts per channel into 4 ohms, with the capability of delivering 600 watts monaurally into 8 ohms when operated in the bridge configuration.

Because the drivers are being operated at and near resonance in the horn enclosures, their loading effect on the driving amplifiers becomes highly reactive. With the massive magnets and voice coils used in the Earthquake drivers, heavy reactive currents are driven back into the amplifier output stages. This, if nothing else, will falsely trip most current limiting protection circuits and cause loud chirps, buzzes and squeaks very unlike an earthquake.

The initial tests and demonstrations were conducted using BGW model 750 amplifiers, connected in bridge, and with the current limiting protection circuits removed. The results were acceptable, and in fact led BGW into the production of their model 750A which was designed without protection circuitry. Instead, it is built to handle all the energy the power supply can deliver, for short periods, and has a fast acting magnetic circuit breaker to remove power from the amplifier under short circuit conditions, as well as crowbar circuitry to discharge the power supply capacitors.

Both the Cerwin-Vega A3000 and the BGW 750A proved adequate for the job and for the initial waves of installations both amplifiers were used. Cerwin-Vega at the time was preparing for production a smaller amplifier, the model A1800, capable of delivering 350 watts per channel into 4 ohms. Now that this amplifier is in full production, all new installations of SENSURROUND will be using it.

It was suggested early in the project that a full bandwidth amplifier was simply not necessary to reproduce the extremely limited bandwidth of the earthquake rumble. It would not be difficult to design an inexpensive amplifier with tremendous power, using low cost, low speed power transistors. High speed devices, necessary for full power response at high frequencies, besides being expensive, are sometimes fragile and require more circuitry to compensate for high frequency instabilities. But the people at Universal wanted a full range general purpose high power amplifier, perhaps with the idea of extending the range of SENSURROUND effects.

The film was released in the three standard formats mentioned before; 6 track magnetic sound on 70mm film (blown up from 35mm), 4 track magnetic sound on 35mm film, and the old standard single track optical sound on 35mm. A system had to be devised that would enable the SENSURROUND effects to be interfaced to all of these formats, as well as allowing the film to be shown on standard theater sound equipment if a particular house didn't want to install SENSURROUND. Of course, listening to the earthquake rumble on a standard theater sound system hardly has the punch of SENSURROUND, but it was decided that smaller theaters should be able to show the film even if they didn't want the system.

But then the term *standard* theater sound equipment hardly has any meaning; old movie houses (and most of them are) usually have old equipment, which over the years has suffered possibly several modifications to update performance, and which many times is undocumented. Circumstances like these make it difficult to design a system that can be snapped in anywhere and work.

W.O. Watson, an old hand in the movie business, called on many of his colleagues of the past, enlisting the services of other retired movie men. Their expertise was invaluable in the field for determining the equipment and installation variables as they exist today, and their suggestions

RICHARD STUMPF
With a typical control box installation

provided solutions to most of the peculiar problems that developed in the field.

The system is not simple, yet it has to be easily installed and easy to operate. Very special attention was taken to minimize bother to the projectionist, who is sometimes known to become irritable and moody when his regular routine is broken.

The final embodiment of the SENSURROUND system consists of the horns required for a given theater, the amplifiers to drive them, and a special control box which is patched into the theater sound system between the projector changeover switch and the audio power amplifiers. See Figure 5.

This control box is the brains of the system and has sufficient trimming capability to accommodate a wide variety of input and output levels and impedances. It also houses the special digital random noise generator and filters which generate the earthquake rumble effect. This rumble effect is turned on and off and otherwise controlled by circuits which are tuned to control tones recorded on an audio track on the release print. The operating principles are similar to those used in the 40's with Perspectasound and Fantasound, both early attempts to create pseudo stereo effects in the theater by steering the signal to different speakers in the house by means of sub-audio control tones on the audio track. See Figure 6.

SENSURROUND, as used with the film EARTHQUAKE, uses two control tones; one at 25Hz which controls the level of the rumble effect, and another at 35Hz which is used to inject certain effects on the normal sound tracks (crashes, explosions, bursting dam, etc.) into the SENSURROUND system for added impact. These control tones are recorded at a maximum level of 20dB below 100% modulation, and have an

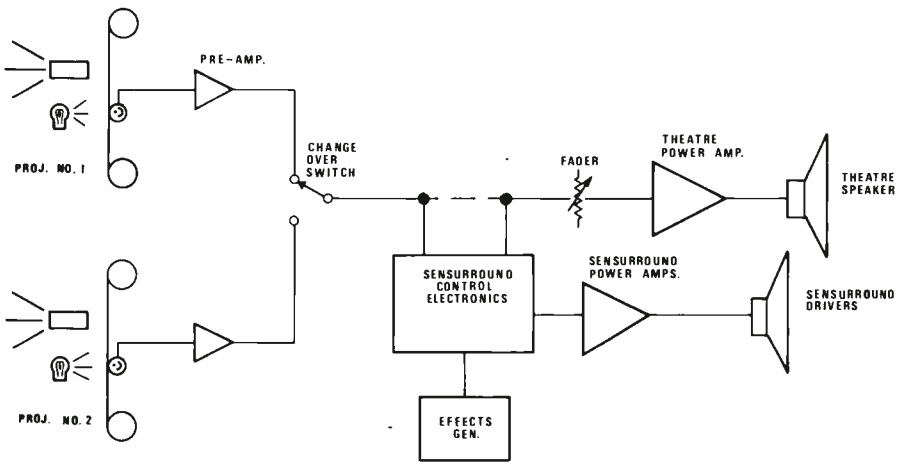
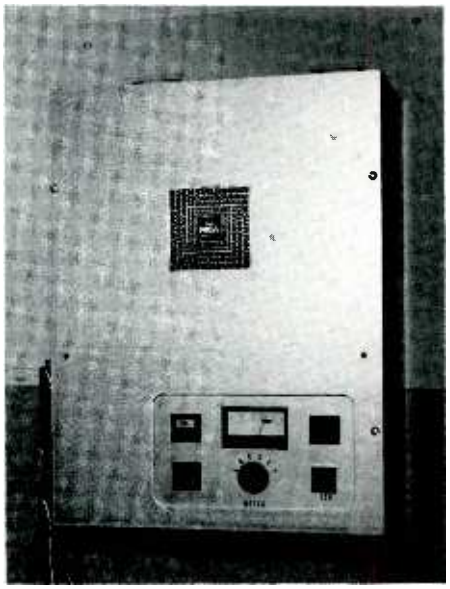


Figure 5 - Installation of SENSURROUND in a typical theater projection system (Optical Soundtrack Release Print)



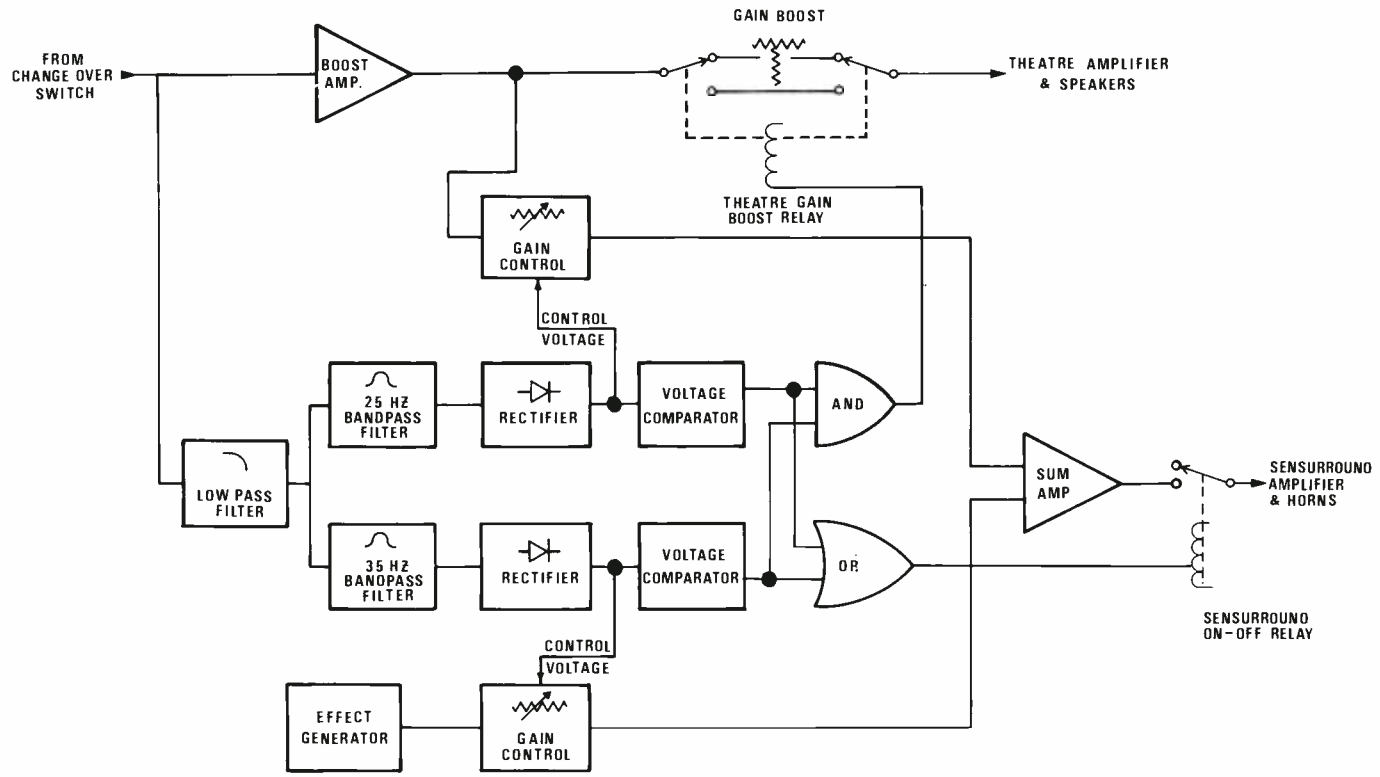


Figure 6 - Block Diagram: SENSURROUND Transmission & Control System

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analog control range of 10dB. They are sorted out from each other by tuneable active filters in the control box.

Because these control frequencies are really not very far apart, these filters are of necessity of a relatively high Q design to enhance selectivity. This, along with the fact that the control frequencies are low, creates a filter with a rather sluggish response time, both in attack and decay. This effect had to be minimized, obviously, so the Q was reduced somewhat from the original design and the selectivity was augmented by putting dip filters in each control circuit which scrub out the adjacent control frequency. Some lag in response time remains, however, and in recording the control tones the studio dubbing mixers had to anticipate the action in the picture, turning the control tones on and off a little in advance of the corresponding action on the screen.

For the 35mm film formats, the control tones are recorded on an optical sound track. The magnetic oxide stripes on a 35mm 4 track magnetic sound print are applied as shown in Figure 7. The space normally occupied by an optical track on 35mm film is left partially exposed, and it is this remainder of optical track that is reserved exclusively for the control tones. Most magnetic sound projectors are also equipped with an optical sound head which is used to reproduce the control tones. These machines may or may not have the necessary preamp installed to use the optical head, so a suitable preamp is included in the control box to be used if necessary. Figure 8 is a block diagram of a typical 4 track installation.

In the case of single track optical sound release prints, the control tones are mixed in with the normal sound track material. The program material is rolled off below 53Hz during the optical sound transfer process so there is plenty of separation between the sound track and the control tones. Figure 5 is a block diagram of a typical optical track installation.

The 70mm release prints for engagements outside the US are handled in a slightly different manner. Four magnetic stripes are applied to the print as in Figure 9, with the two wide outer stripes carrying two audio tracks each. The two narrower inner tracks are used for control tones, and, as they are not sharing the same track nor are they mixed in with the audio tracks, 100Hz tones are used on both tracks, one being for the rumble generator and the other for the effect steering. This leaves only 4 tracks for the regular sound tracks, which are distributed as with a 35mm 4 track print; left, center, right, and surround (or rear) tracks.

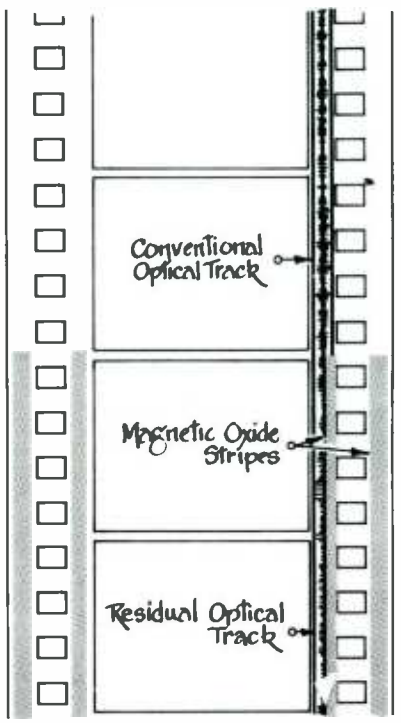


Figure 7 Magnetic and Optical Tracks on 35mm Release Prints

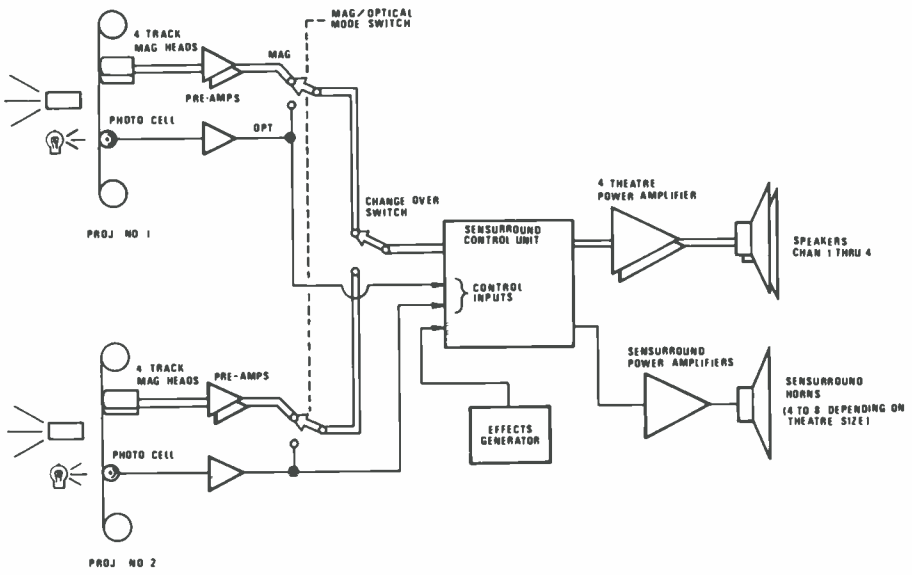


Figure 8 - SENSURROUND as used with 4 Track Magnetic Release Prints

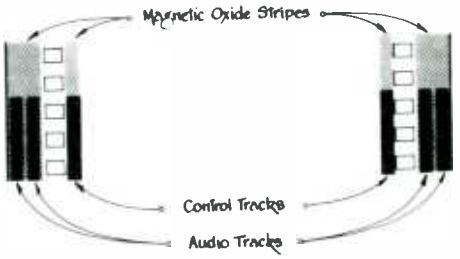


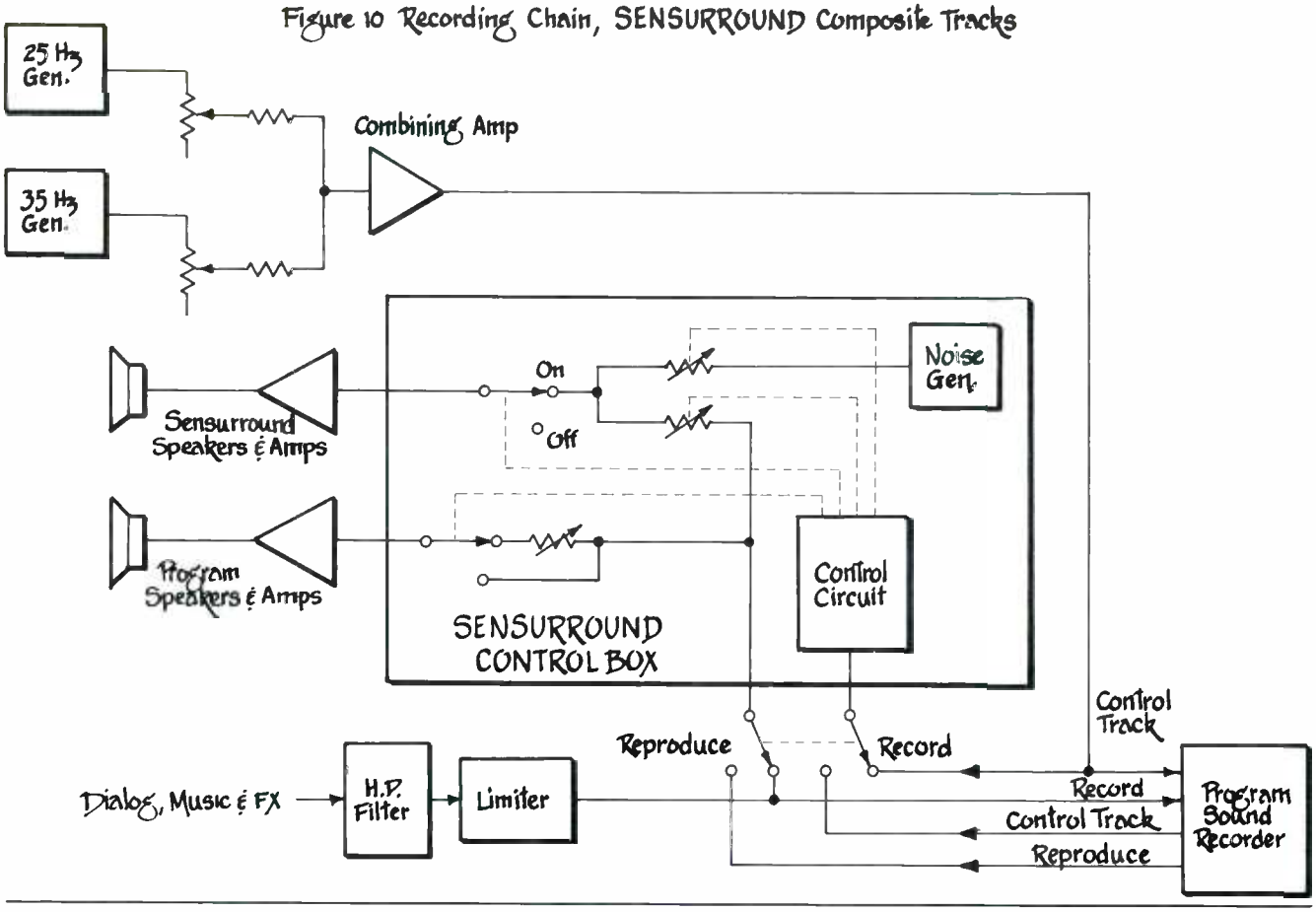
Figure 9 Magnetic Sound Tracks on 70mm Release Prints

Projector speed determines the frequency of the control tones in the theaters, and of course the machines in theaters around the country vary as much as 10% from the standard 24 frames per second. It is critical that both projectors in a given house operate at the same speed, but an overall speed variance can be compensated for by adjustments in the control box which tune the filters. A special test film is supplied with the system for use in aligning the system to the theater equipment.

The preparation of the composite soundtrack was fairly straightforward, and the block diagram of Figure 10 depicts the basic setup. As mentioned, the only thing out of the ordinary involved the 200 millisecond lag in response of the control tone filters which had to be compensated for by anticipating the times for starting and stopping the control tones. This was the job of the dubbing mixers, and at times required more than a little practice. Especially with the 35Hz tone which controlled the injection of effects from the normal track into the SENSURROUND system. Some of these effects were very short; crashes and bangs, etc., and getting them in and out right took some time. The effects that were steered into the SENSURROUND system were chosen with care; voices and music were kept out of the system because they tended to sound muffled and out of balance when reproduced over the horns.

It was found during the development of the system that when the earthquake rumble was activated, the regular sound track seemed to diminish in intensity due to the high level of rumble. Circuits were included in the control box which automatically increase the level of the normal sound tracks by a nominal 6dB when the rumble and effects tones are activated, thereby maintaining a good subjective balance. See Figure 11.

Figure 10 Recording Chain, SENSURROUND Composite Tracks



CONTROL SIGNALS		THEATRE PROGRAM CHANNEL	SENSURROUND CHANNEL		
25 Hz	35 Hz	STEP GAIN	EFFECTS ON	PROGRAM GAIN CONTROL	EFFECTS GAIN CONTROL
0	0	0 db	0	- ∞ db	- ∞ db
1	0	0 db	1	variable - ∞ to 0 db	- ∞ db
0	1	0 db	1	- ∞ db	variable - ∞ to 0 db
1	1	+6 db	1	variable - ∞ to 0 db	variable - ∞ to 0 db

Figure 11 - SENSURROUND Control Signal Functions: TRUTH TABLE

Besides the rumble from the noise generator in the control box, some rumble is recorded on the normal sound tracks in all release formats. The sound quality of this rumble is of course subject to the limitations of the theater system.

Early in the development of the system it was discovered that when the rumble signal was driven into clipping there was no appreciable difference in the sound. However, when recording the rumble onto a variable area optical track (see Figure 12), care had to be taken not to hit or exceed 100% modulation (optical clipping), otherwise the saturated track blanked out the control tones necessary for a SENSURROUND showing. This problem was alleviated by clipping the rumble signal 2dB below 100% modulation during the recording of the op ical track, as shown in Figure 13. The rumble was clipped rather than simply limited because clipping gives the impression of increased level. However the clipping generates cross-modulation products which mysteriously activated the control tone detection circuits. It was only after

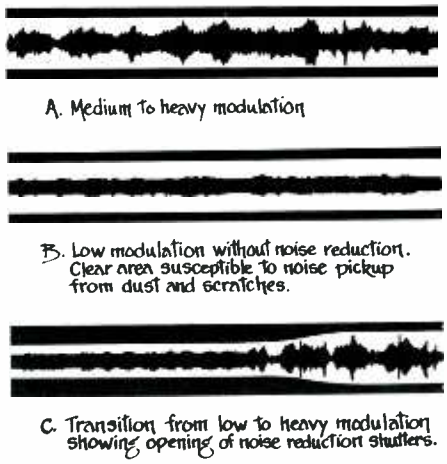


Figure 12 Variable Area Sound Tracks

studying the waveforms with a spectrum analyzer that it was found that the clipped rumble waveform contained components in the same frequency range as the control tones. Additional dip filters were placed in the optical recording chain, just after the clipper.

This clipping also caused problems in the optical recording system, which normally cannot maintain the flat crests of a clipped waveform. For this reason a predistorting circuit (phase correction filter) is introduced in the chain just after the clipper which conditions the signal so it will be recorded with flat crests.

The 25 and 35Hz control tones require some level adjusting coming from the prerecorded control track into the optical re-recording system. This is because the optical system has a built-in low frequency rolloff which must be compensated for linear transfer of the control tones. This is accomplished in the warp filter at the control input to the system.

Another area requiring special attention was in the handling of the noise reduction process commonly used in optical sound track recording. The most common type of optical sound track found today is the variable area track. During periods of low modulation, or low levels, this type of soundtrack would appear as in Figure 12b, if no noise reduction was employed. Dirt, scratches, and other particles are plainly visible in the clear area of the track during quiet passages and show up as noise in the speakers. With the addition of noise reduction shutters in the optical recording system, this clear area of the track is merely closed during periods of low or no modulation, and opens up when periods of heavier modulation occur. See Figure 12c. The action is much like an audio noise gate. The problem encountered with SENSURROUND concerned the attack time in opening the noise reduction shutters. The standard attack time, usually between 15 and 30 milliseconds, was so quick that it caused low frequency thump which excited the control tone detectors, causing spurious response of the effects generator. The attack times were consequently lengthened to eliminate the thump. This in turn created another difficulty; that of opening the noise reduction shutters in advance of the control signals. This was solved by

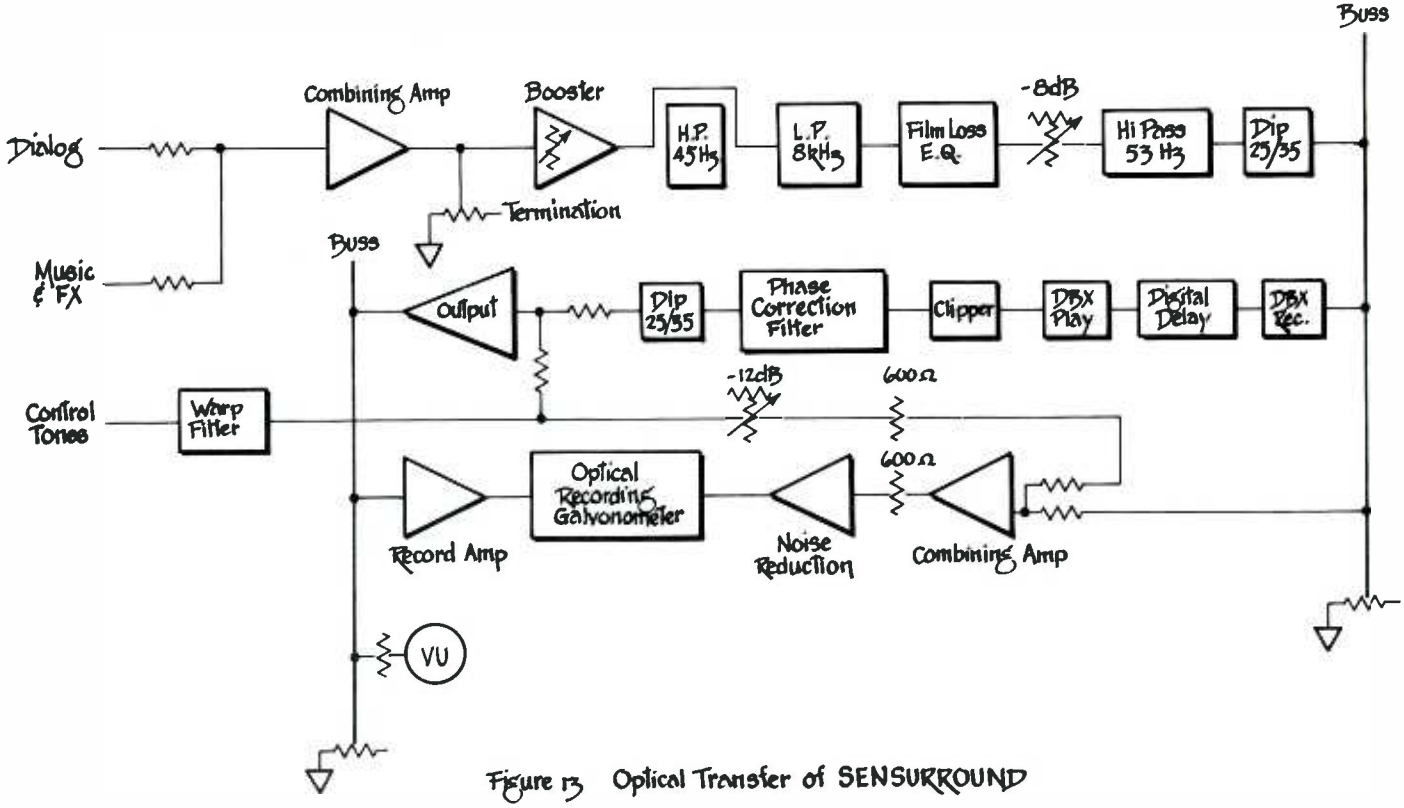


Figure 13 Optical Transfer of SENSURROUND

introducing a digital time delay unit in the audio chain, as also shown in Figure 13, which allowed time for the shutters to open slowly before the beginning of a heavily modulated segment.

When the initial plans were being laid, it was estimated that by the end of the first year of operations there would be 17 theaters equipped with SENSURROUND. That was a very early estimate because since the opening engagements in November '74 there are nearly 400 systems in operation in theaters all over the world, with the majority of them here in the US. Installing and maintaining that amount of equipment in widely dispersed locations throughout the country required the services of a nationwide organization, and the service contract was awarded to RCA. Regional RCA service station personnel were trained by Universal specialists, and they in turn have gone into the field to train local RCA people. Now they are handling most of the installations and maintenance calls. In addition, as the film finishes an engagement in a theater, RCA handles the transfer and installation of equipment into new locations.

A comprehensive operations and service manual was produced as a result of the efforts of the component manufacturers, the studio departments involved, and the people in the field who put the first systems into operation. The manual includes not only installation and calibration instructions, but includes procedures for surveying a theater both before and after installation in terms of structural soundness and possible damage, horn requirements, existing equipment requirements, and sound level measurements.

The RCA service men work closely with the theater managers in evaluating the theater and installing the equipment, and careful sound pressure level measurements are taken once the installation is completed. A model 450F Sound Pressure Level meter from Scott Instrument Labs is supplied to the RCA personnel for this purpose, and the system is adjusted until an overall SPL of 95dB (A scale) in the center of the theater is achieved, with no more than 110dB (C scale) 4 feet in front of any horn. These levels are deemed safe (at the frequencies involved) for human consumption for periods up to 8 hours, and there are only about 7 total minutes of earthquake rumble in the film. According to Watson, there was only one person up-tight enough about the sound level to complain to the health department, but he neglected to leave his name. No cases of structural damage have been reported.

That the project has been successful there is no doubt. Not only has the demand from theaters all over the world greatly exceeded the initial estimates, but the film is drawing record attendances from individual theaters. Mann's Chinese, in Hollywood, grossed $1.3 million in paid admissions in 5 months. The film has also won two Acadamy Awards for technical achievement, one of them for the SENSURROUND system.

For the next couple of years we can probably expect SENSURROUND films to bombard us with a variety of tooth rattling booms, blasts, rumbles and roars. Perhaps further down the line subtler applications will be tried. MCA Universal is certainly looking for continued use of the system and the suitability for a variety of special effects is evident. The world we live in is saturated with very low frequency sounds, sounds we hear or perceive only subliminally, but which give us definite clues to our awareness of reality. These sounds are found in the distant rumble of traffic, the sigh of wind in the trees, even the impact of a closing door. In recording these sounds the low frequency content is usually filtered out because it is troublesome to the recording process . . . but troublesome only because of the difficulty in reproducing it with proper balance. SENSURROUND, or at least some of the equipment developed for it, can be marked as a significant technological step towards extending the potential of the listening experience.

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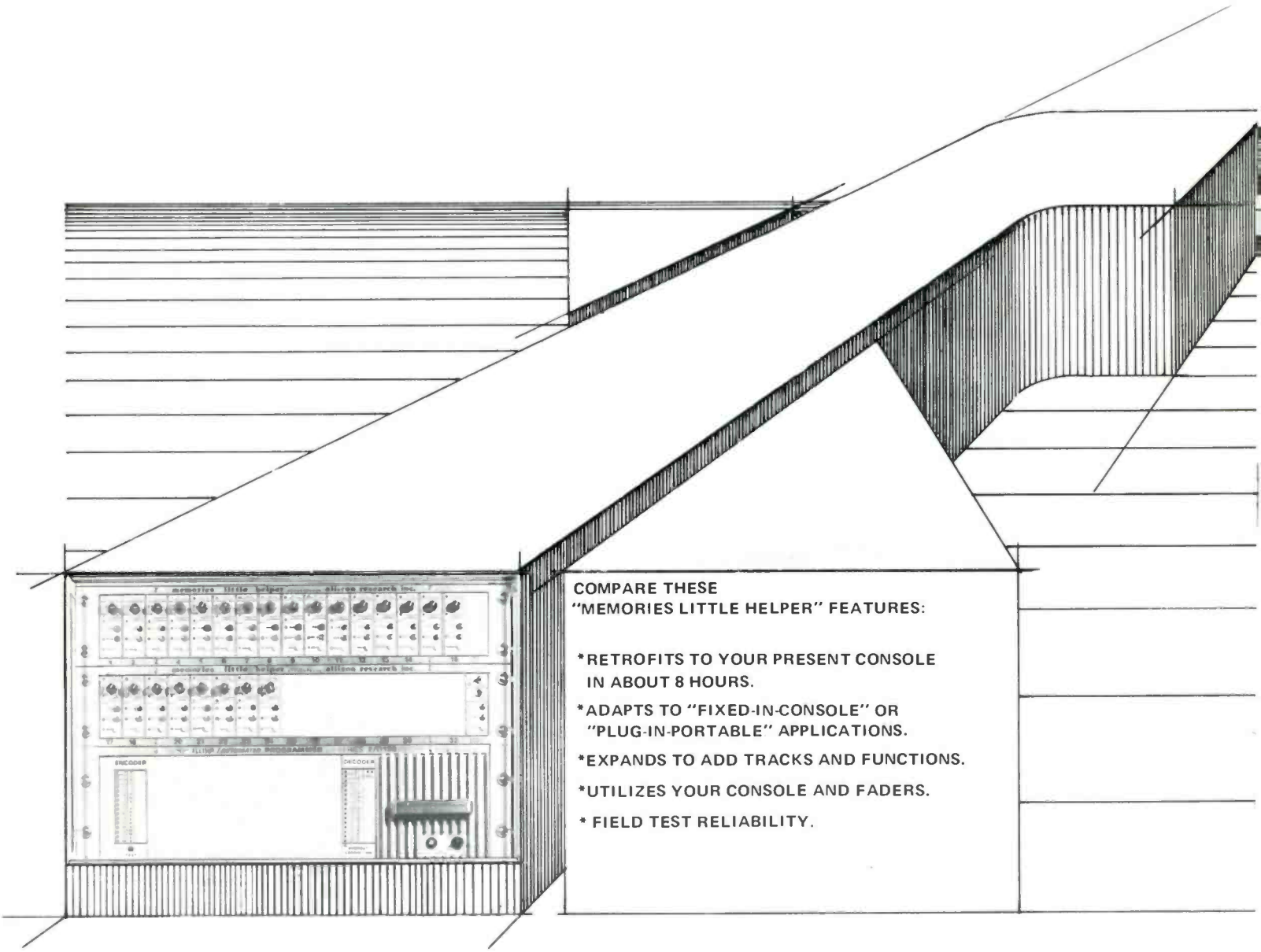

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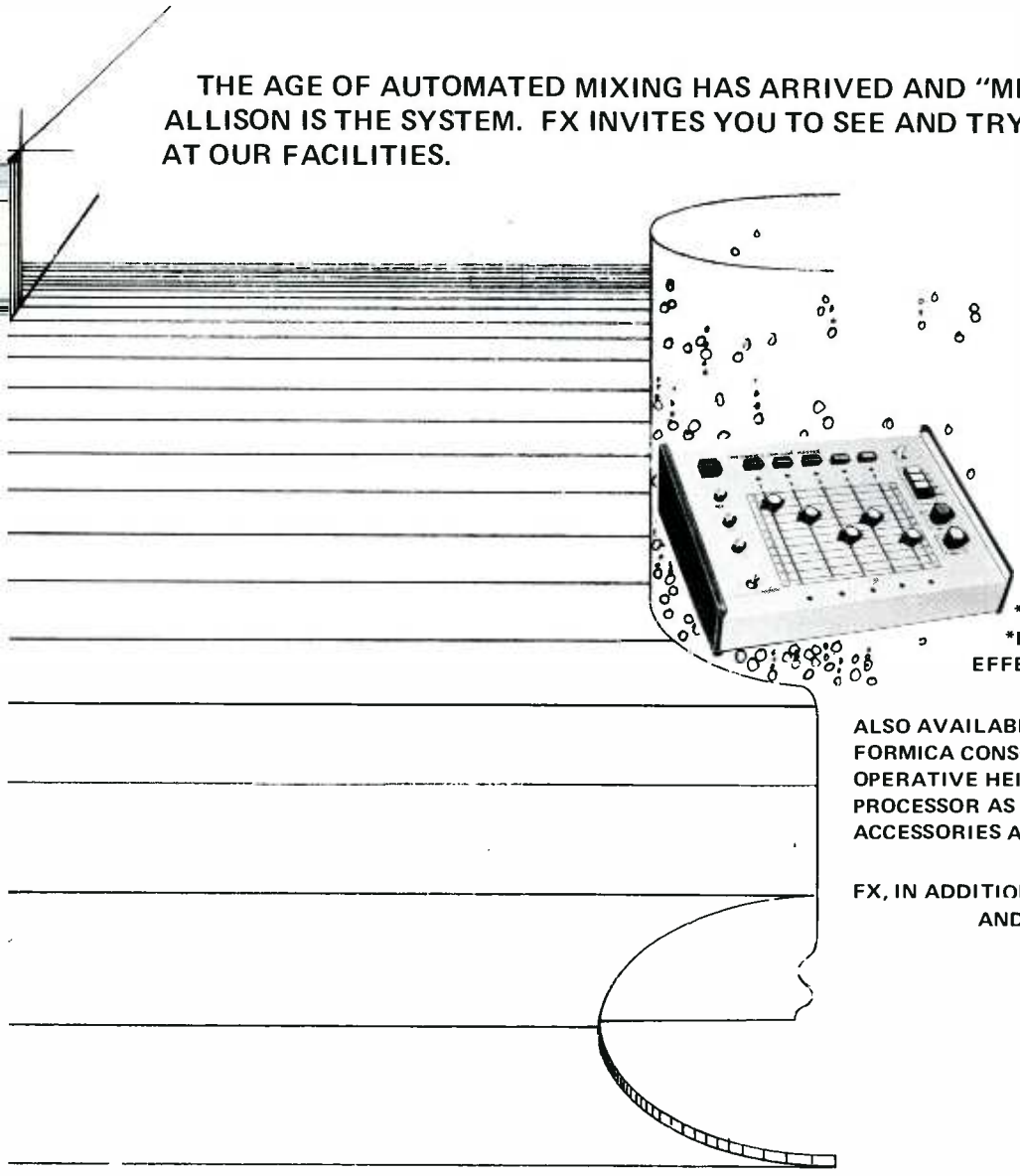

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SESSION A
ADVANCES IN SOUND RECORDING/ REPRODUCING TECHNOLOGY

TUESDAY, MAY 13, 9:30 A.M.
Chairman: Carl S. Nelson
Magnasync/Moviola Corporation,
North Hollywood, California

A–1 EVOLUTION OF THE NEW MARK III CUTTING SYSTEM FOR CD–4
A–2 DIGITAL BIAS/ERASE YIELDS LOW NOISE AND INSERT MAGNETIC RECORDING
A–3 OVERCOMING RECORD WARP, LOW-FREQUENCY TURNTABLE RUMBLE AND ACOUSTIC FEED-BACK IN PHONOGRAPHS
A–4 DEVELOPMENT OF AN IMPROVED MODULATOR FOR DISCRETE QUADRAPHONIC DISCS
A–5 A LOW-FREQUENCY SPEAKER-AMPLIFIER SYSTEM
A–6 PEM 468: A NEW MASTERING TAPE WITH HIGH-OUTPUT, LOW-NOISE AND LOW-PRINT CHARACTERISTICS
A–7 SLEW RATE LIMITING RECOVERY CHARACTERISTICS OF LOW COST IC OPAMPS

SESSION B
SIGNAL PROCESSING

TUESDAY, MAY 13, 2:00 P.M.
Chairman: David E. Blackmer
dbx, Inc., Waltham, Massachusetts

B–1 A WIDE DYNAMIC RANGE PROGRAM EQUALIZER
B–2 AN AUTOCORRELATOR NOISE REDUCTION SYSTEM
B–3 A NEW APPROACH TO TWO QUADRANT MULTIPLIER DESIGN
B–4 UNITY SUM ELECTRONIC CROSSOVERS
B–5 AUDIO FREQUENCY APPLICATION OF INTEGRATED CIRCUIT ANALOG DELAY LINES
B–6 AN AUTOMATIC STEREO SEPARATION CONTROL

SESSION C
AUDIO IN BROADCASTING (AM/FM/TV)

TUESDAY, MAY 13, 7:00 P.M.
Chairman: Donald McCroskey
American Broadcasting Company
Hollywood, California

C–1 A HARD LOOK AT FOUR–CHANNEL SOUND
C–2 NOISE REDUCTION ENCODING WITH FM PRE-EMPHASIS REDUCTION – A PRELIMINARY REPORT ON AUDIO PROCESSING AND AVERAGE MODULATION CHANGES
C–3 A VERSATILE, HIGH-SPEED "OFF-LINE" SWEETENING SYSTEM FOR TELEVISION AUDIO POST PRODUCTION
C–4 AN AUDIO CONTROL FACILITY FOR A TV "ON AIR" CONTROL ROOM
C–5 BROADCAST CARTRIDGES – PAST, PRESENT AND FUTURE
C–6 MATHEMATICAL THEORIES OF THE QS SYSTEM AND THEIR APPLICATION TO THE LATEST QS ENCODING METHOD

SESSION D
SOUND REINFORCEMENT

WEDNESDAY, MAY 14, 9:00 A.M.
Chairman: Edward S. Jones,
Brigham Young University
Provo, Utah

D–1 A NOVEL STADIUM SOUND SYSTEM
D–2 EXPERIMENTS IN THE ENHANCE-MENT OF THE ARTIST'S ABILITY TO CONTROL HIS INTERFACE WITH THE ACOUSTIC ENVIRONMENT IN LARGE HALLS
D–3 PRACTICAL CONSIDERATIONS OF TOURING ROCK PA SYSTEMS
D–4 THE "IDEAL" CENTRAL CLUSTER DESIGN
D–5 SOUND REINFORCEMENT AND EQUIPMENT INTERFACE FOR THE ACADEMY AWARDS
D–6 MODEL STUDIES OF "PRACTICAL" MULTI-TRANSDUCER LOW-FREQUENCY LOUDSPEAKER ARRAYS
D–7 DIRECTIONAL CHARACTERISTICS OF PHASED AUDIO REPRODUCERS

SESSION E
ELECTRONIC MUSIC

WEDNESDAY, MAY 14, 2:00 P.M.
Chairman: Robert Moog
Moog Music, Inc.
Williamsville, New York

E–1 A SYSTEMS DESIGN APPROACH FOR MORE EFFECTIVE UTILIZATION OF VOLTAGE CONTROL IN MUSIC SYNTHESIZERS
E–2 RECORDING SYNTHESIZED INSTRUMENTAL MUSIC
E–3 DYNAMIC SPECTRUM CHANGES OF ORCHESTRAL INSTRUMENTS
E–4 EXPERIMENTAL FOURIER SERIES UNIVERSAL TONE GENERATOR
E–5 LIVE ELECTRONIC MUSIC IN LARGE AUDITORIUMS
E–6 A POLYPHONIC KEYBOARD FOR A VOLTAGE-CONTROLLED MUSIC SYNTHESIZER

SESSION F
SPECIAL APPLICATIONS IN AUDIO

WEDNESDAY, MAY 14, 7:00 P.M.
Chairman: William L. Cara
La Jolla, California

F–1 AUTOMATIC MICROPHONE MIXING
F–2 THE DEVELOPMENT OF DIAMOND CUTTING STYLI
F–3 WHAT'S SO SACRED ABOUT EXPONENTIAL HORNS?
F–4 AN AUTOBIOGRAPHY OF A MONITOR SPEAKER AND OF ITS BIG BROTHER
F–5 STANDARDS AND DESIGN PARAMETERS FOR ARCHITECTURAL ACOUSTIC ANALYZERS
F–6 NEW LOW-COST REAL-TIME AUDIO ANALYZER
F–7 THE V-FET: A NEW GENERATION OF AUDIO AMPLIFIERS

SESSION G
SOUND, HEARING AND THE ENVIRONMENT

THURSDAY, MAY 15, 9:00 A.M.
Chairman: Robert Gales,
Naval Undersea Center
San Diego, California

G–1 PATHOPHYSIOLOGICAL EFFECTS OF NOISE

SESSION G
continued

G–2 COMMUNITY NOISE RATINGS
G–3 PSYCHOACOUSTIC PROBLEMS IN NOISE CONTROL
G–4 AIRCRAFT NOISE REDUCTION
G–5 HIGHWAY NOISE-EVALUATION AND CONTROL
G–6 SAN DIEGO NOISE ABATEMENT AND CONTROL PROGRAM

SESSION H
PHYSICS OF SOUND AND MUSIC AND HUMAN PERCEPTION

THURSDAY, MAY 15, 2:00 P.M.

Chairman: **Stanford Fidell**
Bolt, Beranek and Newman, Inc.
Canoga Park, California

H–1 GEOMETRY OF SOUND PERCEPTION
H–2 ON THE MEANINGFULNESS OF NOISE MEASUREMENTS IN AUDIO SYSTEMS
H–3 STATISTICAL ROOM ACOUSTICS
H–4 A STUDY OF TIME DOMAIN SPEECH COMPRESSION BY MEANS OF A NEW ANALOG SPEECH PROCESSOR
H–5 BRAIN SIGNS OF AUDITORY INFORMATION PROCESSING IN MAN
H–6 RESONANCES IN WIND INSTRUMENTS AND THEIR MUSICAL SIGNIFICANCE
H–7 THE PERCEPTION OF SPEECH

SESSION J
ARCHITECTURAL ACOUSTICS AND ROOM DESIGN

FRIDAY, MAY 16, 9:00 A.M.

Chairman: **Richard Boner**
Boner Associates
Austin, Texas

J–1 REVERBERATION TIMES OF CHURCHES: A SURVEY AND EVALUATION
J–2 MORE ACCURATE CALCULATION OF THE ROOM CONSTANT
J–3 THE EFFECT OF ARENA ACOUSTICAL DESIGN ON TEMPORARY AND PERMANENT SOUND REINFORCING SYSTEMS
J–4 A COMPARISON OF THE ACOUSTIC QUALITY OF A SOUND RECORDING WITH THAT OF THE ORIGINAL: THE EXPERIENCE IN HESSISCHER RUNDFUNK STUDIO 1
J–5 REVERBERANT FIELD ENERGIZER AND ELECTRONIC FORESTAGE CANOPY SYSTEM
J–6 A SYSTEMS APPROACH TO THE LOUDSPEAKER IN A MONITORING ENVIRONMENT

SESSION K
AUDIO MEASUREMENTS AND STANDARDS

FRIDAY, MAY 16, 2:00 P.M.

Chairman: **Juergen Wahl**
United Recording Electronics Industries, No. Hollywood, CA

K–1 STRUCTURES AND SPHERES OF INTEREST OF THE STANDARDS ORGANIZATION
K–2 MEANINGFUL STANDARDS DO SOMETHING FOR YOU
K–3 INTERPRETING FIELD MEASUREMENTS OF DIRECTIVITY FACTOR AND THEIR RELATION TO THE PROPOSED STANDARD METHOD OF MEASURING THE DIRECTIVITY FACTOR OF LOUDSPEAKERS USED IN COMMERCIAL SOUND WORK
K–4 SOME NEW AUDIO MEASUREMENTS
K–5 NOISE MEASUREMENTS IN AUDIO
K–6 EVALUATING OPEN PLAN ACOUSTICS
K–7 GAS LASER APPLICATIONS AND ACOUSTIC MODELING

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* **AUDIOLOGY TEST:** As a project of the Los Angeles Section, there will be an audiometer test given to Convention-goers in a portable soundproof room. Copies of the resulting charts will be reviewed by staff members of the Department of Head and Neck Surgery at the UCLA Center for the Health Sciences. Dr. Donald D. Dirks and Sam Gilman of UCLA are guiding the Los Angeles Section in this undertaking. Those who wish to receive results of their tests will be advised through a number identification system.

EXHIBITS: Indications are that we should surpass last spring's number of participating exhibitors. All the regulars — practically without exception — and many new ones are taking space. Exhibit hours: Tuesday and Wednesday, May 13 and 14—1:00 P.M. to 9:00 P.M.; Thursday and Friday, May 15 and 16 — 11:00 A.M. to 5:00 P.M.

CULTURAL & SOCIAL ACTIVITIES

Once again, the Cultural and Social Committee has arranged a program of off-hour activities for Convention visitors. Some of the highlights include:

- **VISITORS' HOSPITALITY,** adjacent to Registration
  Hosted by Doreen and Bob Rypinski, visitors can enjoy early-morning coffee with a run-down of the day's events. Maps and details on the local area in different languages, recording studio locations, tickets to television broadcasts, and other helpful information available in a relaxed and informal atmosphere.

- **TRIPS AND TOURS**
  For those not directly involved in Convention activities, Pamela Strickland has plans for four enjoyable days in the Los Angeles area, including a visit to the village of Solvang, an authentic Danish community located north of Los Angeles, and a trip to Huntington Gardens and Library in Pasadena, home of many original works of art and manuscripts. Final plans are still in the making; details will be available from the Audio Engineering Society Offices, and will be posted daily near the Registration Desk during Convention.

- **THURSDAY, MAY 15**
  7:00 P.M. **SOCIAL HOUR** • Los Angeles Room
  8:00 P.M. **AWARDS BANQUET** • Golden State Room
  The traditional Social Hour is the only opportunity for many Convention visitors to relax and catch up. This will be followed by a fine banquet celebrating the presentation of awards to distinguished AES members for outstanding achievements. The evening will be topped by a Concert of Electronic Music presented by renowned artists. Performances will include Mike Brigida and Tom Piggott on ARP synthesizers and Roger Powell on the Moog synthesizer.

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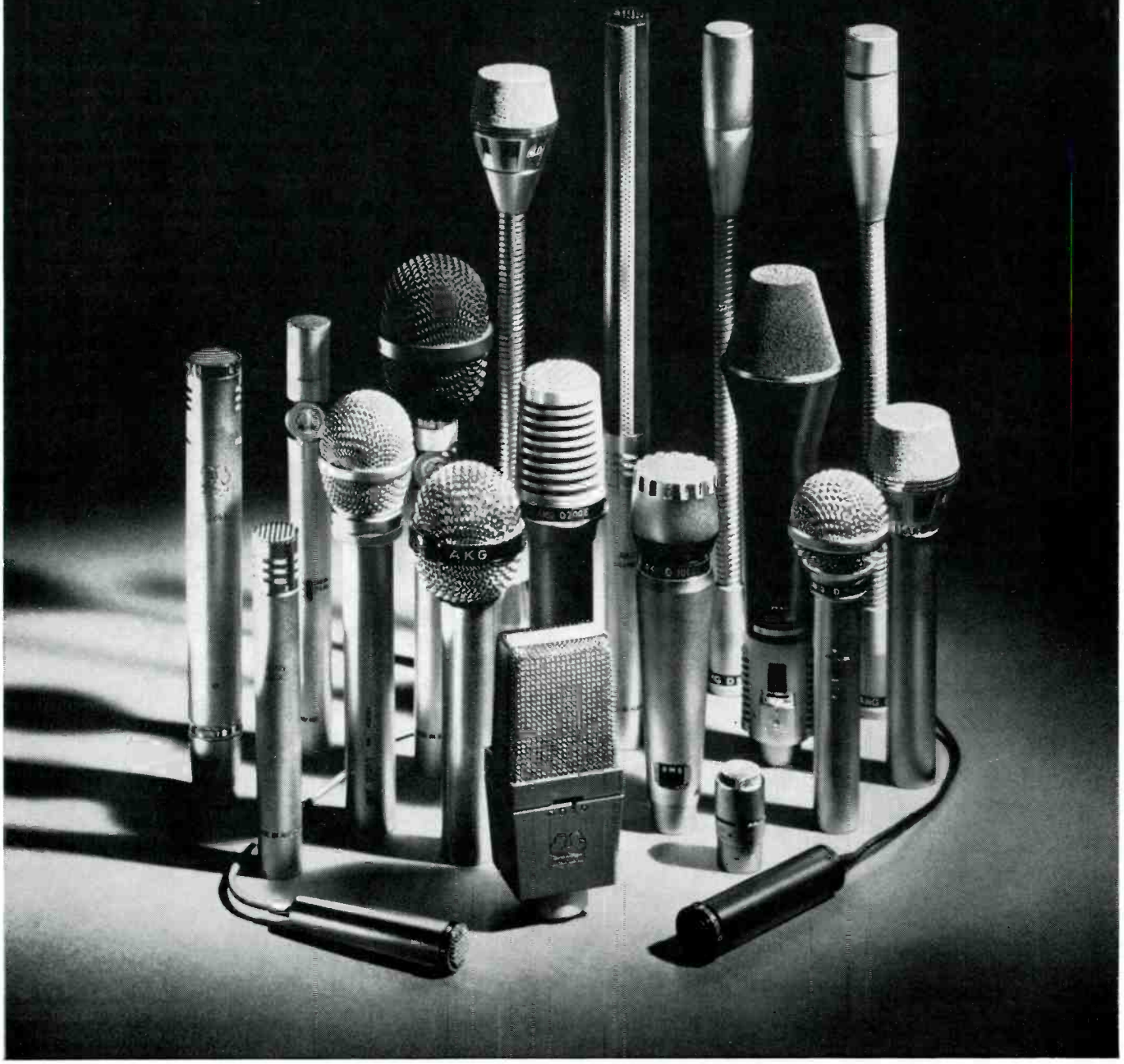

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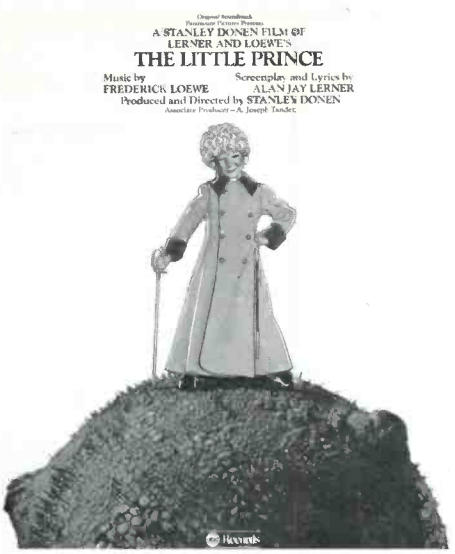
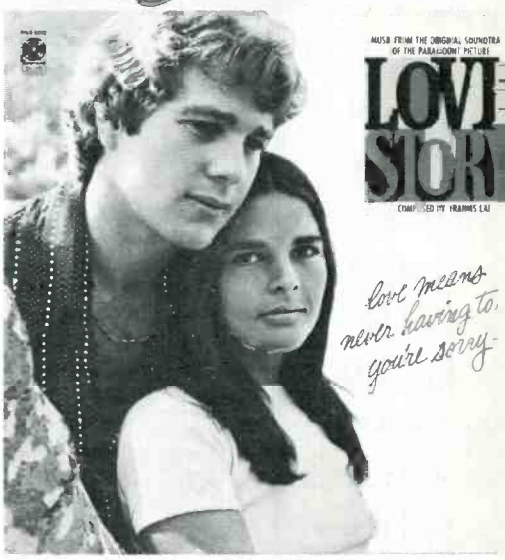
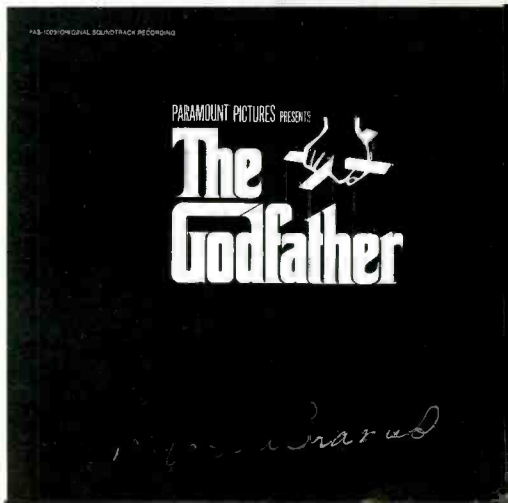






PRODUCING ORIGINAL SOUND TRACK ALBUMS

AN INTERVIEW WITH PRODUCER TOM MACK
BY PAUL LAURENCE

First off, what is the purpose of a soundtrack album? What should it be or contain?

TOM MACK: Well being a record man, I like a soundtrack album to be a record that people will buy for its own sake, not just because they liked the movie. Like any other record, it should be well-programmed, well-paced, and interesting. If you're lucky enough to have a strong main theme, that can help immensely. Often there's a tendency, especially from the motion picture production company's standpoint, to consider the record album purely as a piece of advertising for the film. If I'm doing my job properly, it will serve this purpose as well, but my priorities, understandably, are with the record itself.

Won't a hit picture almost guarantee a big soundtrack album?

TOM MACK: At one time this was true – it was almost axiomatic that if you had a smash hit film, the record would sell and it didn't much matter what it had on it. Conversely, if you had the most marvelous music in the world in a bomb film, it wouldn't sell. Today, the latter is more likely to be true than the former – great music from an unsuccessful film probably won't sell, unless some sort of cult springs up around it or there's a lot of word-of-mouth advertising. But the other is certainly not true anymore – if you have a great film, you'd better have a good record too.

PAUL LAURENCE: What are the responsibilities of a soundtrack album producer?
TOM MACK: My job takes up where the screen composer's leaves off. The composer's one and only obligation is to the film itself, his job being to create music which will enhance or characterize what's going on in the film. If he's doing his job properly, he's not giving a thought to any subsequent records. My obligation is totally the opposite – to sound, to the ear, and to the record. I've got to take the composer's creation and make it into a coherent, well-paced album having fairly normal-lengthed cuts that can be played over the air.

PL: How do your responsibilities differ from those of a traditional producer?
TM: Well, as you know, I'm a traditional producer too. When I work live in the studio, I have an opportunity to work with the artist, the arranger, the material, etc., to build a record from scratch. You're in charge, and you can do pretty much what you like. When you're producing a soundtrack album, most of the time you're dealing with material that's already been recorded. As a result, you're limited in that you can't go back into the studio and do it over again, unless the company is willing to forego the FTC-permitted designation of *original soundtrack album.*

PL: Is that the ruling – that you can't recut any of it?
TM: Pretty much. One thing you can do, though, is *sweeten* existing material. For example, if a piece of music you want to use should be enhanced with voices, that's okay. Or,if you want to put rhythm on a piece of music that was recorded strictly ad-lib – out-of-tempo – that's okay too. This is fairly difficult to do – we call it adding a *rubber band* rhythm section.

PL: Can you elaborate on that?
TM: Yes. Take for example, the main theme from *Love Story.* For the film, it was recorded in symphonic fashion, and had no real beat. We wanted to use it as a single, but couldn't really compete with all the cover versions that were coming out unless we made it more rhythmic. So what we did was bring a rhythm section into the studio, and they overdubbed their parts to the existing music. By *rubber band,* I mean that the beat might have to be *stretched* to coincide with something that's already on the track, but still give you a feeling of a beat. If you wanted to put a metronome to the finished product, though, you'd clearly hear the disparities. Even so, it's amazing how much you can get away with, and how good it can actually sound.

The *Love Story* single, in addition to having that kind of rhythm section, was a composite of three separate segments – three short statements of the theme. As I recall, one cut in particular had to be made because the song was going to be way too long. I've never been happy about that cut – to me it's always sounded awkward. An amusing thing about it, though, was that another record company came along and covered it, cut and all. Evidently, the arranger was told to stick pretty close to the original, and he actually imitated my bad cut!

PL: What is the procedure you go through in making a soundtrack album, and how does it differ from the way you did it say 10 years ago?
TM: Well, I first started doing this around 1950, so let's contrast now to then.

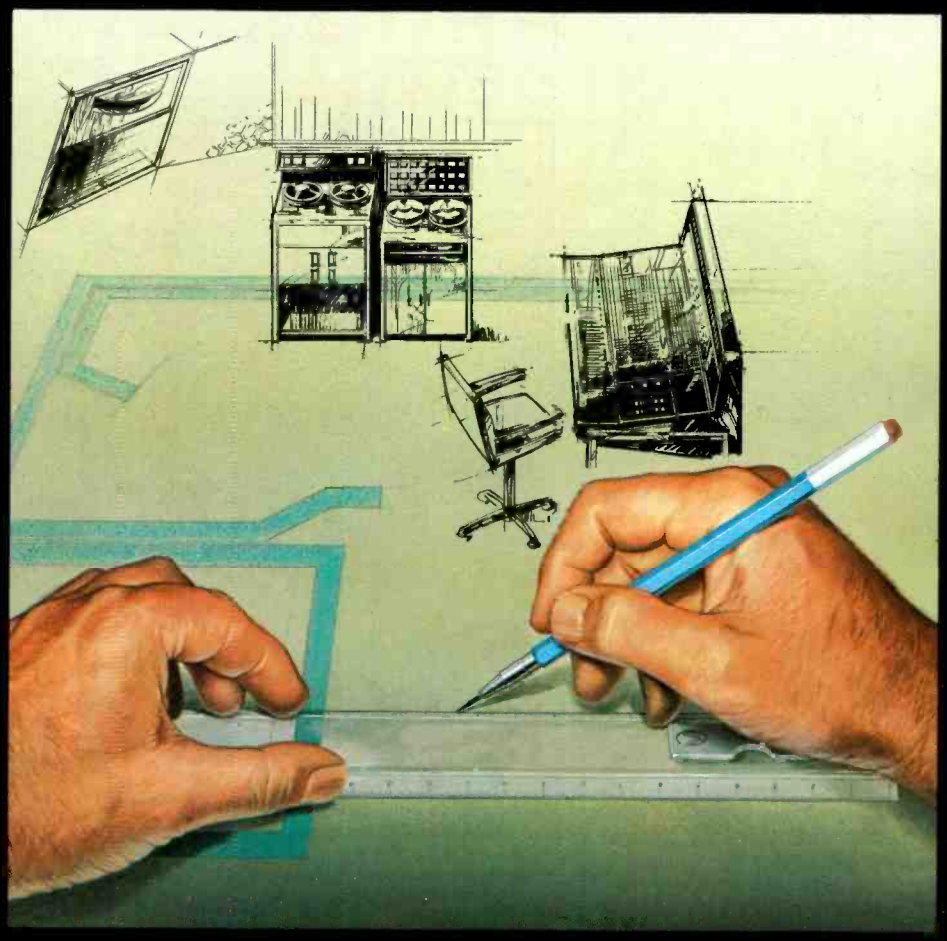
First of all, there are two separate unions involved. One controls sound in motion picture studios (LATSE), and the other sound in the recording studios (IBEW). The IBEW people were not allowed to use sprocket-driven equipment, meaning optical film, of course. In those days, it was therefore necessary for me to go physically to the motion picture sound stage with portable sound equipment, technicians, and a sound-recording engineer, who'd function pretty much as a consultant. After the equipment was set up, we would make an electronic transfer of the sound from the optical film, taking it through the sound stage console and coming out on tape. In so doing, we had to pretty radically alter equalization, for the following reason: back then, when sound was recorded for film, it was designed to be heard over those huge theater speakers. Now a record is designed to be played with home equipment over much smaller speakers. Because of this, we had to increase the signal on the high end, boost the midrange, and attenuate much of the bass. Home speakers just could not handle that much bass – you'd have to take your level down much too far to keep your needle from jumping out of the groove.

This was pretty much how soundtrack albums were produced until the middle and late '60's, when the studios were being renovated and got full multi-channel tape capabilities. It's no longer a makeshift, patched-in situation, but inherent in the system. Nowadays, as far as the technical aspects go, making a soundtrack album is just like making any other kind of album – the record comes from your multitrack master tape, as does the actual film soundtrack.

PL: What are the main problems you're likely to come up against in the course of making a soundtrack album?
TM: Well first of all there's a very basic logistical problem. I can't help but think of the time that someone asked Barney Kessel what the most important thing for a studio musician to know was, and he answered "Never park your car in the alley beside Western Recorders." The main problem revolves around the fact that the music is generally the last element of film production to be completed. Until the music is done and cut to the film, you have no way of knowing the contexts and relative importance of one theme or cue over another. Oftentimes, the final editing will not be completed until days before the release of the picture, which means I might be in the studio for four or five days straight with little or no sleep. The irony is that while the music is the last element of production, the album, for the benefit of all concerned, should precede the release of the film!

continued on page 47 . . .

Acoustical Guarantee



Performance Specifications

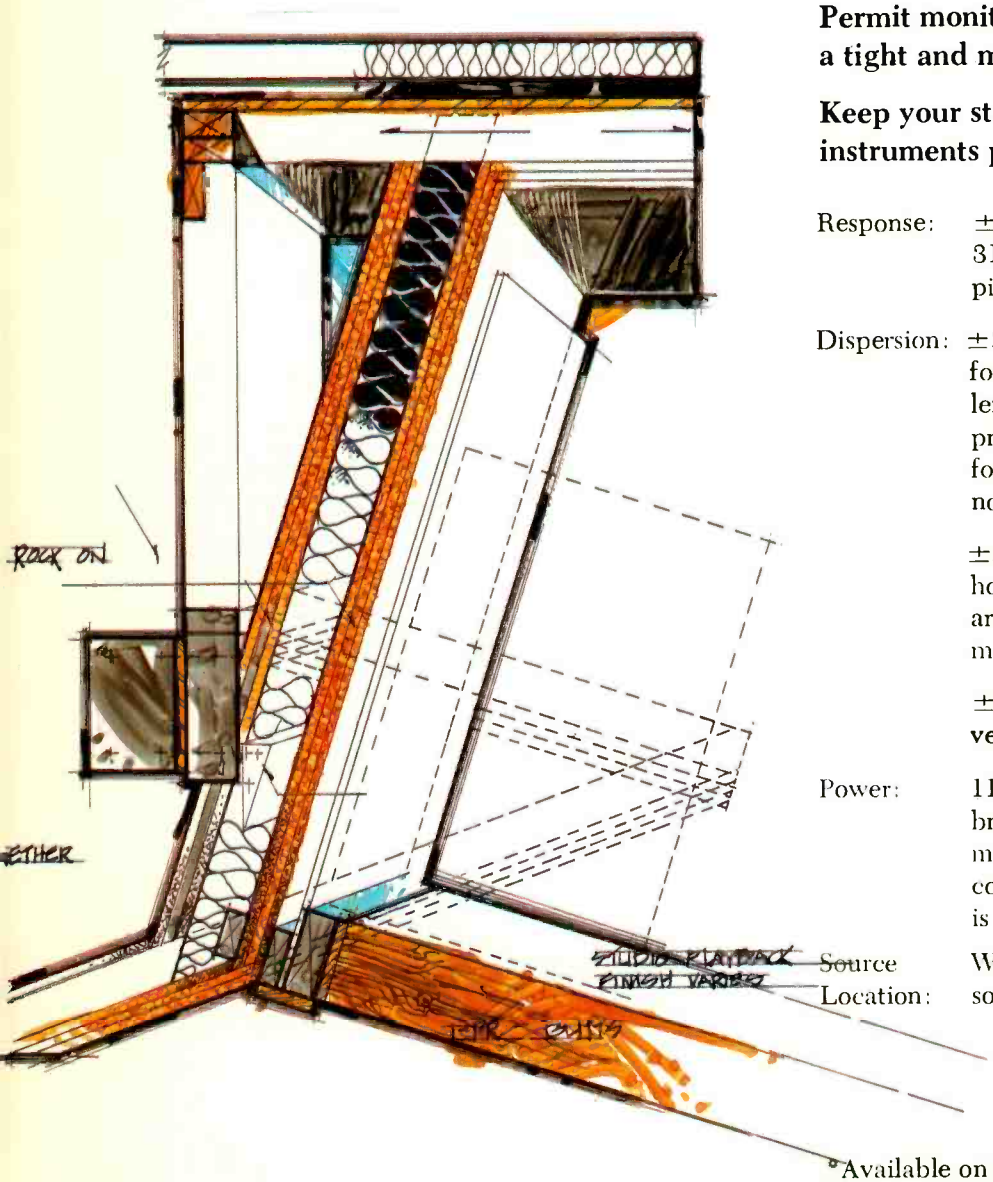
from acoustic design to down beat...

Westlake Audio

"The control room sounds good here but not over there. Stand up and you've lost your mix. Lean back in your chair and all the bass is gone. The monitor has to be loud to hear it. Turn your head and big changes occur. The stereo image moves."

"The drum leakage in this studio is terrible. The strings sound great but the bass is loose and muddy. This room is so dead the sound isn't happening and the musicians can't get into it."

These are subjective observations which producers and engineers have made and lived with for years in many studios. We at Westlake are prepared to talk to you about a guarantee *against* those things happening in your studio.



Guarantee of Acousti by Westlake Audio*

WE WILL GUARANTEE YOU A CONTROL ROOM WHICH WILL:

Allow you to stand . . . sit . . . lean forward or back . . . move left or right and subjectively not change your mix.

Let you accurately pinpoint any musical instrument within a 360° quad listening environment.

Permit monitoring loud or soft while retaining a tight and musical sound.

Keep your stereo "locked center" on all instruments panned to the middle.

Response: ±3 dB upon speaker installation, 31 Hz-16 KHz measured with B & K ⅓ octave pink noise source. *Between speakers, ±1 dB.*

Dispersion: ±2 dB @ 10 KHz across a minimum 10 foot horizontal plane at the console (from left of the engineer to the right of the producer or vice versa) from any one of the four monitors, measured with pink noise source.

±2 dB @ 10 KHz across a minimum 10 foot horizontal plane front to back in the mixing area from any one of the four monitors, measured with pink noise source.

±2 db @ 10 kHz from 6″ above console vertically to 6″ down from ceiling.

Power: 116 dB SPL minimum, linear scale, with broadband pink noise source from one monitor measured at the mixer's ear. The control room potential with four monitors is a minimum of 128 dB SPL.

Source Location: Within 2 dB of *total sum* from any two sources in the 360° quad circle environment.

*Available on all new projects from Jan. 1975 on.

cal Performance Specifications

WE WILL GUARANTEE YOU A STUDIO WHICH WILL:

Have a tight rhythm sound under all recording conditions yet allow the producer and engineer the option of changing the midrange character anywhere from "dead" to "very live" in less than sixty seconds. — *Any location in the room.* —

Provide drum cages which are live inside, something that the drummer can get into, allowing you to get a bright drum sound from an open drum cage.

Let you obtain a natural piano sound with excellent isolation from loud electronic instruments. — *With the piano in the room, lid open and not caged in.* —

Provide an echo chamber with low end "mud" removed by trapping in the chamber, resulting in a chamber that "sings."

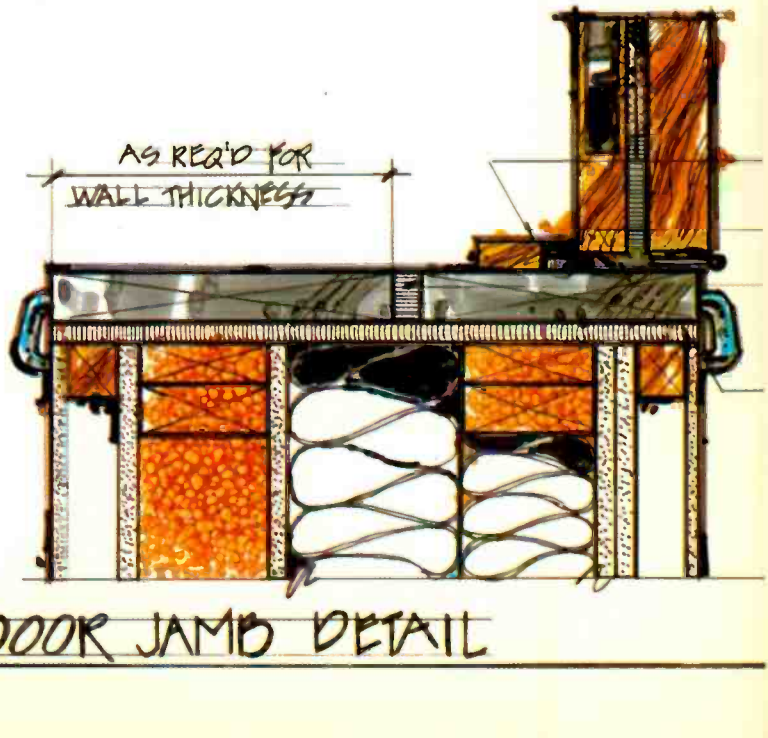
Room Character: The characteristic "room sound" which results from recording in a three dimensional area is eliminated by the utilization of an active ceiling. From 40 Hz up, this produces an infinite third dimension such as would be present in an amphitheater.

Separation: Active traps are built into the studio walls which allows "in-studio" vocals, eliminating the need for the usual vocal booth. 30 dB of isolation can be provided between the band and a vocalist only 10 feet away, resulting in 30 dB of isolation @ 40 Hz or tuned frequencies.

Traps: Drum cages, bass traps and broad band attenuators will provide in excess of 24 dB isolation @ 40 Hz. The piano can be recorded in the studio while still providing over 20 dB broadband rejection of unwanted sound to the piano mikes *with lid open!*

THE CONTROL ROOM AND THE STUDIO ARE YOUR TOOLS AND SHOULD WORK FOR YOU . . . NOT AGAINST YOU.

THAT'S WHAT AN ACOUSTICAL GUARANTEE IS ALL ABOUT!



Kent R. Duncan, President, Kendun Recorders, Burbank, California: *"The new room has been in operation for six months now and our success is as much a tribute to Westlake Audio and Tom Hidley as it is to our long hours and attention to detail (and possibly some good engineering). Our Westlake room made us a 2 studio operation but instead of just doubling our gross, we went from $12,000 a month to $60,000 a month. The incredibly accurate planning of our Westlake turnkey installation resulted in completion exactly on time, response precisely as promised, all equipment functioning within one day of installation, and all within budget! In the past six months we have mastered such acts as Stevie Wonder, Bob Dylan, America, Buddy Miles, Fleetwood Mac, Rick Nelson, Tower of Power, Livingston Taylor, Isley Bros., Rod McKuen, Nitty Gritty Dirt Band, Emitt Rhodes, Richard Greene, El Chicano, Nana Mouskouri, Cleo Laine, Bola Sete, San Sebastian Strings, Jo Stafford, Maxayn, Pharoah Sanders, Archie Shepp, Ballin' Jack, Vickie Lawrence, Maureen McCormick & Chris Knight, Don McLean, Vikki Carr, Bill Medley and even Rodney Allen Rippy. Over half these acts were recorded on Westlake monitors in various studios around the country, attesting to the fact that truly, you are the professional."*

Christopher Stone, President, Record Plant Recording Studios, Los Angeles: *"As you know, we have used Westlake Audio and yourself since the inception of the company for all of our studio design, construction, electrical interface and implementation. During the past four years you have designed and implemented eight studios for us in New York City, Los Angeles and Sausalito. Obviously we are known as a Westlake-designed operation. We have built our total reputation around your studio design and have always been happy with our decision to utilize you on an exclusive basis for all our acoustical requirements and equipment consultation. The success of your design speaks for itself in the form of our success as an independent studio operation."*

John Sandlin, Vice President A & R, Capricorn Records, Macon, Georgia: *"Words alone cannot express my appreciation for the friendly and courteous atmosphere I enjoyed while at Westlake mixing Bonnie's (Bonnie Bramlett) album.*

It was really a pleasure to work with such extremely competent and dedicated people. Thank you for giving me an opportunity to experience the automated mixing facilities and to work around the type of people I love and can relate to.

Take care of Baker, he's incredible."

John Boylan, John Boylan, Inc., Hollywood, California: *"First of all, this is my third project in a row to be mixed on your monitors and once again it looks like we have a winner – a record that sounds as good at home as it did in the control room. From a producer's nontechnical viewpoint, this ability to trust a studio monitor and come out with even results is extremely satisfying. Secondly, the Westlake Monitor never seems to vary in any substantial way from studio to studio, in the control rooms that you've designed. So I have no worries about consistency in today's widely dispersed recording scene."*

Complete, unedited photocopies of these and many other testimonial letters are available on request from Westlake Audio.
Phone or write direct to Tom Hidley, President.

WE PUT OUR MONEY WHERE OUR MOUTH IS!

Below are excerpts from a typical acoustical system acceptance from a client authorizing the release of the final portion of the construction monies from a trust account.

SYSTEM PERFORMANCE ACCEPTANCE

In accordance with the terms set forth in that certain agreement contained within Westlake Audio's invoice number 3930 dated March 1, 1974 mutually accepted by Westlake Audio, Inc. and Sounds Interchange, the undersigned hereby:

1. Acknowledges receipt of and accepts a final sound measurement report from Westlake Audio, Inc.
2. Agrees that Westlake Audio has, as relates to the design and construction of the Sounds Interchange studio facility, Toronto, Canada,it met or exceeded all performance specifications as set forth in the Westlake Audio brochure entitled Acoustical Design The Key To The Success Of Your Studio as amended and signed by T. L. Hidley on February 8, 1974.
3. Acknowledges that all work has been completed in a satisfactory manner and that all materials have been delivered.
4. Acknowledges the fact that Westlake Audio, Inc. has complied with and fulfilled all the terms set forth in a certain Letter of Credit drawn in favor of Westlake Audio, Inc. and hereby instructs the advising bank — Bank of America, Westlake Boulevard, Westlake Village, California, U.S.A. to honor and pay at sight said Letter of Credit on or after December 6, 1974.

SOUNDS INTERCHANGE LTD.

By _____

Dated DEC 6/74

THAT'S WHAT AN ACCOUSTICAL GUARANTEE IS ALL ABOUT!

from acoustic design to down beat...

Westlake Audio

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Los Angeles, California 90048
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Nashville, Tenn. 37203
(615) 254-9998

TOM MACK

continued from page 42 . . .

Another problem is that you may not have enough music for a whole album, in which case you really have to eke one out. This could be because the composer didn't write enough, or because the production people wanted to use the music sparingly. A good example would be *The Godfather.* I would imagine that Nino Rota wrote a pretty complete score, but the film people decided that they wanted a *Dragnet/Police Story*-like feel – which turned out to be very effective – with the main theme in appropriate places and some additional stings, buttons, and cues. As a result of this sparseness, I had to fill out the album using source music, which is material other than that written specifically for that film. You know, music that in the film is played over the radio or performed in a nightclub, etc. I feel, and I'm sure that most composers would agree, that source music is important, both practically, in that it can help you flesh out an album from an otherwise skimpy score, and artistically, because it's good for programming. I like variety on records, and source music can provide that change of mood for the listener. It also functions to place the movie historically and geographically.

PL: What about all those short musical bits you have to deal with?

TM: Ah, that's the other major problem. As I said earlier, the composer's obligation is to the score. This means that he'll sometimes have to come up with very short segments of music. Now when I want to use one of those little bits, I generally have to find a way to link it to something else because it doesn't make sense to have a 30-second album cut, except under very special circumstances.

PL: How do you decide which bits should be used?

TM: Well, the first thing I do, before anything else, is get a 7½ ips rough mono mix of the entire score. I then listen to it, time each selection, and take notes. I try to note the character, quality, and usability of each piece.

PL: What kind of things would you write down?

TM: I'll put down whether the piece is major or minor, maybe the dominant instrument, and what *type* of dramatic cue it is. Sometimes the sequence numbers will tell you which is your opening theme or your ending theme. If a piece of music is labelled "1M1," you can be pretty sure that it's the main title if there is one. The first "1" indicates the number of the reel that it occurs on, "M" is for music, and the last "1" means that it's the first musical cue on that reel.

For my own notes, I usually assign letters – A, B, C, D, etc. – to the major themes. I might use a star if the cut is long enough and usable as is. If a segment is good, but incomplete for one reason or another, I might put "could be used if more available." Perhaps 16 cuts later, I'll hear the fragment that belongs there.

After a few days of this, I'll see the film to see *how* each piece of music is used, to try to get an idea of each piece's relative importance in the movie. As I sometimes make up the titles for the album cuts, it's important to know the contexts in which the musical pieces occur. For example, the composer might have put something like "Bugsy Gets Killed" – not a very good album cut title – which I would then change. Hopefully, it would be to something that still evokes the scene in the movie that it came from.

I also study the scores, because that tells me the key relationships. I can make *paper edits* – see with my eye where two bits could be spliced together at say bar 13 because the chords will resolve there, or maybe there's a pause, etc. I'm looking for the optimum points of joining.

PL: When you've decided which bits you're going to use, how do you join them? Do you crossfade them, splice them, put an explosion between them, or what?

TM: Actually, I may do any or all of those. I often do crossfading, especially when two pieces, because of disparate tempos or keys, won't really segue together. Sometimes I do this deliberately – crossfade two very different kinds of pieces together. This technique can be very effective in a filmic, almost visual way, in that it evokes the way pictures are cut. A lap dissolve, for example, is the cinematic equivalent of a crossfade.

Occasionally, there'll be a situation that calls for a multiple crossfade. One that I very much liked was the end title I *created* for "The Great Gatsby." At the end of the film was a reprise of Daisy's theme, which was Irving Berlin's "What'll I Do?" – a very lush, pretty arrangement. That was crossfaded into a very weird statement of that same theme, but in a totally unrelated key. The weird version was used to underscore the scene where Gatsby is killed in the swimming pool. What was interesting was that the first notes of the weird theme *answered* the ending notes of the first theme. I let that run for 11 or 12 seconds, before crossfading *it* into a bunch of flappers singing "Ain't We Got Fun?" which came at the end of the film and signified the jazz age rolling on without Gatsby.

PL: Will you do one of these multiple crossfades in one pass of your "receiving" tape?

TM: Ah, I'm glad you touched on this. The way I do it requires a 16-track machine and three 2-track machines. First you dub the major sections down to stereo from our 16-track masters, making sure that the musical elements common to each section appear in the same positions each time – left, right, phantom

center, etc. Then, using a white pencil mark for reference, you roll back from the end of our 2-track Section 1 a little to before the point where you'll want the crossfade to begin. On a second 2-track machine, you have Section II set up and ready to roll, with a cue mark just ahead of where you want the signal to come in. On the third 2-track, you create the overlap by recording the tail end of Section I and then the beginning of Section II, using the pots on the console to wipe one out and bring the other in. You stop just a few seconds after the overlap, because that segment is a generation older than the original Sections, and you'd like to have as short a segment of second generation tape as you can arrange. You then snip out this little piece and splice it between Sections I and II. If you were doing a three-part montage, with crossfades at both overlap points, you'd simply repeat the process between Sections II and III.

There may be a better way to do this, but I like this method because it gives you the most freedom for trial-and-error placement of the fade-in and fade-out.

PL: Can you hear any sort of presence or fidelity change at the crossfade itself, where the tape is a generation older?
TM: Well, there are people who claim to be able to tell the difference. Still I can't help but feel that, if it's done correctly, with matching levels, clean tape, clean pots, etc., you really can't. I've never known anyone who could really spot this in a blindfold test situation. Back in the days before multi-channel recording, we used to do transfers that got us as high as fifth or sixth generation. You could tell then, but I think that in this case, the degeneration is pretty negligible.

PL: To digress a bit, what are "stings" and "buttons"?
TM: A sting is a little *tah-dah.* A button is generally some sort of ending. Many pieces of dramatic cueing, and even some themes, just trail off because that's all the picture needed. On a record, though, you'd like it to have a more definite ending. Sometimes, you can find a button somewhere in the soundtrack that will work as an ending for a piece even though it wasn't written for it.

PL: What is your relationship to the engineer? Do you try to work with the one who recorded the music originally, or is there one in particular you like to work with?
TM: There are times when it's desirable to work with the same engineer who recorded the music, because he's already familiar with the material and knows where all the tracks are, but it's far from an insurmountable problem one way or another. I've found that it's very often impossible to work with the original engineer – he may be in Rome, and even if he's here in L.A., he's more than likely working on another project. As a result of this unpredictability, I've developed the habit of working almost exclusively with Thorne Nogar over at the Annex. Thorny's studio is located only a few blocks from Goldwyn, where I'd done a lot of work, which made our working together all the more convenient. Thorny's maintenance man developed a special set of patch cords so that we could take an 8-track machine to the Goldwyn dubbing stage and patch right into their equipment. Even after the sound stages graduated to multi-channel equipment, I continued to take the master tapes from wherever they were done and go to Thorny with them.

PL: What was the most difficult or time-consuming project you've ever worked on?
TM: I would have to say "The Little Prince." It's funny, because at the outset I thought the project was going to be a breeze. The songs had been recorded well in advance, but as the picture was undergoing radical editing changes, it was still undecided which numbers would be in and which ones would be out. Now if this were normal dramatic scoring, that is, dealing with just background music and themes, there would be no problem because you're allowed to use in the record of a film soundtrack anything that was recorded for the film, even if it's later taken out. But with musical numbers by Lerner and Loewe, anything not used in the movie would revert back to them, and as a result I couldn't use those pieces. Because of this, I ended up by mixing down every single song. Now this is where it gets difficult. For some reason, the 16-track tapes contained either vocals *or* background, so for the mix we had to find some way to combine them. We tried using a synch-pulse machine, but that didn't work and so we did it by guess and by golly. We'd line up our tapes and Thorny would push the start buttons on both machines together. He's really got a great pair of thumbs. Sometimes, we could only keep our machines together for a half-chorus. Then we'd have to back up, do it again, and splice. It was laborious to say the least, because we had to do the entire score that way. We could have gone to the film itself and come back from it, but didn't want to for reasons of quality. That project took me an entire month, working practically every day.

PL: Do you use a lot of studio effects, aside from level adjustment and equalization?
TM: I would have to say no there. We do, however, occasionally use sound effects to the same end, i.e., heightening the drama. In "The Dove," there's a scene where the boy is coming back into the harbor somewhere here in California, after having sailed around the world. At

this point in the movie, the theme is playing, and there are some general boat and sea noises. Now my tape had just the music, but I managed to hunt up the noises and dub them back in for the record. Funnily enough, the final boat whistle that crossfades into the downbeat of the theme was the same note as the theme's opening note. I felt that the boat noises were important, because they would evoke to the listeners that scene. Even for those who hadn't seen the picture, they would more precisely place that piece of music in space and time.

PL: What changes would you like to see come about to make your job easier?
TM: Obviously, I can't change the human condition – the fact that music comes last will always be the case. Still, it would be nice to mitigate this somewhat. Because music is not only the last element of production, but also the last element of the budget, the film producer might very well say "My God, we've run over here, don't run over on the music!" That means don't keep the musicians any longer than necessary, use only as much studio time as you absolutely need, etc. As a result, I'll occasionally run into an important piece of music that has an outstanding clam on it. Maybe it's not a terrible mistake, but when the music is exposed to the naked ear without any video, you're gonna hear it. Very often, the film producer will let a thing like that go and say "Don't worry, we're gonna have dialogue there" or "a cannon goes off there."

PL: Can you circumvent a situation like this?
TM: You do the best you can. Sometimes, you can go back to an earlier take and sneak in a bar or two. Often, though, you can't because the tempo has changed.

PL: I imagine that you're rarely lucky enough to have that instrument on a separate track.
TM: Sometimes you do, but then again it's not always a question of one instrument. There are all sorts of unwanted noises that end up on tape – chairs squeaking, mutes dropping, people bumping the music stands. These things can be especially evident in the quieter passages.

What I'm trying to get at is this: it would be marvelous if things could be worked out so that the record producer could be assigned earlier to the task, and could be in the studio while the recording is being done. He could then point out possible problems before it's too late.

To extend this line of thinking even further, it's great when the record company and the film company get together and work it out so that the soundtrack producer is the record producer as well. This is an experience I've been fortunate enough to have on a number of occasions, and it certainly makes my job easier.

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Dimensions: 19'' wide by 8¾'' deep by 1¾'' high.
Weight: 10 lbs.

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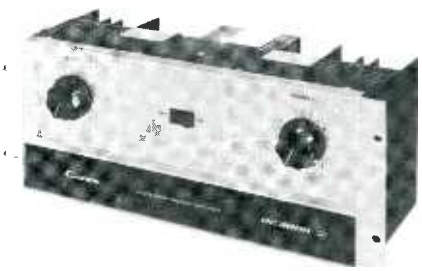
Output Power: 75 watts per channel minimum RMS (both channels operating) into an 8 ohm load over a bandwidth of 1 Hz—20 KHz at a rated RMS sum total harmonic distortion of 0.05%.
Dimensions: 17'' wide by 8¾'' deep by 5¼'' high.
Weight: 24 lbs.

The D-150 medium power amplifier, is by design an ideal audio amplifier with the kind of rugged reliability needed in portable sound systems. Especially where one to one amp/speaker ratios are used. Well known Crown protection circuits are an integral part of the D-150. Protection from mismatched or shorted loads is provided, and a series limiting resistor protects against excessive input signals. Controlled slewing-rate voltage amplifiers protect against RF burnouts. And a thermal switch cuts AC line power if overheating occurs from improper ventilation.

Output Power: 155 watts per channel minimum RMS (both channels operating) into an 8 ohm load over a bandwidth of 1 Hz—20 KHz at a rated RMS sum total harmonic distortion of 0.05%.
Dimensions: 19'' wide by 9¾'' deep by 7'' high.
Weight: 45 lbs.



The Crown DC-300A. High power is first thought of when referring to a "super" amplifier. However, this Crown amplifier is also super in its reliability; super in its capability to deliver sound without distortion, and super in its ability to power any type of load, from 2.5 to 16 ohms, resistive or reactive! And whether it powers a multi-speaker theatre system or is on line with a group of twenty or thirty DC-300A's for an outdoor rock session, this amplifier delivers 100%. And we know that in whatever application you use it, the DC-300A will give you the kind of reliability you're looking for.

Output Power: 600 watts minimum RMS into an 8 ohms load over a bandwidth of 1 Hz—20 KHz at a rated RMS sum total harmonic distortion of 0.05%.
Dimensions: 19'' wide by 16½'' deep by 8¾'' high.
Weight: 92 lbs.

The Crown M-600 power amplifier was designed specifically for applications requiring relatively high power levels. The M-600 maintains all the exacting Crown laboratory performance standards, plus featuring built-in cooling for continuous full power operation.

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new thinking about the acoustic design of control rooms: the EVERYTHING AUDIO approach

by BRIAN CORNFIELD and JOHN LYON

The design of studio control rooms has evolved substantially during the past five years, based, perhaps, on the fundamental recognition that the acoustic space of the control room is, indeed, a critical and directly related component of the monitor system. A particular style of control room has emerged which, seemingly, represents the current thinking of a majority of studio designers and builders, and judging by the frequency response data from some of the more recent of these control rooms one could conclude that the age of truly accurate, trustworthy monitoring has finally arrived.

However, a few evidences to the contrary still exist: Why, for example, does one still see successful producers bringing their own speakers into a state-of-the-art control room? . . . Why do loudspeaker manufacturers still complain that their latest innovations fail to perform in contemporary control rooms, as they did in the manufacturers anechoic chamber? . . . And, if everyone agrees that the objective is a monitor system that reproduces the program exactly, without coloration or distortion of any kind, why is there still no satisfactory standard which everyone will accept? . . .

We at EVERYTHING AUDIO believe that the answers to these questions lie in the compromises that have been made, almost totally, for the sake of frequency response. And, upon analysis of many of these contemporary control rooms it, again, seems reasonable to concludethat the overriding design goal was linear frequency response throughout the audible spectrum, at the monitoring positions. The need to provide the engineer as well as the producer, in their respective normal monitoring positions in the control room, with as nearly identical response curves as possible further complicated the problem.

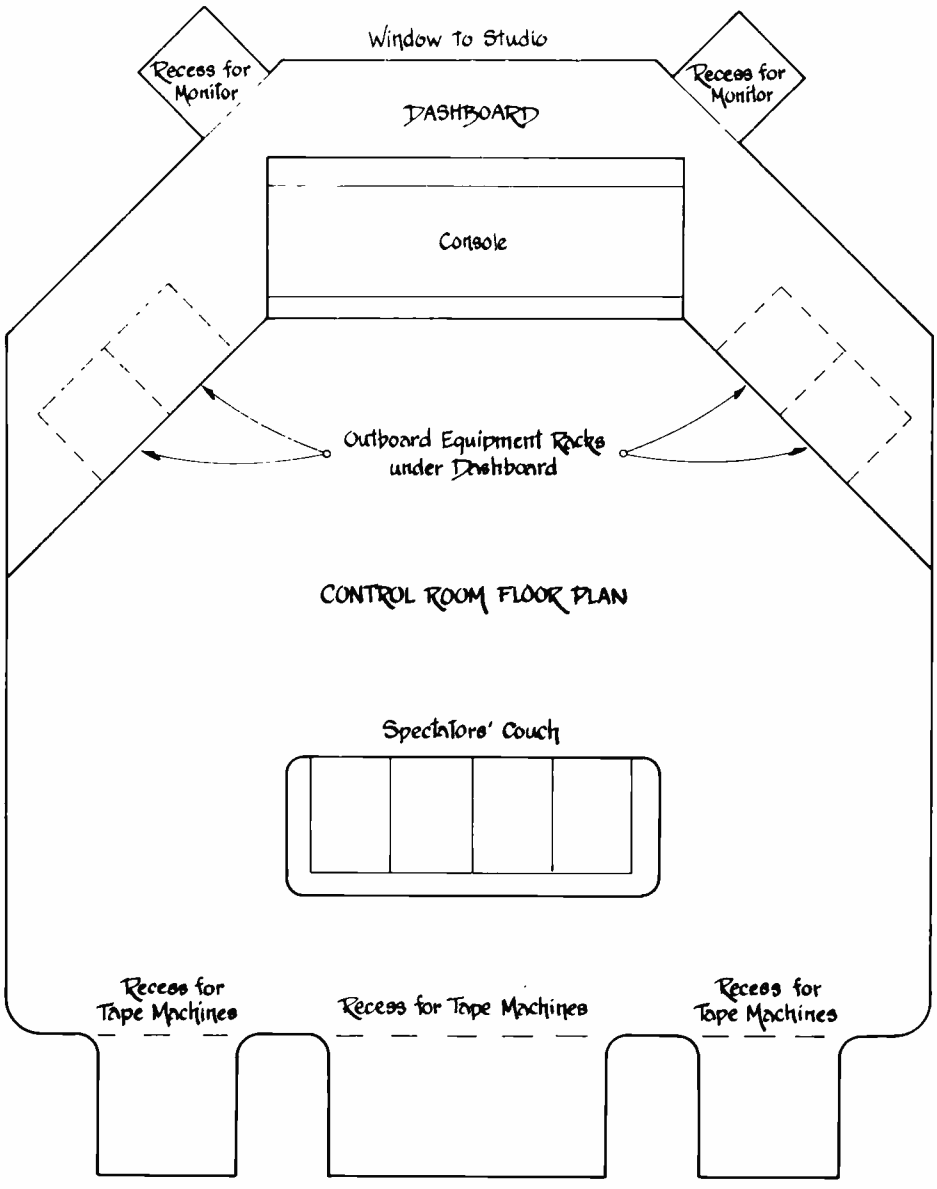
The foregoing, as well as the following review of several other control room design traditions will be helpful in identifying and isolating the reasons for the EVERYTHING AUDIO approach to control room design. Additionally, it has become plainly evident that the many facets to the design of studios and control rooms has created the need for complete knowledge of several associated disciplines to obtain a well informed design outlook. The additional input of this kind of specialized information from a large consulting staff effectively gives EVERYTHING AUDIO the unique ability to grow continuously and take advantage of various new ideas and solutions; unlike a one man system which by definition is limited and tends to settle into a static pattern.

Historically, monitor speakers had been hung on chains above the studio window to provide an unobstructed visual path; little significance was attached to reflections and the low frequency loss due to that mounting method, which, until fairly recently was not yet widely understood. Positioning the console at the speaker manufacturer's recommended listening distance left a space between the console and the window where visitors were inclined to linger, so a couch at that location became a fixture as a matter of course. When the value of flush-mounting the monitors came to be recognized, seeing no reason to relocate them, studio builders simply flush-mounted them where they were . . . above the window. This arrangement creates an acoustical cavity between the monitors and the engineer which marks the beginning of a collection of contemporary low frequency headaches.

Achieving low frequency linearity requires the elimination of standing waves from the listening environment. Standing waves occur when wave trains of the same frequency pass through the same region in opposite directions, such as when a wave is reflected orthogonally by a flat surface or back and forth between two opposite and parallel surfaces. To prevent the latter condition it has long been standard studio construction practice to undulate or otherwise arrange wall surfaces so as to avoid opposing parallel surfaces. Applying this principle to the floor to ceiling mode, contemporary studio builders devised the vee-shaped or *tier drop* ceiling sloping down from above the monitors to a point above the engineer's head. This arrangement has the additional effect of reflecting sound that would have passed overhead back down into the console area from as many as three *tiers*, thereby delivering higher sound pressure levels to the console area. Unfortunately this method also locates a flat reflective surface above and parallel to the console surface itself, thereby accidentally creating exactly the condition it was designed to correct. The resulting low frequency irregularity is a feature of the design itself, and is therefore present to some degree in every control room so arranged. At least one major record label, after having this type of ceiling installed in a pre-existing control room, went to the trouble of having it removed because of the low frequency problems its installation

created. Although it is unfair to judge the builder when he is not allowed the opportunity to alter and coordinate the entire acoustic treatment to suit a single modification, it is interesting to note the correlation between this particular cause and its effect in an essentially controlled experiment. Standing waves occur at particular frequencies and at specific locations or *nodes* in a given room; they cannot be corrected by equalization, because the nodes will continue to exist in the same locations. They must, therefore, be eliminated by design.

The reason for the continued use of the *tier drop* ceiling in spite of its low frequency shortcomings has to do with the effect that it has on high frequency response curves measured by contemporary real time analysis techniques. The methods currently employed to measure the response of a studio control room make no attempt to distinguish between sound waves arriving at the microphone(s) directly from the monitors (the direct field) and those that are reflected to the microphone(s) from the walls, the ceiling, the window or the console surface itself (the reverberant field); the two fields are simply summed. The fact that a high frequency reflection arrives in or out of phase with the direct radiation of the same frequency is unimportant to the measured response curve, although it has variable effects in other ways. What is significant is that reflections often originate from portions of the high frequency driver's dispersion pattern that are far from linear, and from surfaces whose reflection spectrum and dispersion pattern are, at least, irregular. Adding to this the fact that *build-up* of reverberation increases with decreasing frequencies, the result is that even if the monitors and therefore the direct field were absolutely flat, the sum of the direct and reverberant fields would still be irregular, tending to be unacceptably high-end deficient. Following the current practice, the signal fed to the monitors would then be equalized to flatten the response curve of the sum. In the case of a perfectly flat monitor, this would mean introducing errors equal in magnitude but opposite in sign to those measured in the reverberant field. For example, if the monitors were flat but the reverberant field exhibited a high end roll off, the monitors would be over brightened until the sum measured flat. One reason why this practice should be discouraged is a result of research published more than two years ago.[1]



Experimental evidence indicates that the listener identifies the spectrum of a source principally from its direct field rather than from the reverberant field produced by it. Unlike a measuring microphone the human ear is connected to a highly sophisticated computer which is able to evaluate the information it receives in terms of a lifetime of experience listening to sounds in reverberant surroundings and in terms of visual data about the listening environment and thereby arrive at a subjective identification of the timbre of the sound source. The precedence effect in which by analagous means the listener is able to identify directional information with amazing accuracy operates on the same level of subjective interpretation and has been well documented for years. The current technique of monitor system equalization by averaging the direct and reverberant fields is according to these findings, unsuitable for human listeners because it requires altering the accuracy of the direct field from which human ears derive essential information about timbre in order to make up for deficiencies in the spectrum of the reverberant field. This is why a monitor system can sound over-bright or bass deficient to expert ears but still produce electronic measurements that look flat.

But the gap between subjective evaluation and seemingly objective measurement always produces a sense of uneasiness so ways had to be found to introduce larger amounts of high frequency energy into the reverberant field in order to reduce the required amount of corrective equalization. The *tier-drop* ceiling contributed a major reflection of energy back into the console area, but over too narrow a vertical span. A cut and try approach led to a partial solution: by the use of various reflectors and textured surface materials, reflections were diffused into what have come to be called *random sound fields.*

It has been asserted that because of their *randomness* and because they arrive at the console with less than 30 milliseconds of delay, they are not perceived as echos but rather serve to *enhance* the sound. While the question of whether or not this kind of blur enhances the program is largely a matter of taste, the object of a monitor system in the first place is to provide an accurate rendering

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"My business is to compare equipment. When we had an opportunity to use an ES-505 against two XXXXXX's* in our production room, I was very, very impressed — not only with the machine's quality of reproduction, but with its ease of operation and unique features.

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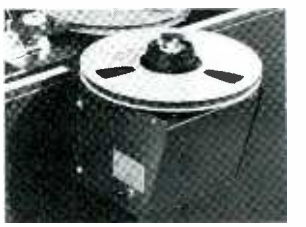

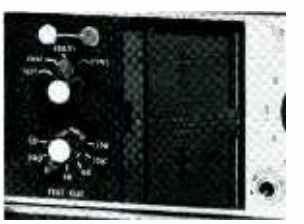

"The emitter follower on the playback head to minimize noise and hum vulnerability is an unusually clever idea. In fact, it's one that's been long overdue in the industry.

"Some exceptional human engineering went into the ES-505. It's not only well built and simple to maintain, but it's completely operator-oriented. For instance, the disappearing head gate which makes for easy editing and head cleaning; and the built-in test oscillator which speeds and simplifies calibration.

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"All in all, the ES-505 is a highly professional recorder/reproducer with a very competitive price tag that makes it a most important addition to our production facility."

Dave Finley

Chief Engineer, KSL-AM/FM, Salt Lake City

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of the program, not an enhanced one.

A playback system that would improve or *enhance* whatever program material was played through it would be valuable indeed, but it would make a poor studio monitor for the same reason. The engineer needs to hear the program as it is, no more and no less.

Elimination of boundary reflections from the listeners area should enable the monitor system to function accurately, so that the engineer can judge the effects of HIS added equalization. Reflected sound arriving at the mixer's position prevents him from accurately knowing how much reverberation is in the program and how much is being introduced by the control room acoustics. To say that the mixer can learn to take control room reverberation into account when he is trying to critically judge how much echo to add to a track, is like saying he can learn to type while wearing mittens. It may be true, but so what? The point is to provide him with a useful tool, not a handicap.

A less obvious but equally important consequence of the presence of a significantly large reverberant field in the console area lies in the extent to which brute force monitoring has come to be used to overcome it. The second major design consideration in most contemporary control rooms is the apparent need to deliver very high sound pressure levels to the engineer. The studio builders responded with bi-amplified massive custom dual-woofer monitors capable of delivering in excess of 110dB each into a room the size of a large living room. Since there are several rather important drawbacks to listening at levels that high, it is necessary to examine why that need was felt. The first and most obvious drawback is the effect on the engineer's hearing. In the long term, such high monitoring levels may produce hearing damage in the form of permanent losses at some frequencies or an overall threshold increase either of which could very well shorten or end his career. This is clearly an area deserving considerable study with the object of establishing standards for absolute level and intermittent duration such as already exist in the area of industrial noise control.

In the short term, ear fatigue will require him to either stop periodically to rest his hearing and restore a normal threshold or, more likely, to turn up the gain of the monitor system to restore apparent sensitivity. It is well known from the Fletcher-Munson curves that when the level increases, the listener hears a subjectively different equalization. The more the level increases, the more the equalization appears to change. Therefore, all the effort to obtain a flat audio response is lost due to our own physiological limitations. So much for the flat monitor system. Meanwhile, under high sound pressure levels, the control room is becoming a low-pitched wind instrument, being played by the control room monitors. The net result is that what we hear certainly isn't the sound the consumer will hear when he listens at home. As we will see in a moment, less widely known research also proves that the perception of dynamics is also affected by changes in listening level.

It should be pointed out that measurements of monitor system linearity and dispersion are not commonly carried out at levels even approaching 110dB. It would in fact be very informative if studio builders would publish the reference levels at which tests are actually carried out along with the usual frequency response data. But aside from the difficulty of maintaining the acoustical performance specifications of the system at such high levels as well as the gradual or traumatic deterioration of transducers resulting from such punishment, there is the more striking fact that not one in a thousand consumers for whom the finished product is intended will ever hear the program nearly that loud!

Why would anyone want to mix that way? Why do engineers and producers feel the need to listen to playbacks at levels so loud that they send acoustic musicians and singers heading for the door with their hands over their ears? Once again the answer lies in the study of the human hearing mechanism. Audiology tells us that in the neighborhood of the threshold of hearing, approximately 40dB SPL, the amount of change in level required for us to perceive a "just noticeable difference" is about 4dB. At very high listening levels, above 100dB, the amount of change required to perceive a just noticeable difference approaches ½dB.[2]

In other words, we are more sensitive to small level differences when the program is loud than we are when it is soft. This means, for example, that if the reverberant field in a control room is nearly as loud at the listener's position as the direct radiation from the monitors, turning up the level can subjectively increase the difference between the direct and reverberant levels, or in other words, it can seem to pull the program up out of the mud. Hence comments such as: "When I turn it up really loud I can hear a lot more detail . . ." etc. In this sense it is meaningful to think of the ratio of direct radiation to reflected sound as an acoustical signal to noise measurement. Those familiar with high intensity indoor concert sound reinforcement will recognize this phenomenon and note that it is considerably abated outdoors. This difference is, of course, the result of the virtual elimination of the reverberant field in the listening environment. Which brings us to Everything Audio's approach to control room design.

One conclusion that is apparent from the preceding analysis is that the recent history of the evolution of control room

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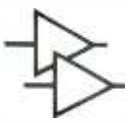

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design is one of individual modifications to older designs in response to individual problems inherent in those designs. The cascading of solutions to problems created by too-narrow solutions to previous problems led to a log-jam of ideas coupling hyper-expensive experimentation with a slavish devotion to outdated convention. The result has too often been an elaborate exercise in the juggling of experimental parameters to produce the desired results on paper that simply doesn't sound very good.

When Everything Audio set out to design an optimum listening environment the first thing we did was to scrap the current approach. Rather than try to make the *tier-drop* ceiling work, or to make the visitor's couch belong in front of the console, we elected to establish the basic list of conditions that must be satisfied, and let the couch fall where it may.

Starting with the monitor speakers, we were already in better shape than might have been expected:

Accurate, smooth reproduction from 35 to 20,000 Hz, ± 3dB. 44dB *SPL at 30 feet with a one milliwatt input. 101dB SPL at 10 feet at one-half rated power input.*

Factory stock monitors are available off the shelf that perform quite respectably in anechoic chambers, certainly well within the limit of minor equalization, at any rate. Why not just make the control room anechoic?

First of all, you can't. Second, even if you could, you shouldn't. The reason that you can't is that there are a certain number of hard reflective surfaces that need to be there: the console and the studio window are the two obvious ones. Two more that are not so obvious are related to the reason why the low frequency response of a speaker system is improved by flush-mounting.

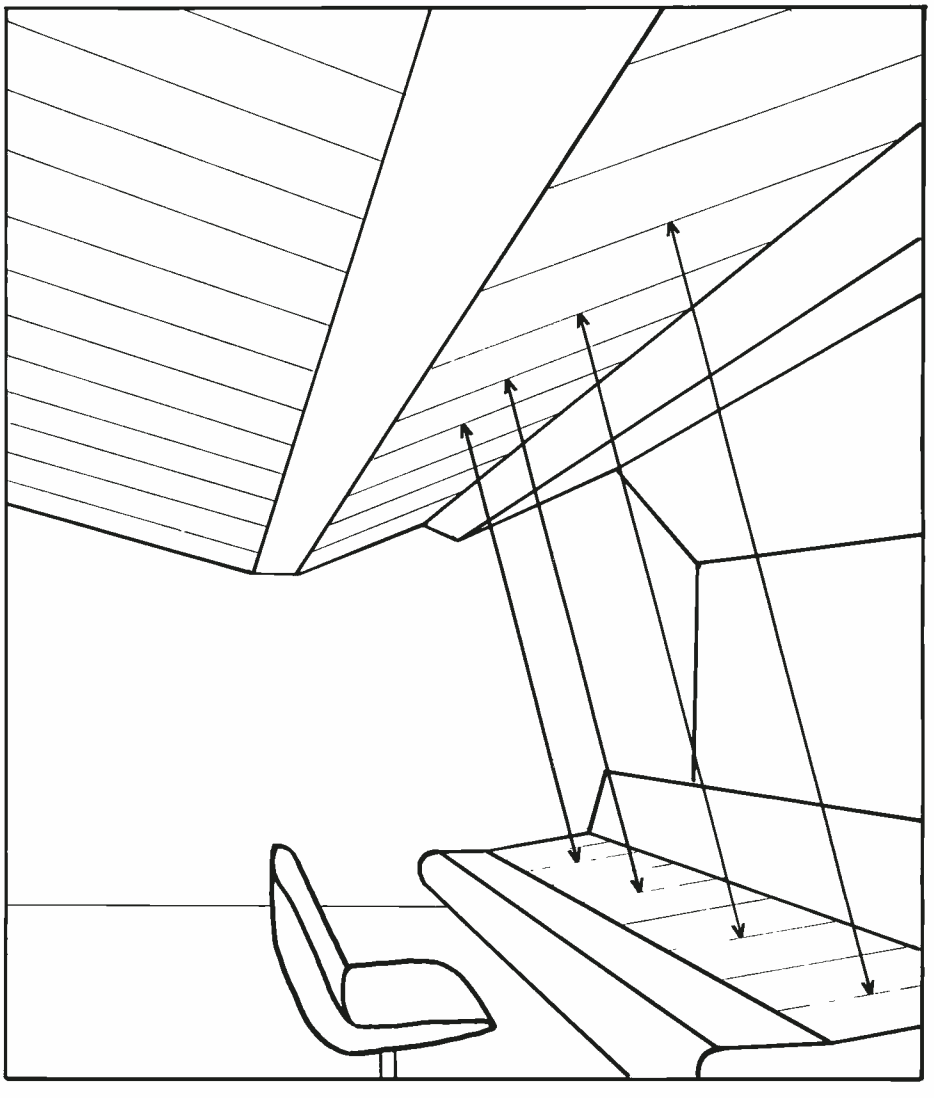
A characteristic of speaker systems in general is that they are more directional at high frequencies than low. Specifically, any speaker enclosure away from room boundaries whether in an anechoic chamber or simply hung on chains will become omnidirectional at a particular low frequency that is a function of its cabinet dimensions, and from that point on down it will radiate half of its energy away from the listener.[3] In the case of speakers hung on chains, that backward radiation will likely strike a wall and return to cause standing wave problems. In an anechoic chamber it is simply lost. If, however, the enclosure is mounted in a wall with its front baffle flush with the surrounding surface, this loss won't occur because the dispersion is confined to the half space on one side of the wall.

Turning our attention to the engineer, it is abundantly clear that all of the information he needs to receive is contained in the direct field radiated by the speakers, and virtually all of the difficulty he has in receiving that information is a result of the reverberant field generated by the room acoustics. So the geometric problem referred to earlier becomes a matter of arranging the console, the window and the two walls containing the flush-mounted monitors in such a way that the engineer sitting at the intersection of the axes of the monitors will receive only direct field radiation with no reverberant interference.

An additional constraint is that if we are able to provide a hard reflective surface between and connecting the monitor walls, at least through the height range of the transducers, the engineer will be able to perceive the exact panned position of any particular track in a stereo mix with greater ease and accuracy because the surface will provide a physical means for wavefronts from both channels to combine without being degraded by surface irregularities.

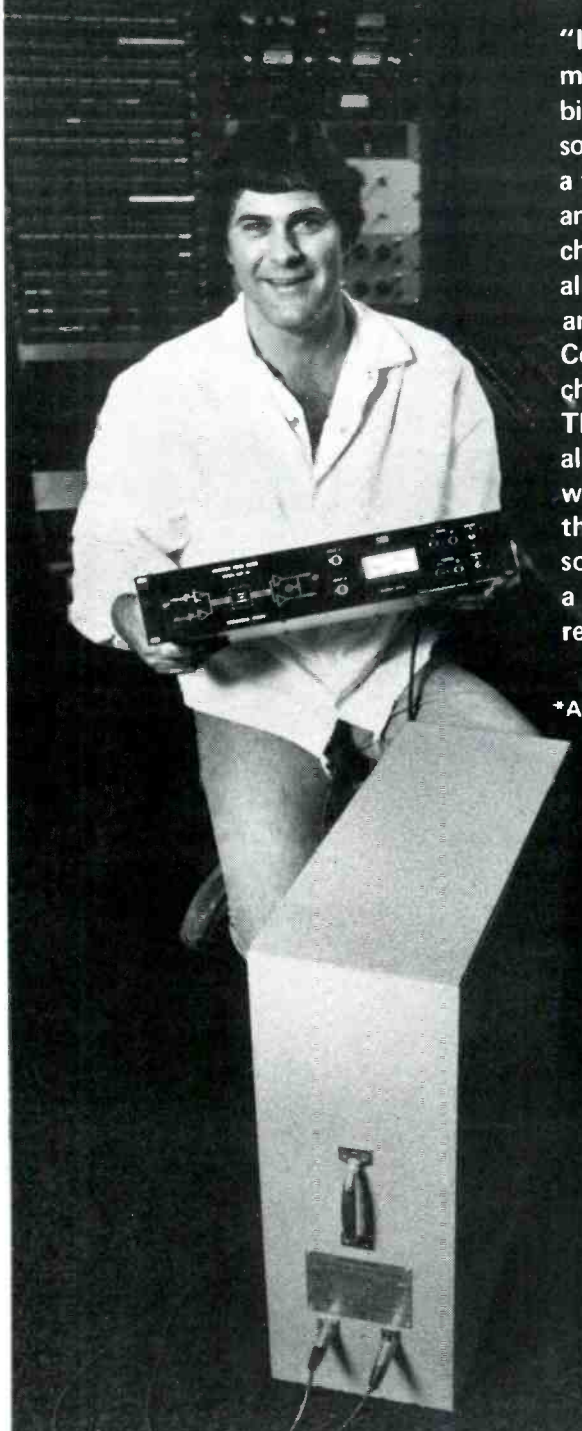
Since the only argument that arose against using the window itself to fill that function was that it would mean mounting the speakers at ear level, (and although that may defy tradition it does seem to be where they belong), we adopted that as a feature of the new design. Still, although simple in concept accurate execution requires careful attention to physics, geometry, materials analysis, acoustic traps and construction.

Two factors determine our perception of directionality, phase difference and intensity difference. At low frequencies intensity differences between our two ears are minor, most of the information about direction at low frequencies comes from phase differences. Unfortunately, phase relationships are a function of where you sit at the console. Even at high frequencies where intensity differences carry the information about directionality, the



REFLECTIONS CAUSED BY PARALLEL SURFACES OF CONSOLE TOP AND "TIER-DROP" CEILING

Don Sciarrotta, Rufus engineer, "tells you something good" about the Cooper Time Cube.



"I think the Cooper delay is something good. It has more of a room sound and it makes the room sound bigger. The electronic delays have an 'electronic' sound. The Cooper's versatile, too. I can use it with a tape machine by itself, or I can put one before it and after it, or I can patch it through my echo chamber. When we recorded the 'Rags to Rufus' album*, we used just the Cooper with the voice bag and the voice at the same time. The double delayed Cooper was used on the voice for the verses and choruses, the bag was Coopered only on the choruses. The result was a great doubled voice sound. The album was gold and also had two gold singles. One was 'Tell Me Something Good'. That's why I like the Cooper Time Cube. With it we got just the sound we wanted. I've had my Cooper for two and a half years without any trouble at all . . . I've replaced one light bulb."

*ABC Records, Bob Monaco Producer

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stereo image will only be accurate for a listener sitting along the center axis of the room.

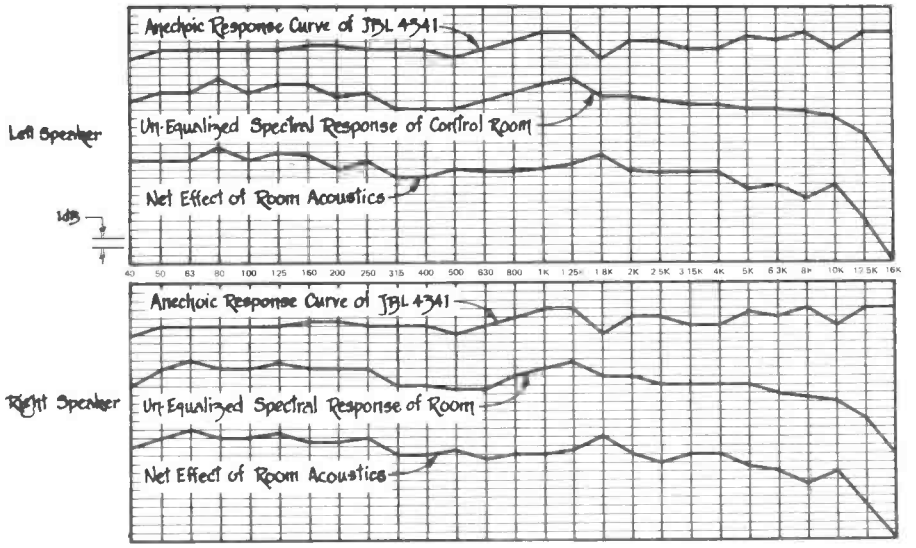
Another problem we faced was the so called *splash-back* of sound striking the back of the console, bouncing to the window and from there into the engineers face. Other studio builders had encouraged console manufacturers to build consoles that were deeper and not so tall. We elected to extend the top surface of the meter panel back to meet the bottom of the window. Extended to either side to meet the walls containing the monitors and dubbed *the dashboard,* this surface prevents, by its acoustical shadow, wavefronts from striking any surface that would cause a reflection to reach the engineer. It additionally provides space for two half-racks of associated electronics on either side of the console similarly shielded from relfecting.

This arrangement required relocating the traditional visitor's couch behind the engineers on a slightly elevated platform where the view is better and the sound more nearly resembles what the engineer is hearing; a point that must be considered since presumably they are at the session to appreciate the performance, both visually and acoustically. The remainder of the room is designed to be as nearly anechoic as budget and prudence allow; in this way it finally becomes possible to have the sound wave pass once and disappear forever.

Two considerations led to the choice of 90° for the angle at which the axes of the monitor speakers intersect. The lesser reason is that scattering (interference) between waves crossing each other at an angle is minimized at 90°. Of more importance is the fact that an angle of 90° is adaptable to and compatible with an idealized quad monitoring situation. Due to the size of that subject, more detailed information on this point must await a future article now in preparation on that subject.

Proven fixtures such as traps, vertical mode attenuating ceilings, flushed-in-machines have remained basically unchanged with the exception of using new plastics for acoustic and structural transmission attenuation, and a marked decrease in construction costs.

A studio of this design is now under construction in Claremont, California. Called aptly enough *Thee Studio,* it was near enough to completion by the deadline for this article to allow a preliminary frequency response measurement. At the request of the client we rolled off the high frequency response above 9500 Hz by means of the cross-over networks in the stock JBL 4341 monitors chosen for this installation. Below this frequency the system exhibits a response prior to equalization of ± 2dB, from 40Hz to 8kHz. This measurement includes the deviations inherent in the 4341 monitor which ac-



FREQUENCY RESPONSE DATA FOR CONTROL ROOM OF "THEE STUDIO", CLAREMONT, CA.

cording to JBL average ± 1¾dB over the same frequency range. This means that for once the accuracy of the acoustic environment is comparable to that of monitor speakers. The horizontal coverage measuring one speaker at a time at 10kHz was ± 1½dB at 48 inches above the floor from end to end of the console; vertical coverage was ± 1½dB from 6 inches above the console surface to six feet above the floor at console center.

NEW CONCEPTS

Ideas under analysis and development at this time include: video interface as an additional source of revenue for the professional studio owner, multi-focus control rooms, no compromise quad mixing, mastering and mobile studios, velocity as well as sound pressure measurement techniques, new money-saving construction ideas and more, soon to be discussed in subsequent articles.

Everything Audio will continue to analyze and study the acoustical field in order to constantly improve *our industry*. Working in a total package capacity we research all new product lines as well as acoustic phenomena, and review our installation and construction techniques and people to keep ourselves in the *second generation* of studio design and construction. At Everything Audio we obtain satisfaction through knowing that our efforst have created a more realistic, creative tool for the people in our industry.

[1]Daniel Queen "Relative Importance of the Direct and Reverberant Fields to Spectrum Perception" J.A.E.S. March '73

[2]H. Davis & S. Silverman "Hearing and Deafness" Holt, Rinehart 1970

[3]Roy F. Allison "The Influence of Room Boundaries On Loudspeaker Power Output" J.A.E.S. June 1974

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Producer: PAUL McCARTNEY

Recorded at: EMI STUDIOS, London
ABC STUDIOS, Lagos, Nigeria

Mastering Studio: CAPITOL RECORDS, Hollywood
Engineer: WALLY TRAUGOTT

Album: BAND ON THE RUN
Paul McCartney & Wings (Apple)

BEST ENGINEERED RECORDING *(Classical)*

Engineer: **KENNETH WILKINSON**

Producer: DAVID HARVEY

Recorded at: KRANNERT CENTER, Univ. of Illinois (Urbana)

Mastering Studio: DECCO STUDIOS (London)

Album: Berlioz: SYMPHONIE FANTASTIQUE
Georg Solti conducting the
Chicago Symphony Orchestra

BEST PRODUCER

THOM BELL

Singles: MIGHTY LOVE (Atlantic) – The Spinners
LOVE DON'T LOVE NOBODY (Atlantic)
– The Spinners
THEN CAME YOU (Atlantic)
– Dionne Warwicke & The Spinners
YOU MAKE ME FEEL BRAND NEW (Avco)
– The Stylistics
I'M COMING HOME (Columbia)
– Johnny Mathis

Albums: MIGHTY LOVE (Atlantic) – The Spinners
ROCKIN' ROLL BABY (Avco) – The Stylistics

SUNN ELECTRIFIES 'CIRCUS MAXIMUS'

sunn MAGNA 5000!

At SUNN, we've been making professional sound reinforcement products for years. Our lab has been the concert stage.

After an extended evaluation of the specifications, utility, and versatility of our MAGNA 5000 control console, Caesar's Palace in Las Vegas installed it as part of the new sound system in the Circus Maximus showroom.

Modular in design, easily interfaced with all state-of-the-art electronic components, the MAGNA 5000 was the obvious choice of Caesars management to meet the exacting demands of the regular bill of superstar performers that appear there.

We invite you to check further into the MAGNA 5000 console. Write us here in Oregon or check out your local dealer or contractor.

sunn MAGNA

Amburn Industrial Park • Tualatin, Oregon 97062

Circle No. 133

BEST ENGINEERED RECORDING NOMINATIONS
(Non-Classical)

Engineers: **KEN SCOTT**
JOHN JANSEN

Producer: KEN SCOTT
SUPERTRAMP

Recorded at: TRIDENT (London)
RAMPORT STUDIOS (London)
SCORPIO SOUND (London)

Album: CRIME OF THE CENTURY
Supertramp (A&M)

Engineer: **BILL SCHNEE**

Producer: LINCOLN MAYORGA
DOUG SAX

Recorded at: PRODUCERS' WORKSHOP
(Hollywood)

Mastering Studio: THE MASTERING LAB
(Hollywood)
Engineer: ARNIE ACOSTA

Album: LINCOLN MAYORGA and
DISTINGUISHED COLLEAGUES,
VOLUME III
Lincoln Mayorga (Sheffield)

Engineers: **TOMMY VICARI**
LARRY FORKNER

Producers: GINA VANNELLI
JOE VINNAELLI

Recorded at: A&M RECORDS (Hollywood)

Mastering Studio: A&M RECORDS (Hollywood)
Engineer: VERNIE GRUNDMAN

Album: POWERFUL PEOPLE
Gino Vannelli (A&M)

Engineers: **RIK PEKKONEN**
PETER GRANET

Producer: STEWART LEVINE

Recorded at: WALLY HEIDER STUDIO 3
(Hollywood)

Album: SOUTHERN COMFORT
The Crusaders (Blue Thumb)

BEST ENGINEERED RECORDING NOMINATIONS
(Classical)

Engineers: **BUD GRAHAM**
RAY MOORE

Producer: THOMAS Z. SHEPARD

Recorded at: COLUMBIA RECORDS
30th Street Studio (New York)

Album: Bernstein: CANDIDE
John Mauceri conducting the
Original Cast (Columbia)

Engineers: **STANLEY TONKEL**
RAY MOORE
MILT CHERIN

Producer: ANDREW KAZDIN

Recorded at: COLUMBIA RECORDS
30th Street Studio (New York)

Album: Copland: APPALACHIAN SPRING
Aaron Copland conducting the
Columbia Chamber Orchestra
(Columbia)

(Classical) (continued)

Engineers: **ROBERT AUGER**
(recording)
PAUL GOODMAN
(mastering)

Producer: MAX WILCOX

Recorded at: WALTHAM STOW TOWN HALL
(London)

Album: Ives: SYMPHONY NO. 4
Jose Serebrier conducting
the London Philharmonic Orchestra
(RCA)

Engineers: **MARC AUBORT** (Elite Recording)
JOANNA NICKRENZ

Producers: MARC AUBORT
JOANNA NICKRENZ

Recorded at: RUTGERS CHURCH (New York City)

Mastering Studio: STERLING SOUND (New York)

Album: PERCUSSION MUSIC
The New Jersey Percussion
Ensemble (Nonesuch)

Engineer: **ANTHONY SALVATORE**
(Recording & Mastering)

Producer: RICHARD MOHR

Recorded at: WALTHAM STOW TOWN HALL (London)

Album: Puccini: LA BOHEME
Georg Solti conducting the
London Philharmonic Orchestra (RCA)

Engineer: **ISAO TOMITA**
(recording)
EDWIN BEGLEY
(mastering)

Producer: ISAO TOMITA

Recorded at: PLASMA STUDIOS (Tokyo)

Album: SNOWFLAKES ARE DANCING
Isao Tomita (RCA)

BEST PRODUCER NOMINATIONS

RICK HALL

Singles: YOU'RE HAVING MY BABY
– Paul Anka (UA)
ONE HELL OF A WOMAN
– Mack Davis (Columbia)

Album Track: AS LONG AS HE TAKES CARE
OF HOME
– Candi Staton (Warner Bros.)

BILLY SHERRILL

Singles: WOMAN TO WOMAN
– Tammy Wynette (Epic)
THE GRAND TOUR
– George Jones (Epic)
I LOVE MY FRIEND
– Charlie Rich (Columbia)

Albums: VERY SPECIAL LOVE SONGS
– Charlie Rich (Columbia)
WOULD YOU LAY WITH ME
– Tanya Tucker (Columbia)

LENNY WARONKER

Singles: MIDNIGHT AT THE OASIS (Reprise)
Maria Muldaur (co-produced with
Joe Boyd)
CAREFREE HIGHWAY
– Gordon Lightfoot (Reprise)

Albums: WAITRESS IN A DONUT SHOP (Reprise)
–Maria Muldaur (co-produced with
Joe Boyd)
SUNDOWN (Reprise) Gordon Lightfoot
GOOD OLD BOYS (Reprise) Randy Newman
(co-produced with Russ Titelman)

A call to one of these guys will tell you more about 3M pro audio gear than all the ads we've ever run.

3M PRO AUDIO DEALERS

RECORDING STUDIOS

Ron Newdoll
Accurate Sound Corp.
114 5th Avenue
Redwood City, CA 94063
Tel. (415) 365-2843

Welton Jetton
Auditronics Systems Division
207 Summit Street
Memphis, TN 38124
Tel. (901) 272-9641

Bill Wilson
LaSalle Audio
740 N. Rush
Chicago, IL 60611
Tel. (312) 266-7500

Bruce Martin
Martin Audio/Video
320 West 46th Street
New York, NY 10036
Tel. (212) 541-5900

Rick Maze
The Maze Corporation
514 Alton Road
Birmingham, AL 35210
Tel. (205) 591-4800

Jerry Milam
Milam Audio Corporation
700 West Main Street
South Pekin, IL 61564
Tel. (309) 348-3112

Tom Hidley
Westlake Audio
6311 Wilshire Blvd.
Los Angeles, CA 90048
Tel. (213) 655-0303

GSA CONTRACTS

Lee Deane
Fidelity Sound Co., Inc.
1200 18th St., N.W.
Washington, D.C. 20036
Tel. (202) 296-9110

TELEVISION APPLICATIONS

Byron Fincher
RCA Broadcast Systems
Camden, NJ 08102
Tel. (609) 963-8000

If you know the recording industry, you know at least one of the men on the sheet above. And you know you can count on getting straight answers from them.

So if you want to know where 3M is really at in this business, these guys can tell you.

The fact is, 3M audio equipment has worldwide acceptance, and we back it up with a superb organization, our Business Products Service, with offices in New York City, Memphis, Washington, D.C., Detroit, Chicago, San Francisco, Los Angeles, Atlanta, Pittsburgh, Providence and Grand Prairie, Texas.

And we have a special flying squad of troubleshooters that we can dispatch from our St. Paul headquarters on a moment's notice for big problems. So when it comes to service, 3M leads the industry. Something you might want to keep in mind when some other companies have been cutting back on their service capabilities.

And we think it's great when the guys in the sales organizations around the country praise 3M equipment.

Because we think word-of-mouth advertising is the best kind there is.

Our ad agency hates it, though.

It's not commissionable.

PROFESSIONAL AUDIO PRODUCTS
Mincom Division 3M COMPANY
3M CENTER · SAINT PAUL, MINNESOTA 55101

Circle No. 134

NEW SCULLY 280B PROFESSIONAL RECORDERS FEATURE SERVO DRIVE, 14—IN. REEL CAPACITY

A new series of Scully professional tape recorders with DC capstan servo drive motors and reel capacity up to 14-in. has been introduced by Scully/Metrotech Division of Dictaphone Corporation. The Scully 280B, 284B and 284B-8 models are designed to provide recording studios and broadcasters with operational conveniences and easy set-up and maintenance.

Printed DC capstan servo drive motors, offered as an optional feature, highlight the new Scully recorders which also have speeds to 30 in./sec., plus 14-in. reel capacity. According to the manufacturer, these features respond to the trend to 30 in./sec. speed for mastering on 1.5 mil tape, and eliminate the weakness of the 10½ in. reel of tape, which is limited to only 16 minutes of total program time. The new 14-in. reel doubles this to 32 minutes. Now mix-downs of a full 30-minute program can be made on the Scully 284B Series.

The recorders also have new motion-direction sensing control logic circuitry, including dynamic braking and functionally illuminated pushbutton controls. These permit the operator to observe the control logic functions either at the recorder, or at his accessory remote control. Easy access to set-up controls is possibly by pullout electronics drawers. With the exception of the Model 284B-8 eight-channel, 1-in. recorder, these products can be either rack or console mounted. Besides DC servo drive, the recorders are available with AC Hysteresis motors. A variable speed system also is offered with L.E.D. display of actual tape speed. This accessory may be locked to an internal crystal reference or to a control track by pushbutton selectors.

monophonic, two- or quarter-track stereo, 1/4-in. transports for up to 11.5-in. or 14-in. reels; plus quad, four-track stereo 1/2-in. and eight-track 1-in. systems.

Dictaphone's Scully / Metrotech Division also manufactures a full line of voice logging recorders with tape speeds as low as 5/16 in./sec.

SCULLY/METROTECH, DEPT. 280B, 475 ELLIS STREET, P.O. BOX 1298, MOUNTAIN VIEW, CA 94042.

Circle No. 135

NEW ATTENUATE-ONLY 1/3 OCTAVE ROOM EQUALIZATION FILTER SET

The UREI Model 529 Room Equalization Filter Set is an *Attenuate-only* version of their popular Model 527-A Graphic Equalizer. The 529 has 27 1/3 octave active filters that each provide up to 15dB of attenuation to adjust for a desired house curve. They are centered on standard ISO frequencies making the 529 compatible with all current room equalization measuring equipment. Filters combine properly for minimum ripple and phase shift. *Band end* tunable high pass and low pass filters attenuate 18dB per octave. Gain is adjustable to +20dB to make up for average level lost in equalization. The 529 is completely self contained with built-in power supply. An accessory security cover (529 SC) may be added to prevent unauthorized tampering or accidental mis-adjustments.

UREI, 11922 VALERIO STREET, NO. HOLLYWOOD, CA 91605

Circle No. 137

AKG PORTABLE REVERBERATION UNIT WITH TRUE STUDIO PERFORMANCE

The AKG BX-10 is the first truly portable two-channel reverberation unit designed to provide both the quality and operating features required in studio ap-

Circle No. 136

plications. The unit is based on the extremely successful Torsion Transmission Line principle used in the larger well-known AKG BX-20.

Independent decay-time adjustment, high and low frequency equalization and reverberation/dry signal mixing are provided for each of the two electronically and acoustically separate channels. Decay-time is adjusted silently through motional feedback, enabling adjustment during program material. Reverberation/dry signal mixing enables reverberation to be added to dry signal without the need for a *reverb return* mixing section in the mixing desk.

The unit may be rack mounted in the control room or used as a portable system. Its unique two-point suspension makes it impervious to acoustic feedback and mechanical vibration. There is no need to *lock-down* the system for transport, nor any need for adjustments after transport.

PHILIPS AUDIO VIDIO SYSTEMS CORP., ONE PHILIPS PARKWAY, MONTVALE, NJ 07645.

Circle No. 138

API'S MINIMAG II, A NEW LOW COST SYNCHRONIZER EMPLOYS SMPTE STANDARD CODE*

A new SMPTE Standard Time and Control Code* Synchronizer/Code Generator, named Minimag II, is being marketed by Automated Processes, Inc. Priced at only $4950, the new unit is said by the company to be less than half the cost of comparable SMPTE Standard Time and Control Code* Synchronizers.

Minimag II is a second-generation design that incorporates every essential function for automatically synchronizing and interlocking any two audio or video multitrack tape machines. It is recommended for audio *sweetening*, for simulcasting TV and stereo FM, for remote overdub recording, and for combining an audio mix with a visual medium.

The SMPTE Standard Time and Control Code* generated by Minimag II is recorded on both the master and the slave units. This can be done before, or at the same time as, program material is recorded. The synchronizer will then automatically compare the code played back from the slave tape with the code on the master tape, and issue a capstan control signal to lock both tapes in perfect sync. Tape machines designed for DC servo operation are controlled directly; synchronous motor machines are controlled through an optional capstan drive amplifier.

Tapes need not be in alignment before starting, because Minimag II has a 24-hour capture range. A thumb-wheel switchbank on the front panel permits offsetting or shifting between tapes to achieve lip sync or delay effects. This offset may be as much as ±49.9 seconds in .1 frame increments.

All functions, such as fast forward, rewind, stop, and play, are slaved causing the slave machine to move precisely as commanded by the master tape. Switches provide for 30, 29.97, or 25 frame rates; and either 115V or 230V operation at 50-60Hz, at the user's option. The internal code generator can be preset to any time of the day and position readout for both master and slave are provided.

Minimag II is compact, measuring 3½" x 19" x 15" deep, and may be mounted on standard RETMA racks.

**American National Standard Time and Control Code, C 98.12-1975.*

AUTOMATED PROCESSES, INC., 80 MARCUS DR., MELVILLE, NY 11746.

Circle No. 139

PHOENIX SYSTEM'S TRANSVERSAL FILTER

A new state-of-the-art feed forward transversal filter is available from Phoenix Systems for achieving real-time phasing and flanging effects, comb filtering, and pseudo-stereo.

The Model 18 is a solid state analog delay system in a 1¾" x 9½" x 5" half rack package with a frequency response of ±1 dB from 20 Hz to 20 kHz at less than .5% THD with input and output capability of +15 dBm at 600 ohms. The signal time delay which generates the phasing effect is variable from .2 ms to 5 ms (1800° of phase shift @ 1 kHz) either manually, automatically via an internal control oscillator, or remotely by a 1V-10V signal. The automatic mode phasing rate is variable from 3 to .3 Hz and any number of units may be interconnected for multiple channel operation. An LED overload indicator monitors peak input levels and noiseless switching is incorporated throughout.

PHOENIX SYSTEMS, 21 BRIDGE SQ., WESTPORT, CONN. 06880

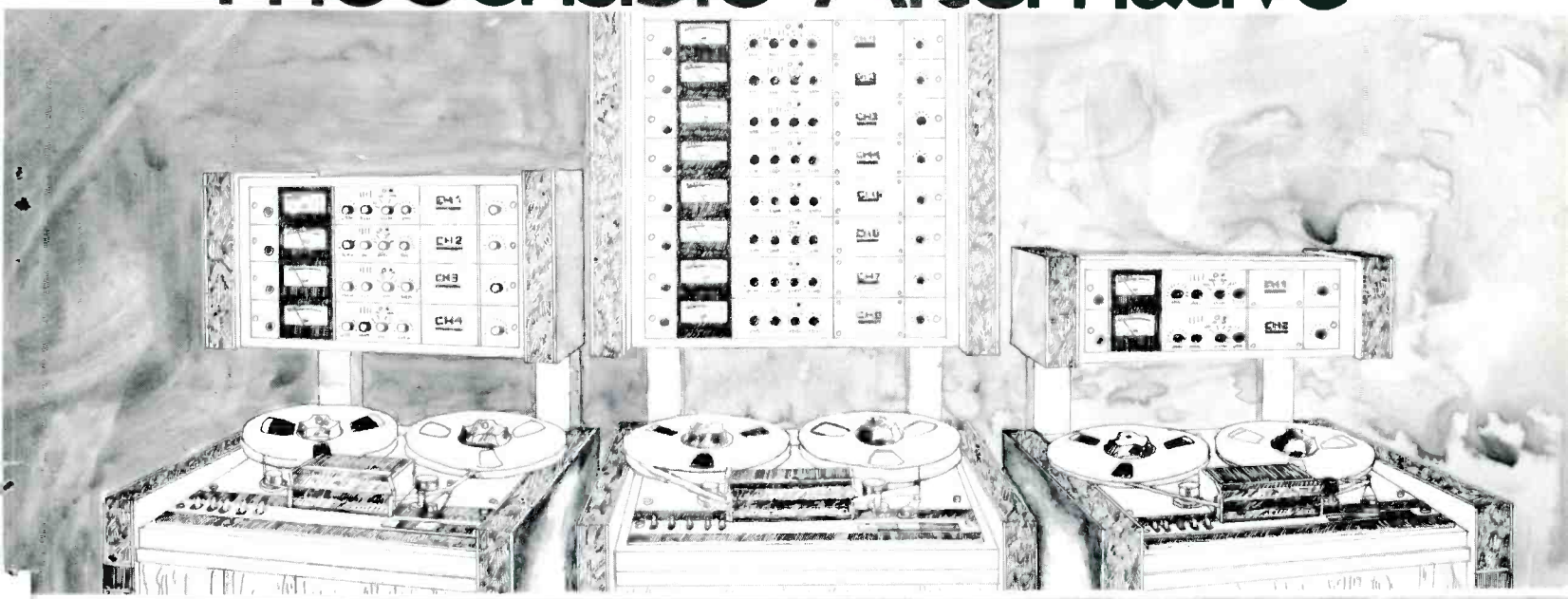
Circle No. 140

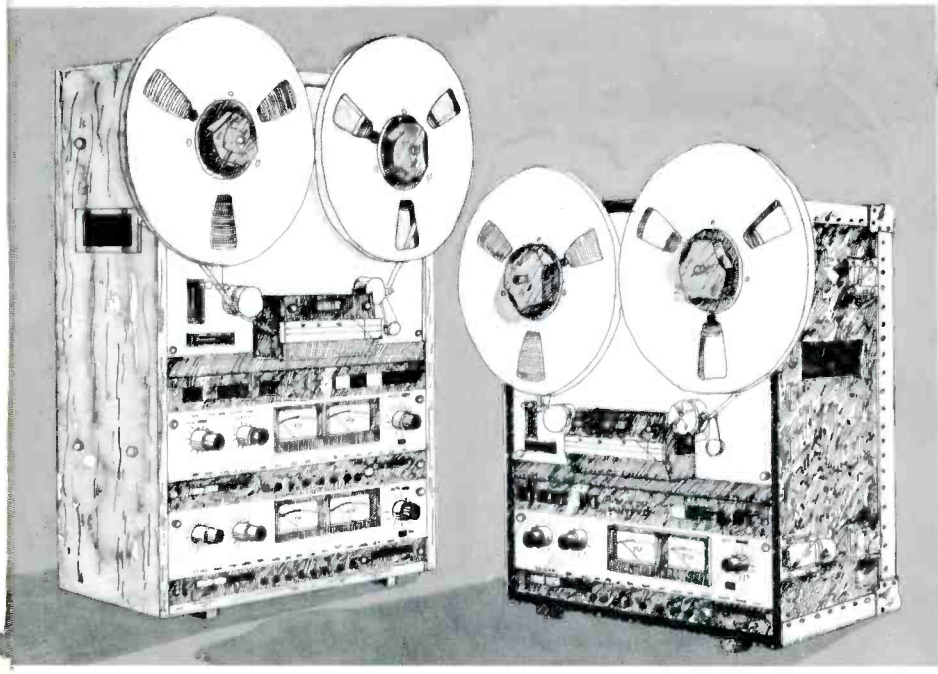
TWO NEW PM MIXERS FROM YAMAHA

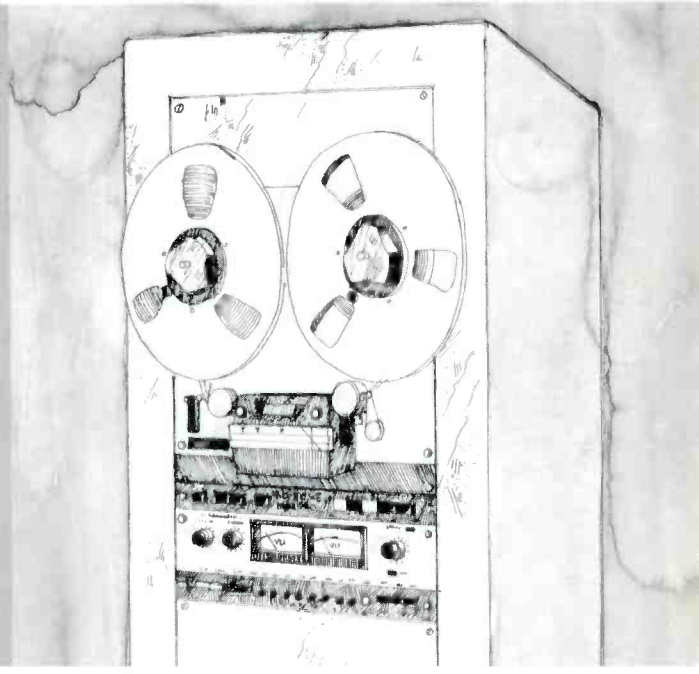
PM-400B: An 8-channel mixer having 2 transformer-isolated balanced output channels equipped with XLR inputs and outputs to help reduce hum and noise, improving the signal-to-noise ratio. Input channel configuration is the same as the

PM-300. The output signal is available at balanced 600-ohm XLR connectors or 70-ohm standard phone jacks. The 600-ohm, line and echo outputs are provided with level selector switches to change the nominal output level for optimum compatability with a variety of equipment. The VU meters can be switched to monitor the master, line or echo buses and are

The Sensible Alternative







Otari Professional Recorders

Otari's wide line of studio and compact tape machines are the Sensible Alternative in professional recorders. Entirely new for 1975, Otari's MX-7300 and MX-5050 have all the professional features and performance of the more expensive recorders, yet sell at a sensible price. And what's more, Otari recorders are backed by a nationwide parts and service network that you can count on to keep you operating.

Check the features. Check the performance. Then specify Otari, the sensibly priced alternative in professional recorders.

Otari Professional Features — All Models. • Motion sense control logic • Front panel edit and cue • Selective synchronization • DC capstan servo option • 600 ohm +4 dB output and XL connectors • Built-in test frequencies and standard reference level. **MX-7300 Series Studio Recorder** • Three electrical switchable speeds on one, two, and four channel • Two speeds (7½/15 or 15/30) on eight channel. Eight channel has compatible one-inch format and optional remote synchronous reproduce. **MX-5050 Compact Recorder** • All professional features and performance listed above • Two or four channels • 7½/15 or 3¾/7½ • Cabinet, portable, or rack mount.

OTARI

Otari Corporation
981 Industrial Road San Carlos, CA 94070
(415) 593-1648 TWX: 910-376-4890

In Canada:
Noresco

R-e/p 67

Circle No. 141

calibrated so 0 VU indicates +4dBm with the level selector at +4dBm. Weight of the PM-400B is 28.7 pounds.

List price is $795.

Circle No. 144

PM-1000: The PM-1000 is a 16-channel input, 4-channel output mixing console. It features modular construction, transformer-isolated XLR connectors for main inputs and outputs, provision for direct playback of a 4-channel tape deck, 2-frequency oscillator for reference levels and testing, talkback mic input and switchable 48 VDC phantom power for condenser microphones. Each input channel is equipped with an 11-step input level switch, conductive-plastic fader control,

3-knob, 5-frequency equalizer, high pass filter, phase reversal switch, pan pot, indicator-type bus assignment switches, cue, and echo mix controls. The four output level VU meters are switchable for master and monitor buses; two smaller VU meters monitor the echo outputs. All meters indicate 0 VU at +4dBm. The PM-1000 weighs 110 pounds.

Retail price is $6600.

YAMAHA INTERNATIONAL CORP., P.O. BOX 6600, BUENA PARK, CA 90620.

Circle No. 145

ROBINS/FAIRCHILD COFFEE PROOF ROTARY-SLIDER

The advantages of a sealed, rotary attenuator and the compact convenience of a slider are combined in a rugged, Rotary-Slider for professional audio mixing.

The new unit, Model RS1000 (patent applied for), was described as the first in a planned line of comparable slide devices, such as panners and D.C. remote controls.

It was engineered to meet the need for low-cost, high-reliability slide faders in both monaural and stereo applications. Among ROTARY-SLIDER's unusual features are two miniature, rotary resistive elements driven by a common slide mechanism.

Use of high-quality, readily-available 1,000-ohm audio taper variable resistors was said to assure accurate, uniform attenuation characteristics, close tracking between stereo channels and compatability with solid-state mixing circuits.

The mechanical working mechanism is simple and rugged to provide continuous,

trouble-free operation without the maintenance ordinarily expected for conventional slide faders. Because the "pots" themselves are sealed, they are dust, dirt and coffee-proof.

The unit is available with or without a single-pole, double-throw micro switch for cue (Model RS1000Q). A standard-size escutcheon panel, with scale calibrated in dB's of attenuation, is also available.

The RS1000 was developed specifically for use in the new ROBINS/FAIRCHILD line of compact, economical, 3000 Series broadcast consoles. It is now being used in many of the company's custom audio consoles and will be sold as a separate component through distributors and to end users and sound contractors for newly-built consoles and to replace faders in existing consoles.

The RS1000 is also available to OEM's for use in their slide-fader consoles.

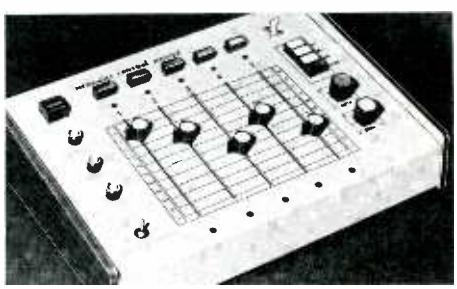
ROBINS/FAIRCHILD devices in development with the "Rotary-Slider" concept include the LUMITEN series of light controlled remote attenuators and such other applications as remote-control of VCA's and lighting dimmers.

ROBINS / FAIRCHILD, 75 AUSTIN BLVD., COMMACK, NY 11725.

Circle No. 146

FX INC. ANNOUNCES MEMORIES CONTROL MASTER

MEMORIES CONTROL MASTER is a hand held control center used with the Allison Research MEMORIES LITTLE HELPER. The unit is part of a package of accessories designed by FX to complement Memories Little Helper.



Features of Memories Control Master include remote master status controls, five programmable group master faders, five programmable group master mutes, programmable automatic panner, programmable vibrato and tremelo effects generator.

Dimensions: 10"x7-7/8"x3-3/8".

FX INCORPORATED, 212 N. 12 ST., PHILADELPHIA, PA 19107.

Circle No. 147

MICMIX 'STUDIO B' SERIES REVERBERATION CHAMBERS

STUDIO B is a new concept in reverberation and features an internal/external mix capability which means it can be used either with or without an audio console. A 5-frequency peak type equalizer is included, together with an EQ gain control which provides up to 10dB of boost at the selected frequency. Input and output are 600 ohms balanced and floating but the input can also be selected by the user for a 5K bridging impedance. Input level is control variable for levels from +4 to +16 dBm and the unit will accept peaks up to +25 dBm without overload.

Incorporating the *natural chamber* sound characteristics of its counterpart, the original Master-Room chamber, STUDIO B models also include variable decay and reverb gain controls, both of which may be operated remotely through an accessory unit. Electronics for the STUDIO B model also incorporates a VU meter and is packaged in a 1¾" rack panel while the small sound column may be mounted in any convenient location up to 10 feet away. Ambient sound levels cause no problem to the Master-Room in monitor areas. XLR type audio connectors and 'Jones' control connectors are utilized and the unit operates on 115/230 volts, 50-60Hz, requiring only 6 VA of power. **MICMIX AUDIO PRODUCTS, INC., 9990 MONROE DRIVE, SUITE 222, DALLAS, TX 75220.**

Circle No. 148

SUNN HIGH POWER COMPRESSION DRIVER ASSEMBLY

The Professional Audio Division of Sunn Musical Equipment Company announces the development of a new 8 ohm, high powered (100 watt rms — 200 watt program) compression driver with a frequency response of 500-16,000 Hertz, and capable of producing 112 dB SPL from 1 watt at 4 feet on axis. The unit has a two inch throat opening and may be affixed to horns utilizing the four bolt method of mounting.

Available from stock — $225.00. **PRO-AUDIO DIVISION, SUNN MUSICAL EQUIPMENT COMPANY, AMBURN INDUSTRIAL PARK, TUALATIN, OREGON 97062.**

Circle No. 150

continued . . .

EVENTIDE INSTANT FLANGER – MODEL PL–201

The FLANGER reproduces electronically the phasing effect obtained by simultaneously recording program material on two tape recorders, and mixing the outputs while physically varying the speed of one of the tapes. The FLANGER utilizes a charge coupled time delay unit to produce a true delay in the signal path, thus generating many more nulls in the frequency spectrum than does the conventional Phaser which employs analog phase shift networks.

Like its predecessor, the Eventide Instant Phaser, the Instant Flanger is fully studio compatible, with internal power supply and high level inputs and outputs. It can produce a striking pseudo-stereo effect, and phasing which does not disappear when mixed to mono. NEW features include the ability to combine the effect of the oscillator, envelope follower, manual control, and remote control in any combination. An exclusive Depth control allows selection of additive phasing, true doppler shift, and subtractive phasing, and a new *Bounce* control effectively simulates the mechanical effect of motor or servo *hunting* that gives tape flanging its distinctiveness.

High reliability light emitting diodes indicate which functions are in operation, as well as power and IN/OUT switch position. All circuit boards are of plug-in construction and are readily accessible. The unit mounts in a standard 19 inch rack, requiring 3½" panel space. It requires 115VAC or 230VAC nominal for operation.

EVENTIDE CLOCKWORKS, INC., 265 WEST 54 ST., NEW YORK, NY 10019.

Circle No. 151

ALLEN & HEATH CONSOLE FEATURES PARAMETRIC EQ

Audiotechniques, Inc., Stamford, Ct., announces the immediate availability of the new Allen & Heath Modular Series audio mixing consoles. The new consoles by the well-known British manufacturer have been designed to economically meet the requirements of the new or expanding multi-track recording studio. The consoles are offered in a variety of configurations from eight input/two output at $4,695 to sixteen input/eight output for $6,795. The totally modular system of construction permits immediate expansion as required.

Design features of the Allen & Heath Modulars include full TT series patch bay, three range equalization with parametric mid range, complete separate monitor mix section with overdub/sync capability,

phantom powering, variable frequency oscillator, cue mix, echo and cue sends, talkback/slate and other features usually found only on the most expensive mixing consoles.

The Allen & Heath Modular Series consoles are described in an attractive four page brochure, available from a number of professional audio dealers or from Audiotechniques, Inc.

AUDIOTECHNIQUES, INC., 142 HAMILTON AVE., STAMFORD, CONN.

Circle No. 152

COMMUNITY LIGHT & SOUND LONG-THROW HF HORN

Designated the EC150, the unit is a 35°, 146Hz flare rate longthrow high frequency horn. As with all CL&S horns, construction is of rugged fiberglas which is said to be impervious to weathering and rough handling.

Large mounting flanges at both front and rear facilitates easy installation.

Tight pattern control at operating frequencies (1.5kHz pattern: 35°V X 35°H) makes the EC150 ideal for outdoors.

The EC150 is available in five throat sizes that will fit all commercially available HF drivers.

COMMUNITY LIGHT & SOUND, INC. 5701 GRAYS AVE., PHILADELPHIA, PA 19143.

Circle No. 153

NEW SHURE SM82 LINE-LEVEL MICROPHONE IDEAL FOR REMOTES

The SM82 microphone contains its own line-level amplifier, peak limiter and 9.8V battery, making it ideal for single-channel remotes where simple, space-saving audio equipment is necessary. Its balanced line-level output can drive telephone lines or other line-level inputs.

The Model SM82's line-level amplifier allows up to a mile of unshielded cable to be used between the microphone and the broadcast equipment without equalization, while the built-in peak limiter makes the SM82 ideal for play-by-play sports coverage by preventing overloading of the remote broadcast amplifier.

Developed after extensive in-the-field and broadcast studio testing, the Shure SM82 features a wide-range frequency response (40 to 15,000Hz) for exceptionally clean and natural sound reproduction, a built-in "pop" and wind filter for protection against wind and explosive breath sounds, and extremely low mechanical handling noise. Supplied with the Model SM82 are an A82WS windscreen and an A57SL locking swivel adapter.

Professional user nit price of the Shure Model SM82 Line-Level Unidirectional Microphone is $165.

SHURE BROTHERS INC., 222 HARTREY AVE., EVANSTON, IL 60204.

Circle No. 154

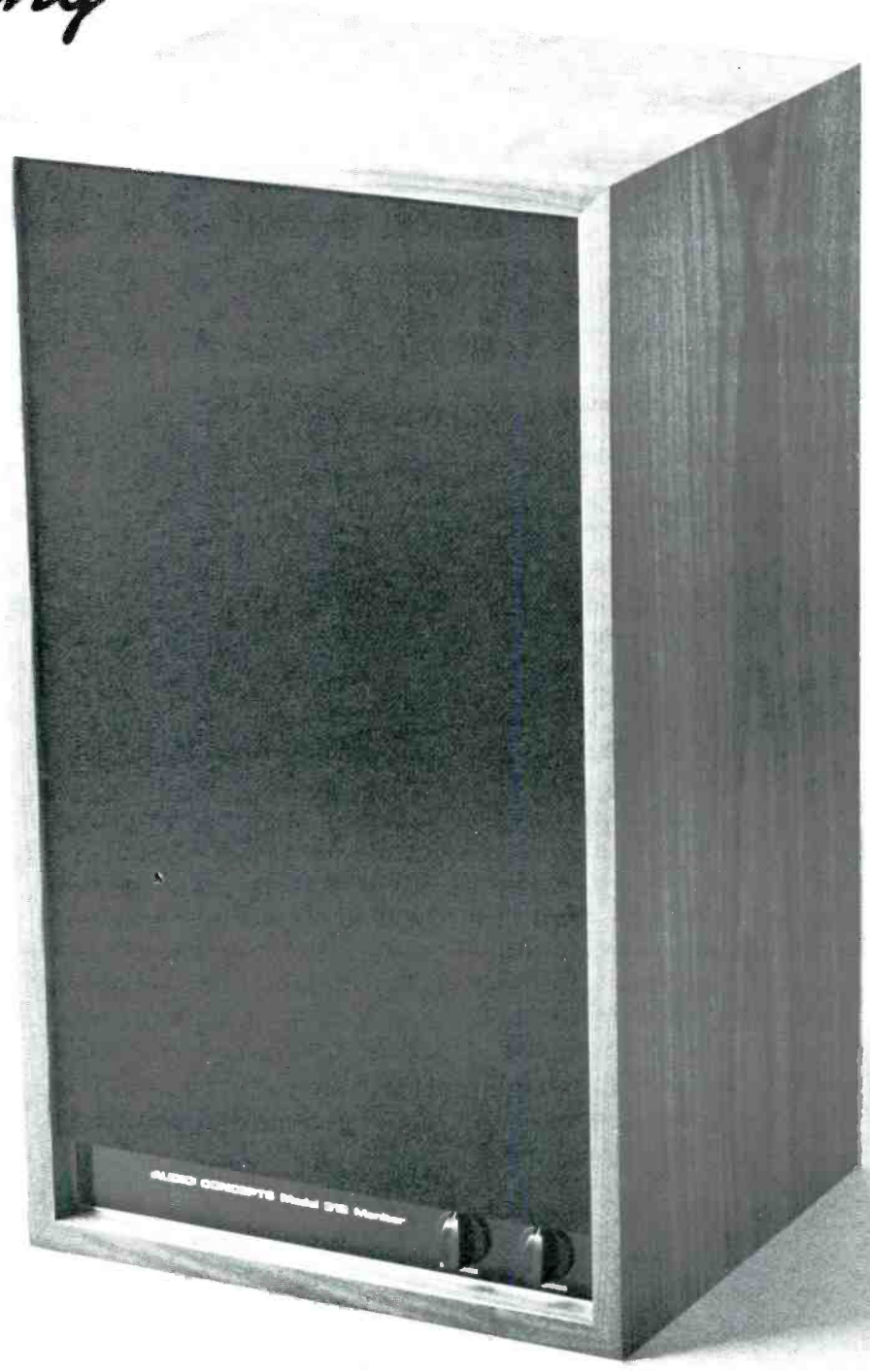
HIGH-FREQUENCY LOUDSPEAKER HORN GIVES HORIZONTAL SOUND DISPERSION FROM SINGLE SOURCE

A horn loudspeaker developed by a British company for entertainment and public address applications can provide wide horizontal sound dispersion from only a single source.

6912 melrose avenue • hollywood, california 90038 • (213) 937-1335

DAVE KELSEY SOUND

Featuring



AUDIO CONCEPTS

Model 312 Studio Monitor System

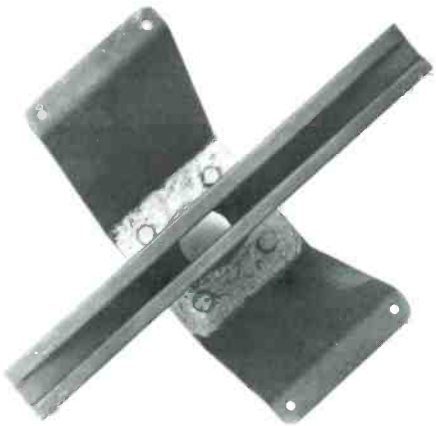
Stop by our booth at the AES show

Circle No. 155

R-e/p 71

The Vitavox 4kHz horn, for upper audio frequencies, achieves its effect through a very high but narrow mouth which circumfuses the sound.

Although designed for use with the company's own S3 pressure unit, which has a frequency response from 4kHz to 16kHz, the horn can, by means of adaptors, be used with drive units made by other manufacturers. The acoustic distribution is over 150° in the horizontal plane and only 30° in the vertical, with a maximum acoustic loss of only 2dB at 12kHz from the response on the axis.

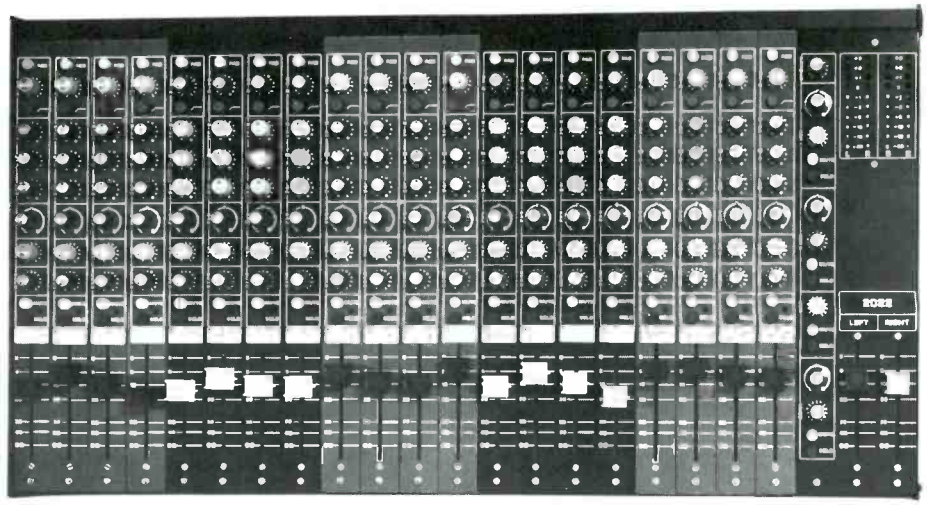


Made of heavy-gauge steel, treated with acoustic damping compound and finished in hammer black paint, the horn has a heavy-duty mounting bracket to support the weight of the associated drive unit.

The horn is 6½" long and should be clamped to a board provided with a slot 0.5" wide by 12" high to suit the horn mouth. The weight is 3 lh. 6½ oz.

VITAVOX LTD.,WESTMORELAND RD., LONDON NW9 9RJ, ENGLAND

Circle No. 156



2005 AD, PHILA., PA. INTRODUCES THE 2022 MIXER

Designed for service in the studio or on location, usages include; recording, broadcast, and sound reinforcement. This mixer combines features usually found only in large studio consoles with new circuit concepts to provide extremely low noise and distortion.

Features include low noise, high accuracy mic preamp with a 20dB pad switch and a 30dB range of control on the preamp gain, a low frequency roll off switch for pop and boom correction (-3dB at 200Hz). Three range equalizer — high and low frequency shelving and mid frequency peak/dip; pan pot allows placement of input signal anywhere in the stereo image. Cue mix for headphones, foldback or stage monitor; echo mix for reverb, echo or other effects; mute button silences this input in all mixes. Solo button assigns this input to the solo buss and automatically switches the headphones over to the solo buss. Also lights the solo indicator and assigns the "S" meter to the solo buss instead of to the echo buss.

L.E.D. overload indicator on every input, long throw professional conductive plastic slide faders; four digital L.E.D. VU meters: left, right, solo/echo, cue. Headphone/monitor output with level control; 2 aux inputs with pan pots, solo and mute; cue master with solo and mute; echo return with pan pot, solo and mute. Three 600 ohm balanced outputs; left, right and cue. Separate input and output for external echo; patch points on main outputs for insertion of graphic equalizers, etc. Two external, highly regulated power supplies in a 5¼" rack mount panel; bipolar tracking ± 15 volts for audio, -5 volts for digital meters. Modular construction: up to 20 input modules may be installed in the mainframe as it leaves the factory, or purchase what you need now and expand it yourself later.

Size: 29" (73.6 cm) wide, 15" (38.1 cm) high, 5.5" (13.9 cm) deep.

Weight: 40 pounds (18.16 Kg) maximum (when supplied with 20 input modules).

MOM'S WHOLESOME AUDIO, 2019 NAUDAIN ST., PHILADELPHIA, PA 19146.

Circle No. 158

Circle No. 157

SUNN DUAL 400 WATT POWER AMPLIFIER

This unit has been designed, according to the company, to withstand the rigors of the touring rock concert, and will reliably deliver 400 watts into 2 ohms from both amplifiers. Built with two discrete power supplies, the only thing common to both amplifiers is the chassis, and the power cord.

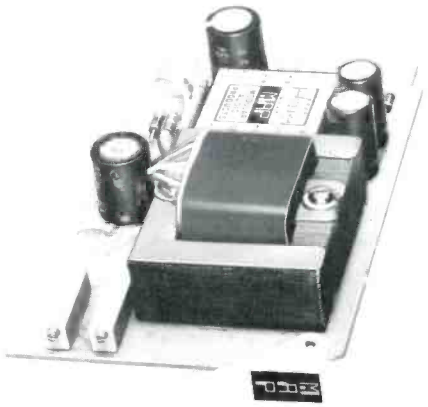
The device is fully protected by AC circuit breakers, DC fusing, and thermal overload protectors. It contains a built-in whisper fan for forced air cooling, and a massive 1100 square inches of heat sink, which enables the Sunn Magna 3800 to deliver full power continuously for an indefinite length of time without overheating. The unit has both high Z and low (600 ohm) Z inputs, and can deliver full output from an input voltage as low as 1 volt rms.

PRO-AUDIO DIV., SUNN MUSICAL EQUIPMENT COMPANY, AMBURN INDUSTRIAL PK., TUALATIN, OREGON 97062.

Circle No. 159

MODULAR INTRODUCES TWO NEW AMPLIFIERS FOR TAPE AND PHONO APPLICATIONS

The two new P.C. card amplifiers are the Models AT-27 and AP-27. Both are



special purpose; low noise equalized preamplifiers designed for conventional magnetic tape heads or phone cartridges.

Model AT-27 provides the standard NAB tape reproduce head equalization for tape machine speeds of 3¾, 7½, 15 and 30 ips by adjustment of its low and high frequency trimmer controls. An adjustable gain trimmer control sets the desired operating lvel.

The Model AP-27 provides the standard RIAA phono reproduce equalization. High frequency and gain adjustment trimmers compensate for losses due to stylus wear and to set the desired operating level.

The MAP Model 1731A audio operational amplifier is utilized as the active element in the AT-27 and AP-27 and provides the circuitry with an equivalent input noise of -125dBm, low distortion, short circuit protection and high reliability.

Delivery is stock to 30 days.

MODULAR AUDIO PRODUCTS, 1385 LAKELAND AVENUE, BOHEMIA, NY 11716.

Circle No. 160

NEW HALF RACK SIZE DIGITAL METRONOME

For accurate pre or post music scoring for motion pictures and television, the new UREI Model 964 Digital Metronome combines newest LSI circuitry with quality crystal clock precision to produce accurate tempos of from one frame per

beat to 40 frames per beat in 1/8 frame steps. It is operated by either front panel buttons or an external sync start signal. Self contained amplifier for headsets is

Circle No. 161

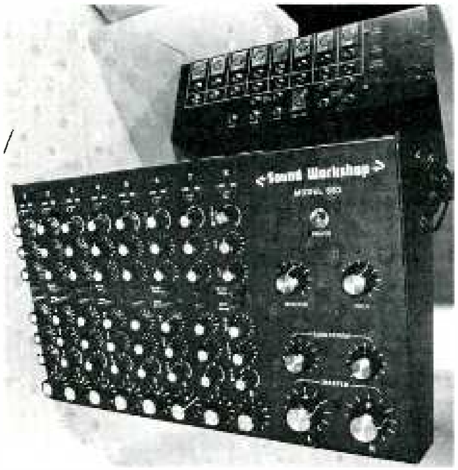
included. Can be used right on stage next to you while conducting.

UREI, 11922 VALERIO STREET, NO. HOLLYWOOD, CA 91605.

Circle No. 162

SOUND WORKSHOP MODEL 882 MIXING CONSOLE SUITABLE FOR RECORDING OR SOUND REINFORCEMENT

Sound Workshop Professional Audio Products announces the Model 882 Recording/P a mixing console, the first in a line of products that will include a discoteque phono mixer, a stereo reverberation system, and an electronic crossover.



The 882 features 8 inputs/2 outputs with each input including; calibrated rotary fader, pan pot, echo send, monitor send, mute, solo, high and low frequency EQ, pre-amp trim, and line/mic switching. The output section includes; dual echo returns, monitor level, solo level, and left and right program levels. A link jack on each input channel allows 8 independent channel direct outputs, or the insertion of external signal processing devices on the individual channels.

All circuit boards are plug in, and 2 or more mixers may be bussed together to provide 16 or 24 inputs. The model 882 operates on 120 VAC or ± 18 VDC, and features low noise and distortion.

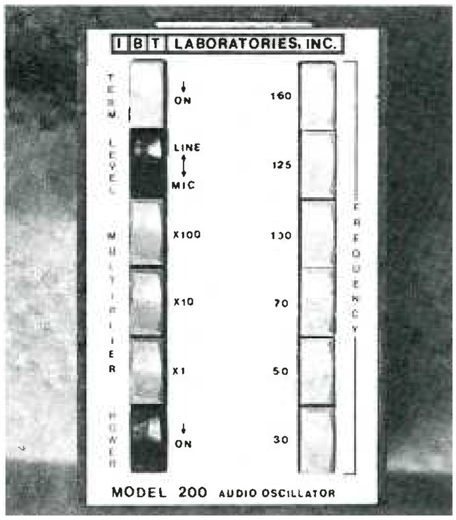
Price: $1200, or $1500 with optional meter/monitor panel.

SOUND WORKSHOP PROFESSIONAL AUDIO PRODUCTS, 1038 NORTHERN BLVD., ROSLYN, NY 11576.

Circle No. 163

NEW AUDIO TEST OSCILLATOR MODULE

A low cost, battery powered audio (sine wave) test oscillator is now available from IBT Laboratories, Inc. The Model 200 is housed in an extruded aluminum case and features 21 pushbutton selectable frequencies ranging from 30Hz to 20kHz. Output levels of 0dBm or -50dBm are available by pushbutton selection, with level tracking of ±0.5dB over the entire frequency range. The output is transformer isolated, balanced, and floating, and presents an impedance of 600 ohms at the high level setting, and 150 ohms at the low level setting. Distortion is below 0.5% at all frequencies. The unit will operate for several days (continuously) on one set of transistor batteries.



Measuring only 2-11/16 by 4-1/16 by 3-3/16, the model 200 will find application in sound system testing wherever a portable, lightweight unit is required.

Other applications include the testing and setup of tape recorders, consoles, audio-visual systems, communication systems, broadcast proof-of-performance measurements, servicing of high fidelity components, etc.

Price $129.95.

Delivery, stock to eight weeks.

IBT LABORATORIES, INC., 5200 MYER COURT, ROCKVILLE, MD 20853.

Circle No. 165

'AUDIO PRO' PROFESSIONAL AUDIO CLEANER INTRODUCED BY AUDIO-TECHNIQUES

AUDIO PRO is a specially formulated buffered, specific purpose cleaner and light degreaser. It is non-flammable, non-volatile, non-explosive, requires no rinsing, leaves no film. The product is said to be safe to use on virtually all non-porous surfaces. Spray and wipe.

The new cleaner has been extensively field tested by audio equipment manufacturers with excellent results. A number of major record companies are now regularly using Audio Pro for studio and equipment maintenance. The product is

Circle No. 164

said to be especially effective in penetrating iron oxide buildup on tape recorder heads, capstans and guides, as well as general cleanup of tape decks, electronic panels, mike stands, keyboards, and consoles.

PRICE: 4 oz. plastic spray bottle – $1.95, 8 oz. plastic spray bottle – $2.45, 16 oz. plastic spray bottle – $3.95, one gallon refill – $10.95. Audio Pro will be send prepaid to any address in the continental U.S. for a shipping and handling charge of $1.00 per bottle.

AUDIOTECHNIQUES, INC., 142 HAMILTON AVE., STAMFORD, CT 06902.

Circle No. 166

THE HEAD SOUND MODEL ESC-8 MIXER

Model ESC-8, the Eight Pack, is a portable eight channel stereo mixing console that features low distortion (less than 0.1%) and low noise (-123dBm equivalent input noise) as well as balanced low impedance inputs and variable input gain. The ESC-8 has extensive control features including seperate monitor mix, pan, high and low equalization, and overload indicators on each channel. It also has a 9 Watt headphone amp, an internal reverb unit, and two accurate peak reading level meters.

Input/output features include preamp outputs (patching or post channel out, optional), reverb mix output, left and right effects input, bridging/auxiliary out (useful when linking two or more Eight Packs together), and monaural output.

Model ESC-8 is very portable. With the cover in place, it is the size of a medium suitcase and weighs only 33 pounds.

HEAD SOUND, INC., 722 BROOKS ST., ANN ARBOR, MICHIGAN 48103.

Circle No. 167

NEW POWER AMPLIFIERS FROM BGW

BGW Systems announces two new stereo/mono power amplifiers. The Model 500D replaces the company's Model 500R stereo power amplifier, while the Model 750A is the new version of the previous Model 750.

Both models have been designed to comply fully with the new FTC Trade regulation rule on amplifier power output specifications. The following improvements have been made to the Model 500D:

1. Vastly improved thermal system with an extremely low thermal resistance from transistor case to ambient of a .2 degree C/watt. This thermal system uses a dual speed thermostatically controlled fan in a low turbulence environment for ultra quiet operation.
2. Ten rugged 150 watt output transistors per channel instead of the previous 6 per channel.
3. The removal of all current limiting circuitry.
4. Improved sonic qualities and operation into reactive loads.
5. Push-button bridging switch for quick change to mono operation.
6. Greatly improved mechanical packaging, capable of surviving severe abuse.
7. Front-panel mounted rocker style circuit breaker.

Both models carry 4 ohm FTC power ratings which make them quite unique in the industry. The Model 500D is priced at $799.00, the Model 750A at $949.00. Delivery stock to one week.

BGW SYSTEMS, P.O. BOX 3742, BEVERLY HILLS, CA 90212.

Circle No. 168

PORTABLE STEREO MIXING SYSTEM

Based on an 8 input mainframe and expandable at any time without recalibration, this mixer's performance is professional in every respect. A portable version

of our studio consoles, it features foldback and echo send channels; echo return; illuminated VU meters; and a separate monitoring system with CUE buttons, capable of driving headphones or stereo power amplifier.

Circle No. 169

Each input channel includes XLR type input; 5 position input attenuator; pre-EQ foldback level; 3 frequency EQ; echo send level; pan pot: sealed straight line fader; and monitor CUE button.

The 8 input Master Chassis is 24" wide, 13" deep, and 4" high and is priced at $1600; a 4 input extender chassis is $600; immediate delivery from stock.

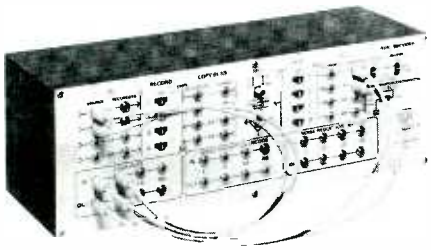
A detailed brochure about the Portable Stereo Mixing System is available free of charge upon request; dealer inquiries are invited.

RICHMOND SOUND DESIGN LTD.
P. O. BOX 65507, VANCOUVER, B.C., CANADA V5N 5K5

Circle No. 170

RUSSOUND ANNOUNCES QT-1 AUDIO SWITCHING AND PATCHING CENTER

The QT-1 is said to solve the problem of interfacing multiple audio accessories such as outboard Dolby or dbx noise reduction devices, graphic equalizers, matrix (SQ, QS, RM) and CD-4 encoders and decoders, limiters and other signal processing units. The QT-1 provides the same flexibility expected of a hard wired studio patchbay.



The QT-1 audio switching and patching center has 72 RCA type phono jacks on the back panel which permit the user to simultaneously connect up to four tape recorders, plus other required accessories and leave them connected to the rear panel of the QT-1. All interconnection functions may then be conveniently performed by front panel switching or changing patch cords just as is done with a permanent console patchbay.

Typical applications for the QT-1 include decoding from one noise reduction format and encoding into another, mixing down from a four channel source to a two channel dupe, intermixing of four and two channel sources, reversing or rearranging channels in a four channel playback, limiting or equalizing on one or more tracks, playing back from one recorder and copying on other recorders while monitoring the recorded signal on the system loudspeakers, etc.

The QT-1 is a passive device with no AC power or active circuitry to introduce noise or distortion into the signal. The only insertion loss is in the tape recorder outputs where resistors are included to prevent overload distortion when multiple recorders are in use at the same time.

The Russound QT-1 audio patching center is supplied in a walnut finish case with neutral beige front panel. The case may be removed for rack mounting. A set of twelve patch cords is supplied as standard equipment and additional cords are available from the factory at a price of $9.95 for a set of four. A 24-page booklet on patching techniques is supplied as an instruction manual with the unit, and is also available from the manufacturer separately at a price of $2.00

Price: $249.95.

RUSSOUND/FMP, INC., CANAL ST., NORTH BERWICK, MAINE 03906.

Circle No. 171

Classified

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SOUND 80, INC., 2709 E. 25 Street
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For Sale: Ampex 440-8-B, 8 track recorder, with remote control, excellent condition, spare electronics, 1½ yrs. old, 6 mos. guarantee. $7,300.00.
Also, Electrodyne 1204 Mixing Console, 12 in 4 out, with 8 track play back, one Cetec graphic equalizer input module, one spare input, recently overhauled by factory, excellent condition, 6 months guarantee. $7,000.00. Call: JOHN, 2000 Beck Bldg., Shreveport, La., 71101
318—424-2678

FOR SALE: One DBX 177 Noise Reduction System approx. 6 mos. old. $1,100. Write:
ABADON/SUN INC., P.O. Box 6520, San Antonio, Texas 78209

FOR SALE: Sony 854-4S 4 Channel recorder with "sel-sync" ($1450) and Sony 850-2 2 Channel recorder with 1/2 track record and 1/2 and 1/4 track play heads ($500). Both machines well maintained and in excellent condition.
Contact: WAYNE YENTIS
R-e/p Magazine
1850 Whitley No. 220
Hollywood, CA 90028
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10½" Reel Specialists: Boxed 10½" NAB ¼" reels $3.00@ flanges $1.25@. 10½" Precision reels $6.50@, flanges $2.25@ Heavy duty or Tapered. Wanted: used ¼" NAB & Precision hubs. SOUNDD INVESTMENT CO., POB 88338, ATLANTA, GA. 30338.

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Continental Film Productions Corp.
P.O. Box 6543, Chattanooga, TN 37408

FOR SALE: Tascam 4-track tape recorder with remote control. Excellent condition, 6 months guarantee, $2,200.
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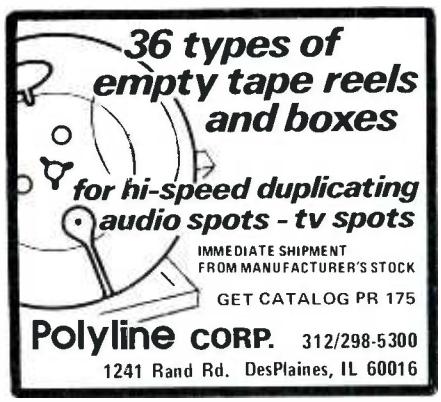

Circle No. 173

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She uses true RMS level detection, which you know is more reliable and accurate than other methods. Her dynamic range is enormous and her noise contribution practically negligible. Her output is automatically ground loop compensated and she is protected against turn-on and turn-off transients. She is beautifully packaged and small enough that you can take her with you wherever you go. Or you can bolt her into the rack where she'll give you a lifetime of faithful service.

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296 Newton Street
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R-e/p 77

Circle No. 174

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Ampex AG-440 2 track w/console — $2300.
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STEREO DISC MASTERING SYSTEM
Neuman transistor VG66 amplifier rack; SX68 cutterhead; AM32b variable pitch and depth lathe, with Leitz microscope, playback arm, suction system and stylus microscope; 2 racks with 2 Pultec EQPI-A. Fairchild 670 limiter, 2" oscilloscope, complete TRS Jackfield, control panel with program and preview metering and controls; 2 Fairchild 688 amplifiers and 2 KLH 4 speakers. Console with Ampex 300 deck, advance heads and 4 solid state playback preamps. Best price over $25,000.00.
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