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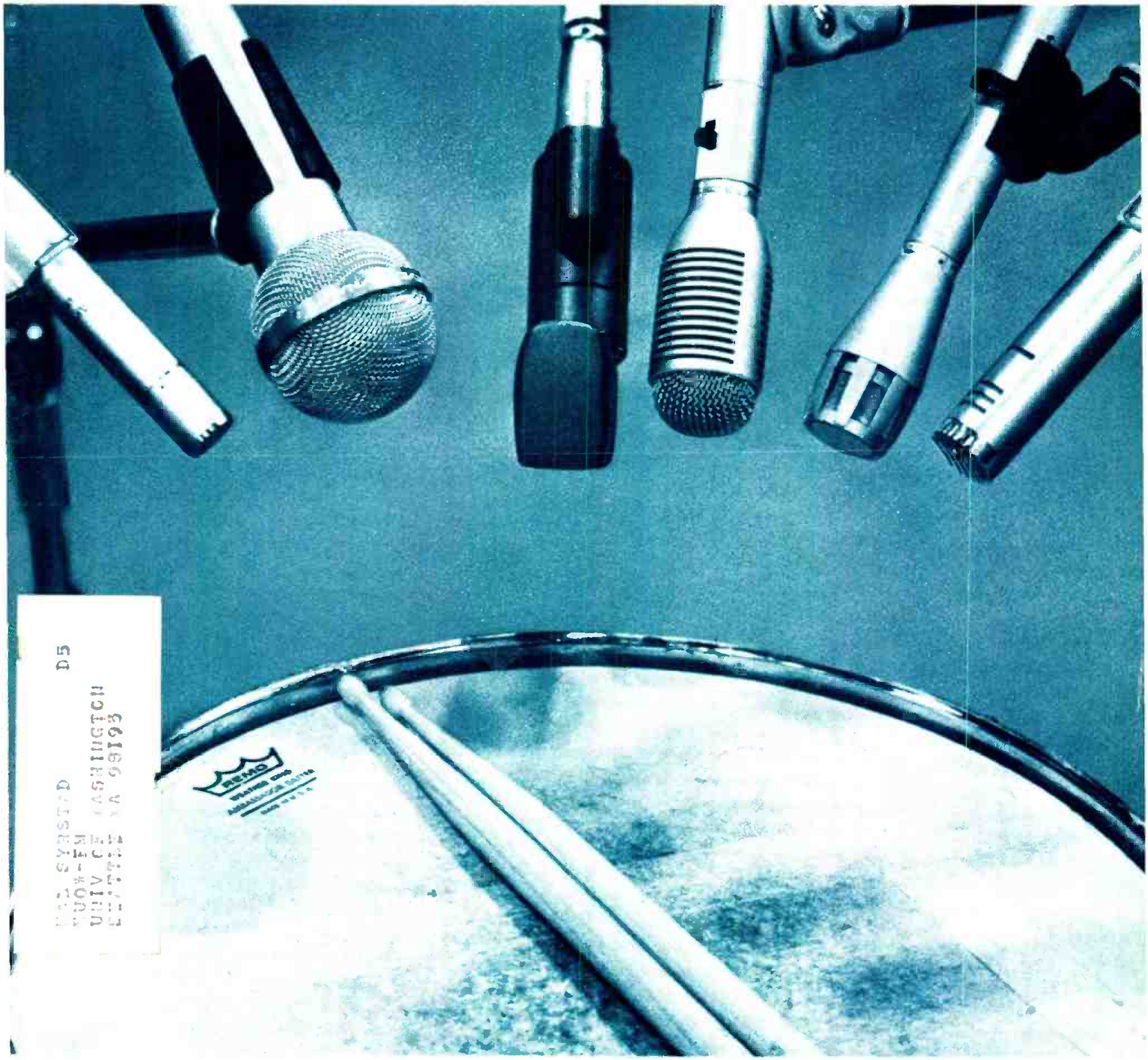
THE SOUND ENGINEERING MAGAZINE

NOVEMBER 1974 $1.00



IN THIS ISSUE

660 clamp-mounted...$69.80: 661 stud-mounted... $71.30; 671... $60.00 (suggested resale net— slightly higher in western states).

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● Expert advice in planning a studio continues in Michael Rettinger's RECORDING STUDIO ACOUSTICS, Part 3, in which he discusses how to plan room dimensions to effectively handle the maximum number of instruments likely to be used in the studio.

There's more of W. E. Anderton, continuing his article on PROFESSIONAL SOUND RECORDING. In part 2, Mr. Anderton goes into the pros and cons of such matters as the four-band compressor-expander, noise-reduction for discs, decoders, the application of noise-reduction to optical sound tracks, mobile recording units, pulse code modulation, quad, and digital systems.

John Woram ranges over the entire audio scene in a special devoted to a complete roundup of significant new product developments and techniques, including lots of pictures of what he's talking about, mostly snapped at the New York AES show last fall.

ABOUT THE COVER

● A cluster of microphones poised over a drum indicates the diversity of possibilities for bringing out the best drum sound, or any other sound, for that matter.

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THE SOUND ENGINEERING MAGAZINE

NOVEMBER 1974, VOLUME 5, NUMBER 11

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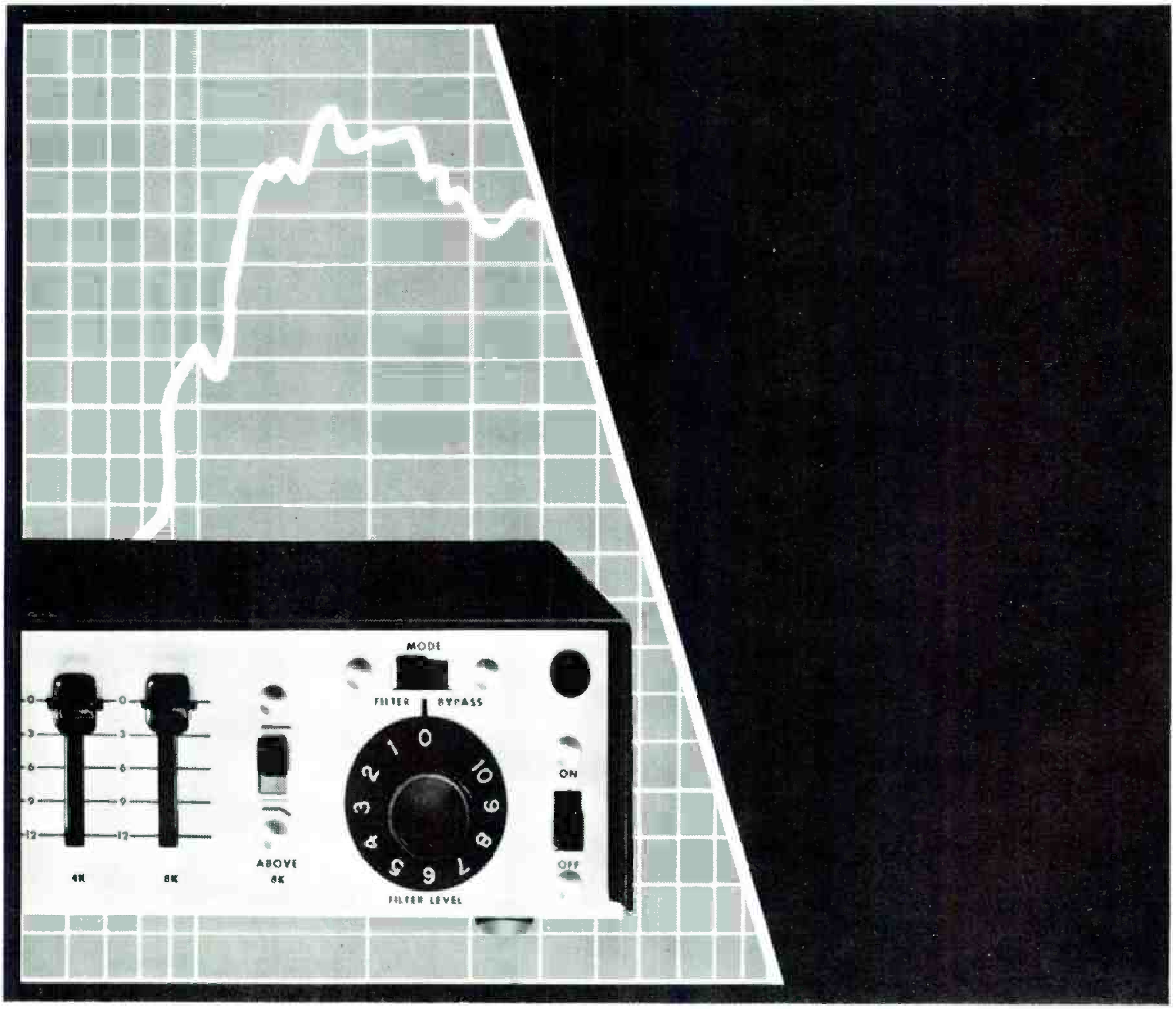

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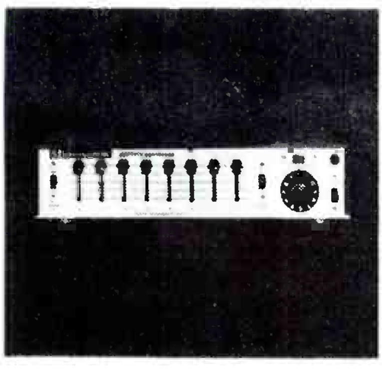

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db letters

THE EDITOR:

Mr. Rienstra has presented some interesting comments on the historical background of the Haas effect. It should be noted, however, that many researchers have contributed significantly to various aspects of this phenomenon. For example, the blending or fusion of a spatially separated sound source into an apparent single image had generally been known for many years prior to the Haas study.

At the 14th meeting of the Acoustical Society of America in 1935, Fay[1] and Hall[2] reported and demonstrated the directional effects on the auditory illusion of varying the time difference and relative intensity levels of speech sounds from two loudspeakers widely displaced in a horizontal plane. The reports emphasized that a suitable time delay in the amplified sound produced the illusion that the reinforced sound (as well as the direct sound) appeared to originate at the mouth of the talker.

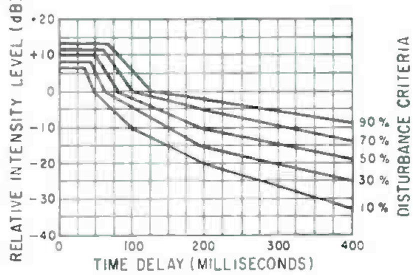

Fig. 1. Constant percentage disturbance contours for the transient acoustic response of rooms. (after Bolt & Doak)

In subsequent years, other investigators applied their own descriptive terms to the effect: e.g., the *Rotation of Sound Image* described by de Boer[3] (1940), the *First Arrival Effect* by Langmuir[4] et al. (1944), the *Law of the First Wavefront* by Cremer[5] (1948), etc. However, the appearance of the Wallach et al. study at Harvard (1949) and the Haas dissertation at Göttingen (a few months later) pro-

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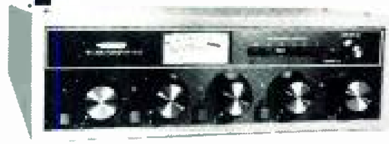


vided the most commonly used designations for this phenomenon, possible because both of these independent works included a detailed analysis of their respective experimental data. At any rate, the terms *Precedence effect* and *Haas effect*, derived from these studies, are now being used interchangeably. Reference is sometimes made in the literature to the *Haas-Meyer effect*, in recognition of the eminent Professor Erwin Meyer, who initiated and directed the Haas study.

Mention should also be made of the data produced by Bolt and Doak.[6] who have extended the work of Haas on the intelligibility of speech signals. Using Haas' data on the percentage of listeners disturbed by time delay effects, the original curves were modified and extrapolated to allow for the difference between the average German speech rate of 5.3 syllables per second, and the speech rate of 4.5 syllables per second considered to be average for English. The replotted curves (FIGURE 1) indicate the constant percentage disturbance as a function of echo intensity level and echo delay time relative to the direct sound. Applying these results to speech reinforcement systems, the 10 percent disturbance contour is considered as the reference in determining whether time delay interference calls for the insertion of a time delay unit. Thus, for delayed signal levels and time differences falling above the reference curve, time delay correction is required, whereas for the range of levels and delays below the curve, no such mechanism is needed.

SIDNEY L. SILVER

REFERENCES

1. R. D. Fay, "A Method for Obtaining Natural Directional Effects in a Public Address System," *Journal of the Acoustical Society of America*, vol. 7, Jan. 1936, p. 239(A). (Abstract of 1935 Report).

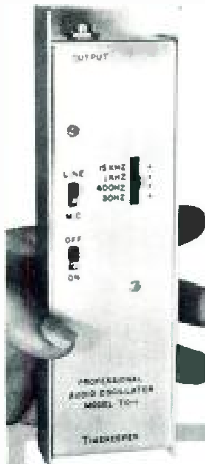
2. W. M. Hall, "A Method for Obtaining in a Public Address System the Illusion that the Sound Comes From the Speaker's Mouth," *Journal of the Acoustical Society of America*," col. 7, Jan. 1936, p. 239(A). (Abstract of 1935 Report).

3. K. deBoer, "Stereophonic Sound Reproduction," *Philips Technical Review*, vol. 5, April 1940, pp. 107-114.

4. I. Langmuir, V. J. Schaefer, C. V. Ferguson, E. F. Hennelly, "A Study of Binaural Perception of the Direction of a Sound Source," OSRD Rep. 4079, PB No. 31014, 1944. (Available from the Library of Congress).

5. L. Cremer, "Geometrische Raumakustik," (Geometrical Room Acoustics), Die Wissenschaftlichen Grundlagen der Raumakustik, vol. 1, 1949, p. 126.

6. R. H. Bolt & P. E. Doak, "A Tentative Criterion for the Short-Term Transient Response of Auditoriums," *Journal of the Acoustical Society of America*, vol. 22, July 1950, pp. 507-509.



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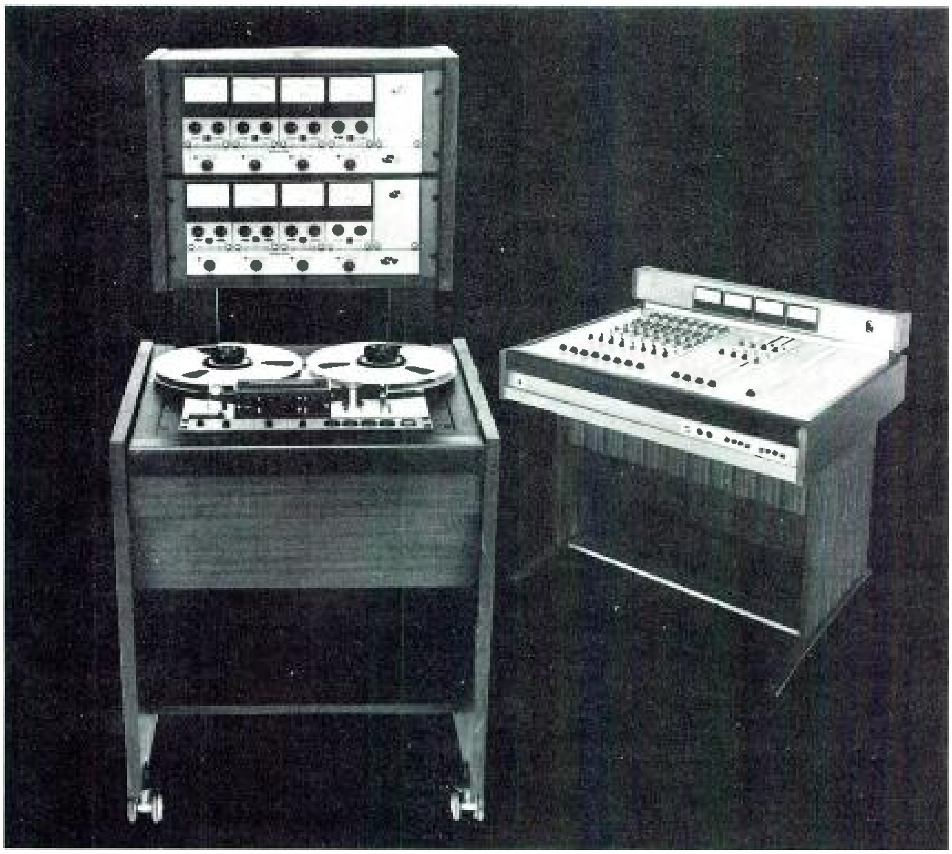

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db the sync track

JOHN M. WORAM

• Due to some last minute schedule changes, the convention report is being put off till next month, and next month's column is now going to be this month's column. So who cares?

Me. You see, I haven't written next month's column, since I was going to finish this month's column first. But now, this month *is* next month, or something like that. Anyway, I've got to do next month's column fast, or get left out of this month's issue.

How about something nice and controversial for a change? Like microphone placement, naming names and everything! Specifically,

MIKING THE DRUMS

Every once in a while, someone will turn up in class with a record containing the latest example of the greatest drum sound ever, and there'll be all sorts of guesses about what mics were used, and how they were placed. Nine times out of ten, I have no idea how the drums were miked, and I'll say something to suggest that the sound is not being created by the microphone—it's what's up front that counts, that is, the drummer.

This sort of comment rarely pleases anyone, so I'll drag out my personal collection of records (both of them) and compare drum sounds. Some actually aren't bad, if I do say so myself. Others are not so hot.

So, how come? Same microphones, same drums, same engineer—yet the sound varies from record to record. And that's because each record represents a different drummer. Certainly, my "standard" drum setup gets rearranged a bit to fit the needs of the particular session, but by far the greatest influence on the drum sound is you-know-who.

When working with a pro studio drummer, there's not much to do. The secret is to set up the mics, do a take, and play it back. The drummer will listen, and depending on what he hears, alter his playing somewhat. Or perhaps you will bring up the snare drum mic, or back off on the floor toms, or whatever. But you leave the playing to him, and keep the knob-jockeying to a minimum.

Presumably, he knows more about drumming than you do. That's why he was hired to play the drums, and you were hired to stay in the control room. If that prospect bothers you, perhaps you've missed your calling.

Anyway, by now you're getting the ultimate drum sound; that perfect combination of a pro musician and good engineering. Your adoring fans tell you what a great engineer you are, and you modestly agree.

But, if you're so damn good, how do you explain those infrequent recordings where your drum sound is, to put it politely, gross? It's easy—you blame it on the drummer! But as you do so, remember that all those great drum sounds are *also* the "fault" of the drummer. This does not minimize your importance as an engineer—it just puts your job in perspective.

About those occasional lousy drum sounds: I remember a particularly awful session all too well. Our drummer was damn mad about the drum sound I was getting. He demanded that I do something. Why wasn't I getting that same sound he heard on all of Elton John's records? Why was I so stupid that I couldn't figure out what those British engineers were doing?

I dutifully went out into the studio to absorb the full essence of his artistry. I think he could have done better by beating a pillow with a couple of damp rags. I suggested he consider a wrist transplant, but he didn't seem to care much for that. So it was going to be up to me to transform him into Buddy Rich. Needless to say, I failed. (If I had succeeded, I wouldn't be writing this crummy column, would I?).

Well, it wasn't a total disaster. Allison came to the rescue with a couple of her Kepexes and some of the slush was eliminated. So, it was a little better, but still a long way from great. I don't suppose I've been forgiven to this day.

Speaking of Kepex, if the bass drum lacks the apparent tone you're looking for, you can do wonders with a Kepex. Set an oscillator to the desired pitch, and feed it through the Kepex. Gate the Kepex with the bass drum mike, so that the oscillator takes on the attack and decay characteristics of the bass drum. Then mix the Kepexed oscillator in with the sound from the bass drum mike itself. It can be very effective. It can also be overdone, so listen carefully.

PLACING THE MICS

Getting back to the microphones, the question comes up—how many are necessary?—and, where should they be placed? How many depends in great measure on the type of recording you're doing. I sat in on a jazz session recently in which one mic was used and that was placed somewhere over the drummer's right shoulder, facing the drum set. The microphone heard pretty much what the drummer himself heard. And the effect was rather good. The drums sounded like drums, and not like a set of tuned mouse traps exploding on the ear lobes.

Of course, if you want that "up tight" sound, one mic, or two, will not do much good. A good rock record needs a larger-than-life drum sound, (usually) and multi-miking becomes a necessity.

So, you'll often find a separate mic for snare drum, floor tom tom, and bass drum, with others on the various cymbals. And then there's the question of how many tracks to use. Of course, a skillful engineer can mix a collection of drum mikes together onto one track and come up with a well balanced sound. However, the drums are, after all, a fairly large sound source, and a one track mix—no matter how well balanced internally—will probably sound comparatively small. One might want a larger sonic spread.

TRACKING THE DRUMS

If so, one track is not enough. Sometimes, the engineer will put the bass drum on its own track, with all the other mics on another track. This allows for good control during the mixdown of the bass drum but doesn't do much for creating a stereo spread. After all, the bass drum is in the center of the set, with the rest of the set spread out on both sides. During the mix, if the bass drum is panned to one side with everything else on the other side, it can sound a little strange.

So, if two tracks are available, it might be better to divide the collection of drum mikes more or less evenly between the two. Later on, one of the tracks will be panned left of center, the other right of center. Or all the way to the sides for maximum spread. To keep the bass drum in the center, it is simply fed to both tracks while recording.

Some might put the bass drum on a third track, but this may not really be necessary. If the drums have been well-recorded, with the bass drum in the center of the two track pick-up, a little equalization to both tracks will raise or lower it as required. (This is where the good engineering comes in.)

As for the other microphones, since the snare and tom tom are on opposite sides of the bass drum, their respective mics will probably go to the separate tracks, thus keeping them apart in the mix. And if there are overhead mics, the same will apply. With this type of setup, sometimes the snare and tom tom mics might be panned in a bit so that in the stereo mix they lie closer to the center, with the overhead mics remaining at the extreme left and right.

A basic setup that has worked well for me uses five microphones: two overhead (left and right), one on the floor tom tom (left), one on the snare drum (right), and one for the bass drum (center). Of course, if this approach is used, then even more mics may be needed for more involved drum sets. Which brings up an interesting point.

All too often, those super-sets, with two bass drums, and half a dozen tom toms, just don't make it in the studio, no matter what they do for the drummer on-stage. And that brings up yet another point.

One of the most difficult tasks facing the engineer may be to convince the group that the recording studio is *not* the concert stage. What works on stage may not work in the studio, and vice-versa. Approach these super-sets with caution: you may be better off—for recording—with a smaller setup.

NAMING NAMES

As for the microphones for that basic setup, I've been using a pair of good omnis overhead—usually Electro-Voice RE-55's, spread a few feet apart. They're quite flat, and give a good overall sound to the drums. There's no leakage problem here: the drums are making plenty of noise, and the mics are mixed in at rather low level—just enough to catch the upper cymbals, and create some "space."

For snare drum, I've been using the AKG 452. The 452 is the same as the better known 451, except that it's wired for 48 volts, making it electrically compatible with the Neumann phantom power supply. And, with all the accessory gadgets that are readily available for this microphone (right angle swivels, extension tubes between capsule and preamp, etc.) it's great for reaching into tight spots without getting in the musician's way. I've used the omni capsule, due to the close working distances, and for all-around uniform response.

To mic the floor tom tom, I've used the Electro-Voice RE-15. It's got a good low end, and the off-axis response is excellent, so the rest of the drums do not sound muddy. Good off-axis response is an important consideration for any drum mic. No matter where the mic is pointed, it will certainly pick up other parts of the drum set too. A poor off-axis response will degrade the total drum sound—I guess that's why I use so many omnis.

For the bass drum, I'm now using a Shure SM-57. Here, the proximity effect can be used to advantage, and the fall-off in high end response off axis may help to minimize some of the leakage from cymbals and snare.

Now that I've mentioned the unmentionables, I should say that anyone who runs out and buys these microphones does so at his peril. These mics work well for me, under the conditions I usually find prevailing. Of course, you would have to know a lot about the studios in which I work, the musicians, the arrangers, and the type of music being recorded. In short, it would take a book to fully explain all the conditions that dictate the use of the microphones I've just mentioned. And every other engineer in town could write a book explaining why that choice was just about the dumbest collection of mis-used mics in the history of the recording industry. So, do your own experimenting, whenever and wherever possible. ■

NORMAN H. CROWHURST

• As every reader of this magazine knows, audio reproduction involves every link in a chain. And like any chain, it is no stronger than its weakest link. That is an overworn cliche—and I dislike cliches—but it happens to apply here. For several decades now, I have watched audio people spend fortunes to make amplifiers, and every other electronic link in the chain, as efficient as possible.

For a long while, not too much seemed to be done about microphones and loudspeakers, but in recent years many innovations have been made, so that there is no longer any reason why either of them should be the weak link. However, they sometimes are, for reasons we discussed at some length during the workshop at Brigham Young University. But there is an even stronger claimant for the position of weakest link: the acoustics of the room itself, or the rooms themselves, where more than one is involved.

It seems that most people are looking for simplistic (where did that word come from—I did not know it till I worked in educational circles—what's wrong with just plain "simple") answers. Frequency response is usually regarded as the most important parameter of everything from the microphone through all the electronic equipment to the loudspeaker, and most people at least realize, by now, that what happens before the direct microphone input or after the loudspeaker output can make a difference.

This is usually addressed, to parody one of our simplistic, blank-faced comedians such as Jackie Vernon, by asking, "Has anyone thought about the frequency response of the room?" The implication seems to be that a room has a simple frequency response, just like any electronic part of the system. That kind of comment reminds me of the story of the accountants who had nicely balanced the books when the boss came in and asked if they had added in the date! The frequency response of a room is about as relevant to the rest of the system as the date is to dollars and cents.

Of course, you don't add the date into a column of financial figures, but you could not say that the date is not relevant to finance, either, could you? If you disagree, please tell me where I can still get a 10¢ hamburger. Or for that matter, a silver dime for 10¢? The frequency response of the room, while not part of the reproduction system, is significant.

How do you take the frequency response of a room? To simplify things a little, let us suppose that we have loudspeaker, and flawless electronics to a perfect microphone and perfect loudspeaker, and flawless electronics go with them. The microphone and speaker were checked out in a perfect anechoic chamber and were presented

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with response curves that looked as if they were drawn with a ruler, let's say.

Now, how do you take the frequency response of a room? From force of habit, you may start connecting an oscillator through amplification to the loudspeaker, and then connect the microphone to amplification that feeds a response plotter. Okay? Or did I hear someone mumble something about "standing waves?"

So how can you offset the effect of standing waves, which every room but an anechoic chamber will generate, any time a single frequency is produced, with or without other sounds present, in the room? One answer that was very popular at one time, was to warble (or wobble, whichever you prefer) the oscillator frequency so it was always changing and the standing waves never had time to build up.

True, that will stop the pen zipping wildly back and forth across the plotter, showing that standing waves are no longer building up. But it will also stop the curve from showing the real dips and humps that the acoustics of the room introduce to the scene. We want to get rid of one without the other, please.

WHITE NOISE VS. PINK NOISE

Now we hear whisperings about "white noise," and "pink noise," and "real time analyzers," with references to a name we seem to have heard before, Don Davis. One fellow, after finding that he is not alone in not knowing the difference between white and pink noise, asks. Surprisingly almost everyone seems hazy about that and, even after considerable discussion, we find people using the terms differently. So I can only tell you how I understand those terms: someone else may define them differently!

Noise is due to random events, usually the random movement of charges at a microscopic level, that are amplified to form a sound source. In white noise, the random content averages the same energy content in the same number of cycles bandwidth, at any frequency in the spectrum. Thus the hundred cycles from 10,000 Hz to 10,100 Hz will contain 10 times as much energy as the hundred cycles from 1,000 Hz to 1,100 Hz, if it is white noise.

To disappoint some of our conservative friends, pink noise is not white noise, after bouncing it off the Kremlin! If it were, the outside paintwork of the Kremlin would be priceless. Pink noise is formed by passing white noise through an electronic weighting network that de-emphasizes the higher

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The current STUDER A80.

WOW and FLUTTER—0.06% peak wtd, measured per DIN 45507, ANSI S4.3-1972 (approx. 0.04% RMS) @ 15"/sec.

DISTORTION—< 1% total harmonic @ 200 nWb/m tape fluxivity, NAB-15"/sec (note: this is mainly 3rd harmonic and is a function of the tape itself; 2nd harmonic distortion typically 0.1% at 10 dB ABOVE operating level)

STARTING TIME—<0.5 sec to 0.1% FLUTTER for all tape widths (note: start time considerably less if start is made from EDIT mode.)

ERASURE—approx. 85 dB @ 1 kHz

HEAD LIFE—extremely long due to use of "VACODUR," an alloy so hard it must be cast and ground but is magnetically soft and without the disadvantages normally associated with ferrite heads.

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frequency content. To be truly pink, the energy distribution should be such that, for example, the band from 100 to 110 Hz, that from 1,000 to 1,100 Hz, and that from 10,000 and 11,000 Hz, each contain the same amount of noise energy.

White noise contains so much energy per cycle. Pink noise contains so much energy per octave. White noise, passed through a network that has a downward slope of 3 dB/octave through the whole audio spectrum comes out as pink noise.

The reason that noise is used to measure acoustic properties is that it contains all audible frequencies, randomly assorted. Over time, the frequency content is such that all frequencies within the audio range issue from the loudspeaker at a uniformly average level, but none of them is steadily maintained so as to excite standing waves.

To measure frequency response then, we just put in noise, white or pink, according to what you want to measure, and then analyze the output, to see what is still the noise, and in what relative intensity. Thus, if we use pink noise because a flat frequency response with uniform energy per octave is what we want, we analyze what comes out in appropriate bandwidths, which could be octave, half-octave, third-octave, tenth-octave, or whatever.

NO BANDWIDTH, NO NOISE!

There, for a start, is a difference: noise does not have a specific frequency. If you allow only one frequency, zero cycles bandwidth, you can have no noise energy at all because the noise energy, of either type, is measured per cycle or per octave, something that designates a bandwidth. Thus it should be fairly obvious, no bandwidth, no noise!

The only thing that can get through a theoretical zero width filter is just that frequency, at steady level, like a sine wave. If the level fluctuates at all, as it inherently does with noise, then more than one frequency is present, because fluctuating amplitude is amplitude modulation, which is made up of the main frequency, called a *carrier* in radio, and side-bands, determined by the rate of amplitude fluctuation.

From there, we find different ways that the measurement can be made, in regard to just the electronic part. You can feed complete pink noise into the loudspeaker, pick up whatever reaches the microphone, and analyze it with, say, third octave filters. There are 3⅓ octaves to a decade, which means there will be 10 third octave bands in a decade: 10 from 20 to 200 hertz, 10 from 200 to 2,000 hertz, and 10 from 2,000 to 20,000 hertz.

So that divides the audio spectrum, from 20 hertz to 20 kilohertz, into 30 tenth octave bands. You feed the amplified output from the microphone to a set of these filters and then scan the filters to see how the energy is distributed among the 30 bands as it comes back to the microphone.

Such an analyzer is called a *real time analyzer*, because time is involved in making such a measurement. When your signal is a sine wave of single frequency, the wave has an instantaneous and continuous amplitude. If the sine wave has, say, 1 volt peak, then every cycle will have an amplitude of 1 volt, quite regularly. All you need, to measure it, is to have enough of a cycle to be able to identify a peak.

But when you use noise, you are faced with an energy/bandwidth situation. In zero time, or in zero bandwidth, as we have already pointed out, there can be zero noise. To measure any noise present, we must have a finite bandwidth, such as a third of an octave, and we must take some time to pick up energy within that bandwidth.

So a real time analyzer takes the signal and separates it by bandwidths, storing elements of noise that fall in the slot provided by each filter. Then a sampling circuit scans each filter output to see how much it has picked up in the time between samplings.

TWO WAYS TO SCAN FILTER OUTPUTS

There are two ways this can be done. One way is to feed the whole noise spectrum into the room, all of the time amplifying and analyzing the whole noise spectrum reaching the microphone. Or else, the noise may be separated into bands at the outset and each band given so much time, in succession, into the loudspeaker, while what comes back is measured.

With the first way, all the frequencies are in the room at the same time: a complete pink noise. These frequencies are separated, by their bands after they have negotiated the room's acoustics and come back to the microphone. With the other way, only noise within one band comes from the loudspeaker at any given instant, for whatever duration is needed to sample that band.

There is some discussion about which is the better way. The proponents of the second way argue that if the original components emitted by the speaker produce byproducts different from those radiated, such as by intermodulation, then those byproducts will be classified according to the frequencies at which they come back, not according to the frequencies that stimulated them.

On the other hand, if the speaker is emitting only noise in one narrow band at any instant, the sound in one band will still be traveling round the room, from loudspeaker to microphone, when the next band starts out from the loudspeaker. Thus there is inevitably a time spill-over from each band to the next. Unless steps are taken to reject "late returns," this can invalidate readings probably to as great an extent as the first method.

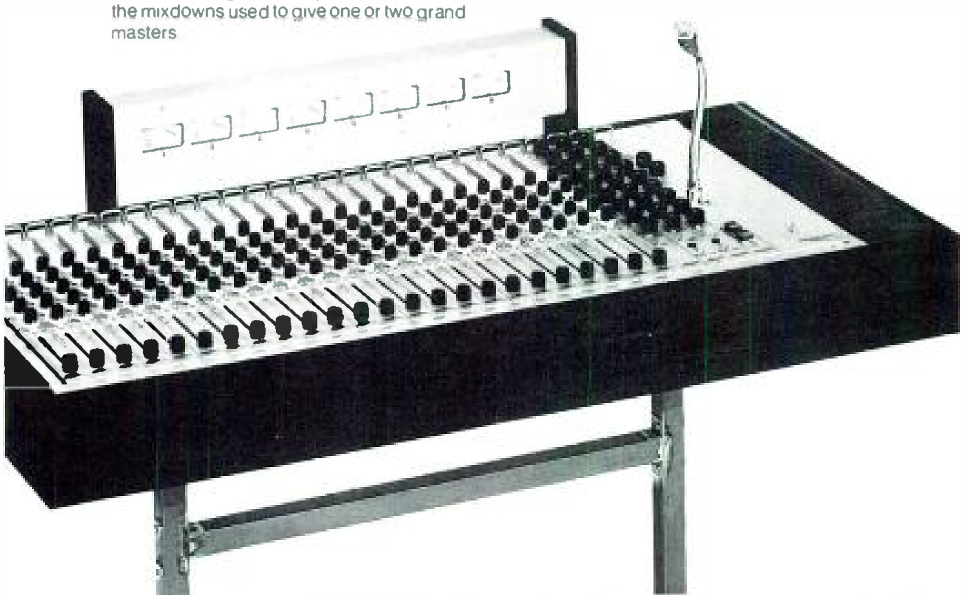
As time is involved in measuring, or collecting the contents of each band, we favor the use of full band from the loudspeakers, receiving full band at the microphone, and only then submitting it to the real time analyzer. This means that all bands are being monitored, virtually all the time, although they are sampled in sequence, as a rule.

This discussion turned out to be longer than I expected when I started, although the time spent discussing it in the workshop should have warned me. So we will have to continue this another month. ■

PROFESSIONAL MIXERS FROM INTERFACE ELECTRONICS

Model 8X4-100 Series 100 portable eight-input four track mixer shown at left includes trackswitching, echo send, equalizing, panpot, conductive plastic sliders, VU meters, and options including interchangeable modules (see right), internal reverb. Used for remote or small-studio recording on up to four tracks, mixdown of up to eight tracks, small sound systems (with up to four submasters), stage monitoring, production work. The Series 100 also comes in 16 and 24 input mainframes.

Shown below, the **Model 24X8 Series 300 mixer in 24C8 console,** a 24 input eight track mixer with pushbutton trackswitching, multifrequency equalizing, echo send, panpot, cue/solo, 6" conductive plastic sliders, monitor mixdowns, masters, VU's, talk/slate, module outputs, fully wired and ready to operate. Also available in 16 and 30 input mainframes. Used for studio recording up to eight tracks (more using module outputs,) mixdown of up to 24 tracks; also suitable for large sound systems, wherein the track masters may be used for submasters and the mixdowns used to give one or two grand masters.



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Interface Electronics products may be ordered direct from Interface Electronics or from any of our dealers.

Dealerships are still open in some areas.

SERIES 100 four track mainframes are fully wired with input and output connectors, power supply, VU meters, and masters, for Series 100 plugin modules listed below, in the Model 8X4 shown at left, also in the 16X4 and 24X4, which are folded aluminum pans for console or trunk mounting. Modules include:
Model 100D—Basic module with track-switching, panpot, echo send, high and low equalizers, high and low rolloffs, solo switch, slider attenuator, gain set switch with input pad, line/mike switch.
Module 100B—Similar to the model 100D but with three equalizers with a choice of three frequencies on each.
Model 100C—Input module with 40 db compressor with high compression ratio to ride gain on varying signal to hold constant record level, includes equalizers, track-switching, gain set pot, slider, echo send.
Models 100AQ and 100CQ—Single input modules with four-way pan between the four tracks; CQ also has compressor as above.
Model 100R—Combination sound-system and stage monitor module feeds stereo sound system through panpots plus independent monitor feed to four monitor busses plus echo send, equalizer.
Model 100J—Stage monitor module provides eight monitor sends from each input plus three equalizers with a choice of frequency on each, rolloffs, gain set switch with input pad position, line/mike switch, mute.
Model 100Q—high-level four-input module with level, cue switch, four-way pan, for feeding the four track output.

SERIES 200 two track stereo mixers come in standard 8 × 2 portable two track panpot mixer with Bauxendall equalizers, echo send, conductive plastic sliders, setup oscillator, master and VU meters; can be slaved to give 16 or more inputs, also nicad battery option, 16 or 24 input versions on special order.

SERIES 300 offers eight track 16 and 24 input fully wired mainframes with power and XLR type input and output connectors, plug-in input modules with nonexclusive pushbutton track selection, panpot, echo send, cue (which doubles as monitor-only solo), three octave-wide peaking boost or cut equalizers with a choice of three frequencies on each, adjustable input gain and input pad, line/mike switch, and a six inch conductive plastic slider. Each module is provided with balanced 200 ohm mike input and bridging single ended line input, as well as module output. Using module outputs, more than eight tracks can be fed. The fully modular system also includes masters and setup oscillator on the output module, and up to three mixdown-monitor modules with automatic transfer of cue to monitor if desired, and mixer-playback switch; the talk-slate module includes slate track select and talkback/slate microphone.

ALL INTERFACE ELECTRONICS mixers are capable of performance comparable to the finest professional equipment, and insure reliability through the use of plug-in integrated circuits, plug-in modules, conductive plastic sliders, gold plated card and IC connectors, tantalum or computer-grade condensers.

COMMON SPECIFICATIONS FREQUENCY RESPONSE:
± 1 db 20—20,000 Hz
EQUALIZING:
± 12 db at specified frequencies
DISTORTION:
less than 0.1% THD @ 400 Hz, + 3 VU
NOISE:
less than 0.6 microvolts equiv. input
INPUTS:
MIKE: 200 ohms balanced, XLR type connector max. level 0.5 volts RMS max. level 5 volts with int. pad
LINE: 10K unbalanced phone plug
OUTPUTS:
TRACK: approx. 1 volt RMS at zero VU unbalanced, to not less than 600 ohms, XLR connector
ECHO: same as track, but phone plug
ECHO RETURNS:
1 volt RMS into 5K required, phone plug

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MARTIN DICKSTEIN

• Since we started discussing the 16mm film projector, back in August of last year when we mentioned how to take care of the projector, we have received a tremendous amount of mail related to this subject. In March of this year we printed a letter from a reader who shared his experience and knowledge of the subject. In June, 1974, we mentioned a projector which was a double system unit. In August of this year we again printed a letter from a reader who shared knowledge with the rest of us. In this column we are again going to quote some of the correspondence we received on the same subject.

This letter is from Kay Kibby, Advertising Manager of W. A. Palmer Films, Inc., San Francisco.

The German-made Siemens projector described by Mr. Dickstein is a unit which was first marketed some fifteen years ago and has not been manufactured for about three years. At that time, Siemens turned over their projector manufacturing rights for this machine to the Bauer Co. Bauer, in turn, rather than continue to manufacture the somewhat antiquated design of sound transport, redesigned the unit and now markets an improved machine under two different labels, their own brand, Bauer, and Sonorex, represented by the Arriflex Company.

We would like to call your attention to the American-made Palmer Interlock Projector which performs essentially the same functions as the Siemens and has been on the market for some nine years. We think the Palmer projector is easier to thread and less likely to damage film or sound track during running than its German rivals. It also has excellent recording and playback performance: 50 to 15,000 Hz plus or minus 2 dB and Peak Flutter (ANSI S 4.3-1972) plus or minus .1 percent.

The Palmer Interlock Projector also is powered by a synchronous motor, a vital feature for transferring sound from ¼-inch tape recorders with sync tone to full coated magnetic film.

We are enclosing a new product release and specifications describing our latest models, which should be of interest to your readers. They feature an innovation, Dynamic Sync Control, a Palmer exclusive which permits picture and sound track relationship to be adjusted while the projector is running.

Our sincere thanks to Kay Kibby. What follows is the release on the Palmer Projector:

PALMER SYNC CONTROL FOR DOUBLE SYSTEM PROJECTORS

A new line of interlock projectors makes film editing and previewing easier according to the manufacturer, W. A. Palmer Films, Inc. of San Francisco. Designed for double system running of picture and separate 16mm magnetic track, the projectors combine the Palmer magnetic film transport with a Singer Model 1020 projector.

The new models feature Dynamic Sync Control—an innovation which permits picture and sound track relationship to be shifted in either direction while the projector is running. This is particularly valuable when editing "lip sync" film, according to the company. Other standard projector features include synchronous motor, single lever forward and reverse operation, horizontal tilt control, 2-level projection lamp, and Kodak Ektanar projection lens for bright, sharp pictures. Made in the United States, the projectors are portable, easy to operate, and gentle on all kinds of film.

Four models provide various playback and recording combinations. A special solid state record amplifier comes with recording models. The amplifier unit is built into a compact base cabinet—and turns the interlock projector into a professional-quality 16mm recorder. The amplifier unit is also available as a separate item.

Palmer Interlock Projectors were first introduced in 1965, and are now in use throughout the world. Prices for the new models range from $2,730 for Model PGPS-2 Magnetic/Optical Interlock for playback to $4,120 for Model PGSS-3 Dual/Magnetic/Optical Interlock for recording, transfers and playback.

A PREAMP POSSIBILITY

As long as we're on the subject, here's a letter from a reader who would like to get some help in improving his projection sound system. Our thanks to Bob Baer. Following the letter, we'll remind you of some upcoming events.

I own an RCA model 400 16mm sound projector. It's the junior model, which means the amplifier is only seven watts.

I only use the preamp section and then go into my own equalization circuit which is followed by a line amp section and then finally followed by a Heathkit W4-M power amplifier. The final result is a nice sound, but it's somewhat noisy. I know 16mm optical film has a poor frequency response, but then again so does cassette tape.

I was wondering if you could supply me with a schematic of what *would be in your opinion a satisfactory preamp for reproducing 16mm sound. If this is not posible, could you refer me to a publication that would have such an item?*

The preamp circuit I'm using now I got out of the Audio Encyclopedia, page 267. It's the cathode equalizer circuit which, as I said before, is crude but adequate.

We should like to hear from all readers who can help Mr. Baer; we will publish letters and diagrams as they arrive. It's a great chance to get some ideas exchanged, and perhaps we will, in a near future issue, include specs and information on some models which come with factory incorporated preamp outputs for this purpose as well as presenting some ideas for outboard preamps which some of you may have already used successfully. We'll also listen to you if you took the preamp output of the projector amplifier (which did not come with a provided plug or receptacle) and matched it to another mixer input for feeding through a sound system. All ideas are welcome and will be reviewed and read. Once again, thanks to Mr. Bob Baer for asking.

Two upcoming conventions we suggest you make plans for are the *116th SMPTE Technical Conference and Equipment Exhibit*, November 10-15, 1974 at the Four Seasons Sheraton, Toronto, Canada, and the *36th Annual National Audio Visual Convention and Exhibit*, January 9-13, 1975 at the Las Vegas Convention Center. Nothing like being forewarned, we always say. For further details, the SMPTE is in Scarsdale, N.Y., and the NAVA can be found in Fairfax, Va. Please tell them you saw it here. ■

LONDON PACKAGE DEAL

Getting back to that missing A.E.S. convention report, there's another convention coming up. March 4, 5, 6, in LONDON. A package trip is in the works, and it will include round-trip air fare from New York City, a room at the convention hotel, and continental breakfasts. Departure will be on Saturday evening, March 1, with the return to New York on Sunday, March 8. The price will be about $430, give or take a few bucks. Complete details will be in the next issue, but it's not too early to start drooling about it right now. As an added benefit, the editors of **db** will be on board, so it offers an unprecedented opportunity to tell them exactly what's wrong with the magazine for as long as it takes to get to London. Think about that, boys and girls! ■

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PUBLISHER'S MEMO

Economic Future for the Professional Audio Industry

With economic pundits making dire predictions for the future of general business conditions, it might be a good time for the audio industry to think ahead and try to see what the future holds for us.

A recent seminar held by the Owen-Corning Fiberglas Corporation, while focused on the homefurnishings industry, shed light on some interesting figures which also bear significantly on our particular segment of the economy since so much of our ultimate product is basically consumer oriented.

It is projected that during the nineteen-seventies the nation's population would grow only 13 percent, but that the number of people between the ages of 20 and 34 would increase nearly 40 percent. Reflecting the fact that this span is the prime period for marriages, it follows that the next few years will see a burgeoning of new households; it is estimated that in 1980, 2.5 million marriages will occur, an increase of .5 million over this year. Added to that is the trend for young single people to establish their own homes.

Not only will the number of homes increase, but the spending power of the young householders will also rise. One factor is the decrease in the birth rate; smaller families have a larger share of income available to spend on discretionary purchases, such as home entertainment units. Many of these families will have two incomes; the proportion of married women who work is now 40 percent, with the number growing. The resultant income has lifted the number of families with incomes in excess of $15,000, the amount necessary under present conditions to provide leeway for luxuries, to 22 percent of all families, or about 12 million. It is estimated that discretionary income, above $15,000 per family, will top $200 billion, or 23 percent of total income.

Among the items that will be vying for this surplus money are the components for home entertainment: t.v. sets and more sophisticated sound equipment, tapes, and recordings. In fact, budget-harried families now are forming stay-at-home habits because they cannot afford expensive outside entertainment; this trend may well lead to the purchase of the equipment which increases their enjoyment of evenings at home.

All of this forms a rather bright prognosis for the consumer audio field. And it follows that professional audio, feeding into consumer products, will also feel an intensification of demand. So, although the economic atmosphere seems uncertain right now, it is likely that as the inflationary spiral, hopefully, eases in the second half of 1975, newly released discretionary funds in the hands of these young families will commence to flow into those consumer products which ultimately receive professional services and thence to the benefit of the entire professional audio industry.

Robert Bach

db new products&services

AUXILIARY HIGH FREQUENCY SPEAKER

● The frequency range of the Vocal Master sound system, (VA300-S or VA 301-S) from the same manufacturer, may be extended by high frequency speaker model VA305-HF to 18,000 Hz, providing true high fidelity response for vocal and high frequency instrumental reproduction. It is also suitable for the reinforcement of full-range musical material. The accessory speaker, designed to be placed on top of the Vocal Master column, uses twin horn-loaded, high-frequency dome drivers and an integral 5 kHz crossover network intended to be connected in parallel with the Vocal Master speaker column; up to three VA305-HF speakers may be used in combination with the speaker column with each main amplifier.
Mfr: Shure Brothers, Inc.
Price: $119.50
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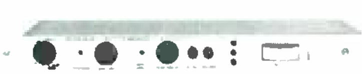
VERSATILE LIMITER



● A wide range of limiting or compression needs can be filled by model 201 average and peak responding limiter, including applications in studio recording, mastering, broadcast, film, and sound reinforcement. The average level and peak limiting functions operate simultaneously and independently. Front-panel controls are provided for all functions, including variable attack/release time and response action. The limiter features an open-loop control approach and distortion-reducing circuitry.
Mfr: Inovonics, Inc.
Price: $480.
Circle 41 on Reader Service Card

MODULAR CONTROL CONSOLE

● Designed to be used for either fixed or remote recording as a production or on-the-air console, model 1604 will accommodate 16 inputs, 4 echo channels, 2 foldback circuits, 4 output channels, 4 submasters, 4 speaker monitoring, slate, tone, and intercom circuits, and audition and cue facilities. These include the necessary foldback, audition, intercom and program interlock features for broadcast application. Model 1604 may be equipped with factory pre-wired options at any time, adding remote control of tape machines and turntables, or remote input pre-selection. Table-top mounted or free standing, with plug-in external connections. A variety of interchangeable modular equalizers is available.
Mfr: Automated Processes, Inc.
Circle 42 on Reader Service Card

DUPLICATOR WORK MASTER RECORDER

● A shifting head assembly on model 511 eight-track, one-inch recorder eliminates the need to use a conventional eight-track studio recorder for duplicator work master production. In addition to the eight-track two channel shifting head assembly, the unit is comprised of two single channel reproduce/record amplifiers, and a 7½-15 i.p.s. transport. Circuitry is solid-state. The recorder is available either as a cabinet top model or as a floor console unit.
Mfr: Audio/Tek Inc.
Price: Table top: $4,850.
Floor console: $4,950.
Circle 43 on Reader Service Card

DIGITAL DELAY SYSTEM

● Featuring a 90 dB dynamic range, Delta T model 102 series digital delay system has up to five delay outputs, each independently adjustable on the front panel. Additional slave units (up to 50 outputs) may be cascaded for long delay requirements with no degradation of audio output. The system offers up to 320 ms. of total delay per main frame in 5 ms. increments in 40 ms. modules (or up to 128 ms. delay per main frame in 2 ms. increments in 16 ms. modules). A five-position l.e.d. headroom indicator verifies correct operating settings. Transformer coupled inputs and outputs are standard. Modular options are available for expansion.
Mfr: Lexicon, Inc.
Circle 44 on Reader Service Card

OPERATIONAL POWER AMPLIFIER

● Repairable d.c. operational power amplifier model 433 may be used as a medium audio power (16 watts peak continuous program) amplifier as well as an earphone monitor or servo. Capable of 4 watts rms at a total harmonic distortion of 0.5 percent, it features a plug-in semiconductor and i.c. repairability. The device measures 1 x 3 x 2 inches on a ¼ inch anodized aluminum heat sink.
Mfr: Opamp Labs, Inc.
Price: $35.00 (1-99)
$30.00 (110-999)
Circle 45 on Reader Service Card

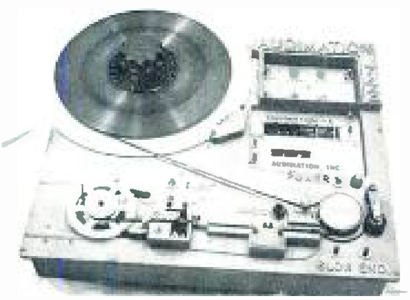
SERIES SEQUENCER

● The ability to quantize-digitize pitch, permutate, retrograde, and initiate multiple envelopes within a single note is claimed for model 400 modular series sequencer, which features voltage controlled envelope generators. Simultaneous independent control of pitch, timbre and loudness is also possible on every note of a sequence; envelope accents can be produced at any stage of a sequence. The basic sequencer is a 16 stage by 3 control voltages per stage unit, equipped with a voltage control clock, logizers, gated trigger, power supply, input and output jacks, and the quantizer unit. Any number of 16 x 3 program panels, (the manufacturer's model 416) may be added to the basic sequencer unit to increase its length and/or depth. An additional module, model 401, with further capability including two voltage controlled audio oscillators, can be added to basic model 400.

Mfr: Electronic Music Laboratories, Inc.

Circle 46 on Reader Service Card

DIGITAL LOCK SYSTEM



● Designed as an integral part of the manufacturer's 742 cartridge winder, the Digital Lock System avoids false cuts by discriminating as close as ½ Hz from the designated cue tone frequency, regardless of recorded levels. The principle behind digital lock is the matching of "signature keys," the cue tone and a pre-set tone; the meeting of these two unlocks and activates the stop-cut-eject sequence of the winder. It can be optionally equipped to react to pure silence, the absence of program material. In addition, it can be modified to permit the detection of encoded information as well as the basic cue tone. Controls on the 742 winder allow for independent setting of high pass and low pass cue frequencies from 3 Hz to 50 Hz. Adjustment is also provided to select the length of cue tone and the immunity to distorted wave forms. The system has a built-in safety feature which automatically stops the winder if a cue tone is missing or distorted beyond recognition.

Mfr: Technicals, Inc.
(Audimation, Inc.)

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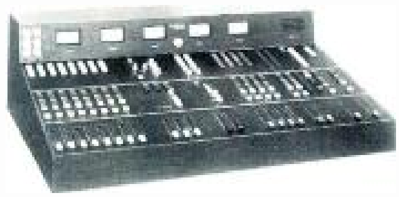
TELEVISION AUDIO CONSOLE



● Built to the specifications of television engineers, state-of-the-art model 1632 is also adaptable for use in a.m. and f.m. monaural broadcasting. Four inputs per mixing channel provide a total of 64 inputs. There are two monitor output channels for separate studio and control room monitoring with muting relays, a headphone output from the control room monitor, a cue channel with built-in speaker, talkback facilities for two studios with built-in mic, and slide-wire attenuators and illuminated pushbutton switching. Model 1632 also features equalized reverb send and return facilities. The unit uses all solid-state i.c.-op amp electronics on plug-in p.c. boards. It has an aluminum cabinet. A number of optional features are available, which can be easily added to the basic unit.

Mfr: Robins/Fairchild

Price: $9,995.

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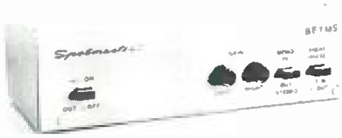
TURNTABLE PRE-AMP

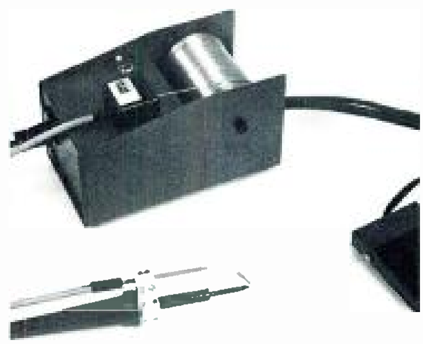


● A phase reversal switch on one channel that allows five modes of operation—mono in/mono out, stereo in/mono out, stereo in/stereo out, dual channel mono in/dual channel mono out, and single channel mono in/dual channel mono out is a notable feature on Spotmaster model BE TMS turntable pre-amplifier. The BE TMS has an output level switchable between −10, 0, and +8 dBm into a nominal 600 ohm load. Transformer coupled output is available as an option. Gain of the pre-amp is 54 dB at 1 kHz and frequency response is ±2 dB, 30 Hz to 15 kHz (RIAA) NAB. Also featured are channel separation better than 50 dB and distortion less than 0.5 percent at +8 dBm. Controls monitor power, left and right channel gain, mono/stereo selection and right channel phase reversal.

Mfr: Broadcast Electronics, Inc.

Circle 49 on Reader Service Card

AUTOMATIC ONE-HAND SOLDERING



● Jigging and clamping, necessary for precision soldering, can be eliminated, with the components hand-held, with the use of the Freehand Industrial Solder Feeder, a one-handed device which features automatic, motor-driven feeding of solder through a universally adjustable feed tube directly to the iron's tip. Actuated by a floor pedal, the unit feeds from as much as a five-pound spool of solder mounted on the motor housing. Interchangeable drive units allow use of all sizes of solder with the same basic feeding unit. Pre-soldering of terminals and repetitive parts handling is also eliminated because leads can be tinned and soldered in one operation. It is claimed that the device permits quicker, individual positioning of additional solder for desoldering. Constructed of 16-gauge aluminum, the unit includes an off-on power switch, a pilot light, and grounded receptacle.

Mfr: Schurman Products, Inc.

Price: $140.00.

Circle 50 on Reader Service Card

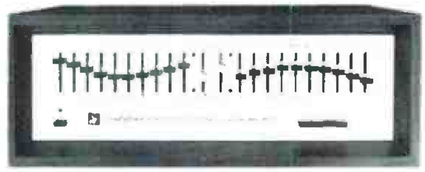
SOUND LEVEL RECORDER

• Continuous monitoring and permanent recording of sound pressure levels as a function of time is accomplished with the db-402 sound level meter-strip chart recorder system. This is achieved, without constant operator surveillance, over a 50 dB dynamic range, with data presented in a linear wide four inch chart format. Selection of six amplitude ranges, each 50 dB wide, over the input range of 35 to 130 dB allows continuous recording of most signals without adjusting the range switch. The device, which weighs eight pounds, is completely portable.
Mfr: Metrosonics, Inc.
Circle 51 on Reader Service Card

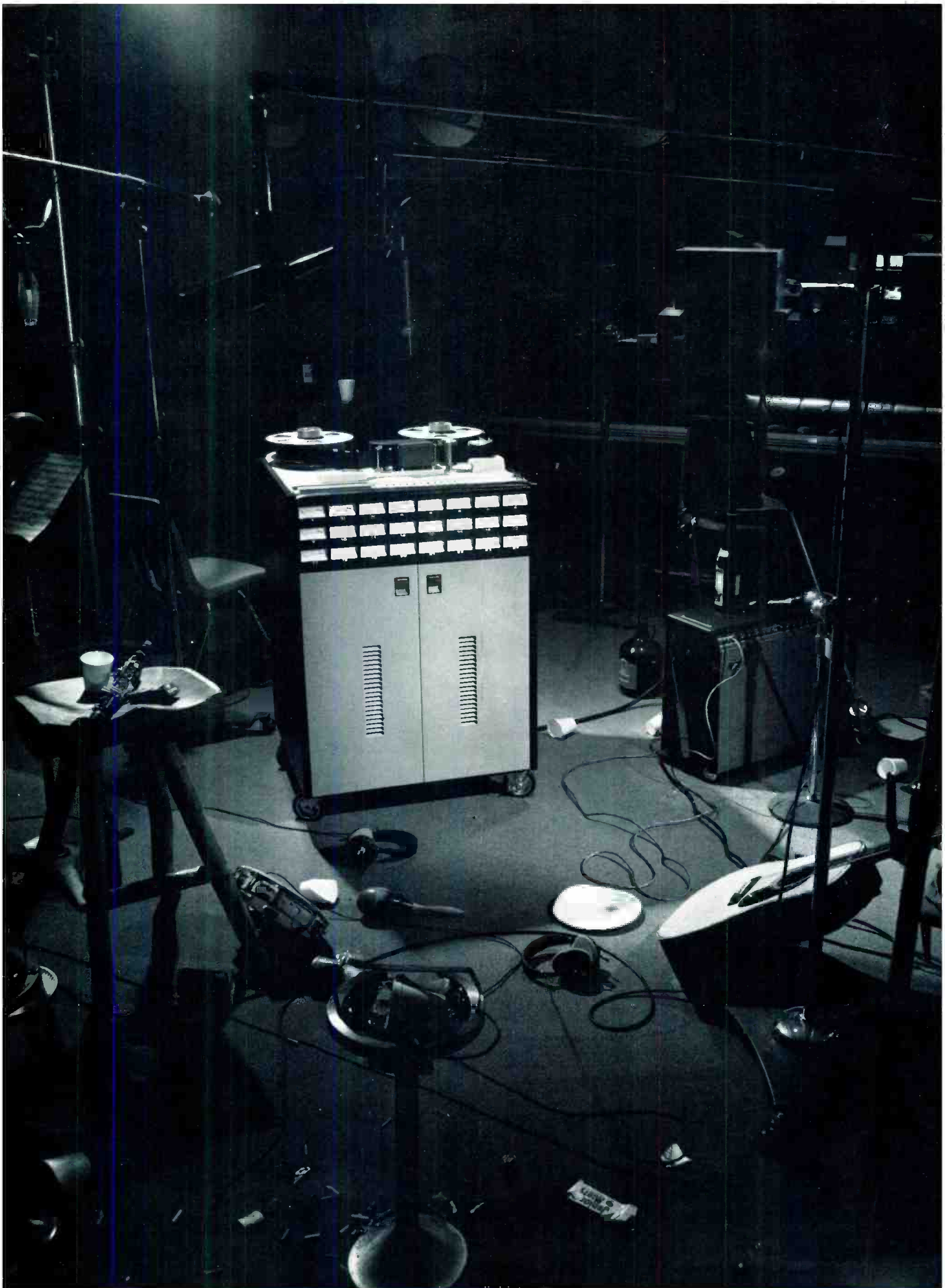
UNIVERSAL ELECTRONIC COUNTER

• Model 7026 electronic counter, with a bright .55-inch planar readout visible at a distance of several feet, offers a total display range of 1 to 1999999, a frequency range of 5Hz to 50 mHz, and is guaranteed accurate to ±1 count. It's engineered for line voltage operation with a power requirement of 117/234VAC, ± 10 percent, 50-60 Hz, approximately 25 watts, and has an operating temperature range from −40 degrees C. to +70 degrees C. The sample rate is continuously adjustable from five readings per second to hold. Hold input on an optional rear panel connector provides sampling by either contact closure or external potentiometer with display storage being selectable by rear panel switch.
Mfr: Simpson Electric Company
Circle 52 on Reader Service Card

RECORD/PLAYBACK EQUALIZER



• Model RP2212 equalizer may be plugged into any receiver or preamp providing tape monitor inputs and outputs or it can function through its own tape monitor inputs and outputs with front panel pushbutton selection. Four l.e.d.s provide a visual front panel display for balancing input to output signal ratios. Front panel pushbutton selection also provides either an equalized or unequalized output for speaker/room equalization or equalized tape recording; separate outputs are provided for tape recorder and amplifier hook-up. The unit incorporates two completely separate ten-octave equalization panels, with plus or minus 12 dB boost and cut provided individually for each octave. Separate equalized signal zero-gain controls are used for each channel, enabling exact balancing of input to output with a +6 dB and −12 dB range.
Mfr: Soundcraftsmen
Price: $349.50
Circle 53 on Reader Service Card



A very rugged tape recorder for a very delicate business

Ampex designed the MM-1100 multi-channel tape recorder with your business in mind. We've been building professional recorders longer than anyone else, and we've learned that producers, engineers, and studio operators have a lot to worry about. So we build the one piece of equipment you can plug in and forget.

Coddle your talent, if you must, but shove around your MM-1100
That's right, shove it around. That's why we put wheels on the MM-1100. Dolly it from studio to studio, or truck it across town. The heavy cast frame and solid steel cabinetwork will keep all the little parts and things where they belong, and the only "installation" routine you'll go through is to plug in the power and input lines.

Pinch your pennies, hoard your dollars, but squander your discrete channels
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
The tape transport in the MM-1100 is a real grizzly bear. It has plenty of control power, even when the reel is a sixteen-incher loaded with 2" tape. And even with all that mass to control, the capstan servo zeroes in on synchronizer commands fast enough to use two MM-1100s in parallel for the 46-channel stuff you always wanted to try.

Your maintenance guy can work banker's hours because our "doctor" makes house calls
Ampex field service sells a lot of equipment for us. We build a fantastic machine, and it almost never develops a problem. If it does, however, you'll appreciate our policy of having service engineers who know both our product and your industry. They're available to keep your group together and working.

The business gives you a headache; Ampex gives you a tape you can sell
Full technical specifications are in our MM-1100 brochure, which we'll be glad to send you free. Working models are in studios all around you, and if you don't have a friend who'll let you inspect his MM-1100, we'll arrange a demonstration. In fact, we'll do anything we can to get you rolling in the multi-track business, if you'll just tell us how we can help.

In a delicate business like production recording, there just isn't any substitute for the Ampex MM-1100—a very rugged tape recorder.

AMPEX

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

Circle 36 on Reader Service Card

WILLIAM E. ANDERTON

Professional Sound Recording A British View – part 1

The present and future of the commercial recording industry, as seen by the Assistant Editor of Wireless World magazine.

NO DOUBT that music and commercialism go hand in hand, but this commercialism has led to the vast industry involved with sound recording. The growth of pop music over the last two decades has made more resources available to development engineers. And, there have been new demands made on equipment from discerning recording engineers and artists. The result has been more money, better performance and more facilities.

Recording complexities have been built up at such an incredible rate that it has become necessary to reconsider the industry as a whole, to ask why or if complexity is necessary and how simplification is possible. My purpose here is to provide a unified view of the developing recording industry, to show how demands from engineers and artists have led to changes in recording techniques, and to reveal the influence this has had and will have on equipment performance and the facilities deemed necessary to produce a sound recording of the highest quality.

An idea of the resources necessary to equip a large sound studio is provided by a contract recently received by Rupert Neve and Company to supply six recording consoles to the world-wide operations of EMI. The consoles have a total value of over $625,000!

WHY EQUIPMENT COSTS RISE

The main reason for the growth in equipment cost has been the development of multi-track recording from 2 tracks to 4 to 16 and now 24 tracks, with 32 forthcoming. The pop world has been the chief cause because the final product is more dependent on intermediate electronic processing than the original sound. So that this processing can be performed with the most flexible control over the finished product, it is necessary to separate the information to be recorded as much as possible. Close miking techniques can provide a multi-track tape with one instrument or sound source per track with good separation and all easily accessible for processing.

Commercialism again plays a major part in this type of facility being commonplace. If a recording artist appears at a studio and asks for 24-track facilities—necessary or not—and they are not available, he will take his money elsewhere. Fortunately, this type of demand and supply has not lowered equipment performance. Specifications of professional equipment are not questioned; it's the use of this equipment and the reasons for its development which must be examined.

Future possibilities are staggering. As the cost of mini-digital computers falls to around one-fifteenth the cost of a comprehensive mixing desk alone, nothing will prevent the use of on-line computers in studios capable of being programmed to accept input from a control potentiometer and converting it to any desired control instruction. Digital equipment will become more and more a part of the recording process and in fact is already appearing in the form of automatic mixing facilities and digital delay lines.

Capabilities offered by many new devices almost point to the redundancy of the recording artist. One such device works out the fundamental frequency of an input signal and converts this to a related d.c. voltage of the particular input. You can feed this with another output voltage proportional to the input waveform average amplitude into a voltage-controlled synthesizer that's pro-

This article was adapted from an article which appeared in *Wireless World*, June, 1974.

CHARACTERISTICS OF TOTAL RECORDING SYSTEMS

	Condenser pressure gradient microphone	Multi-channel sound mixing console	Channel control amplifier	Noise reduction unit	Multi-channel tape recorder (15 ips)	Disc cutter and associated amplifiers
Frequency response	30Hz–16kHz ± 2dB	20Hz–20kHz ± 1dB	15Hz–20kHz ±0.5dB	30Hz–20kHz ± 1dB (record/replay)	60Hz–15kHz ± 1dB	40Hz–16kHz ± 1dB
Sensitivity	2mV/dyne/cm²	Adjusted between -80dBm and + 10dBm		OdBm line in and out	Max input + 22dBm	Recorded level settable in steps up to + 8dB
Distortion	0.5% at 128dB SPL	T.H.D. at 20dBm into 600 from any output 0.075%	mic. input + 20dBm and 1kHz is 0.01%	0.2% from 40Hz–20kHz at + 8dBm	1% max (NAB)	0.3% at 1kHz (cutter drive)
Noise reduction				10dB from 30Hz to 5kHz rising to 15dB at 15kHz		
Noise level	Self-noise -17dB ref. 2 × 10⁻⁴dyne/cm²	-80dBm residual output noise	-125dBm equiv. input noise		62dB (s/n) (NAB)	Better than 70dB (signal to rumble)

This table compares some of the common characteristics of a total recording system from microphone to disc. These are only intended to provide the order of magnitude of specifications of a typical professional system. The parameters shown are not necessarily the most important for the assessment of an individual piece of equipment.

grammed to produce the waveform characteristic of *any* predetermined instrument. The result is whatever instrument the producer cares to record, not what was played originally. Rather a far-fetched idea but indicative of the control now offered by instrumentation over signal processing.

Again, the point is that this type of facility is well within the capabilities of circuit design—it's the use of the final product which is questionable.

MULTICHANNEL RECORDING

The essence of recording 16, 24, or even 32 tracks simultaneously is to treat the recording stage as simply an information-gathering process, concentrating purely on achieving a good signal-to-noise ratio on each tape track and not considering balance at all. If 32 microphone signals are to be condensed to a 16-track tape, some mixing will be involved, but the extent of such mixing will be minimal. Certain economic and practical advantages then follow.

Since obtaining an effective sound balance may be time consuming, deferring this process to separate mixdown (final mixing of the multitrack tape to the stereo or quad master) sessions is more economical. Reason: the musicians do not have to be paid for repeated replays. Overdubbing is now also possible in which some parts of the music are recorded separately, while musicians are fed with the already recorded sound via headphones.

This recording system demands the monitoring of sound during recording to provide an estimate of how the final stereo or quadriphonic mix will sound. An independent mixer must be provided having as many inputs as there are tracks and as many outputs as there are monitoring loudspeakers. During the recording, a trial balance may be achieved on the monitor. For this reason, stepped-level controls are often provided to facilitate logging. Where the overdub technique is employed, the input to the monitor mixing matrix consists partly of prerecorded signal and partly of console output.

Since the multitrack technique presupposes a substantial degree of separation between musicians, a further essential provision is the "foldback" mixer, or cue mixer, to provide performers with a headphone signal enabling them to keep in time.

MIXING AND WHAT IT ENTAILS

Mixing the signals from the master multitrack tape is performed either with the same console or, in studios, in a tape-mixing room having a console adapted specifically to this purpose.

The work of building up a good mix is lengthy and tedious, being a process of trial and error, dependent to a large extent on the engineer's memory and endurance. For this reason, attempts are being made to provide the tools for time saving, ease, and greater accuracy of operation.

The automated mixdown process is simply one in which the controls on the console are linked with voltage-controlled devices. Provision is made for logging the settings of these controls in digitally coded form. The digital code may be recorded on one of the tracks of the multitrack master tape, thus insuring an accurate synchronized store of information relating control operations to the program. By re-running the recording, individual operations may be modified by means of an updating facility.

The type of facilities available from a 16-track mixing desk would be as follows: 24 input channels, each having line and microphone inputs with comprehensive equalization and faders on each channel; eight output mixing groups (each group provides a combination of sub mixes) with remix facilities plus eight output tie lines for 16-track recording; four limiter/compressors with linking of control lines for stereo or quadriphonic operation; four equalizer units terminated on a patch panel; four reverberation groups (send and return) with pan pots and group selectors on the return lines; four foldback or cue

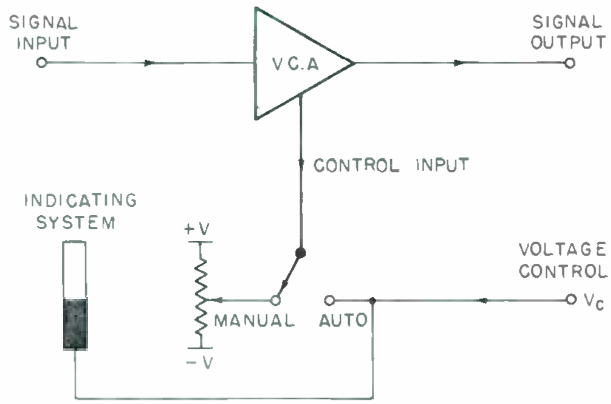


Figure 1. A simplified system using voltage-controlled amplifiers as the faders themselves.

groups: a four-speaker monitor system with 16-track playback to the monitor matrix; four studio playback outputs, up to 21 vu meters and pre-fade listen available on all input channels (this facility allows the operator to monitor a particular input or group prior to feeding it to the main output) plus patching and talkback facilities with pan potentiometers on all input channels and monitor tracks.

If this list were extended for 24-track recording, it's obvious that the amount of control required when all input channels are in use calls for the automation facility—not to replace or devalue the status of the recording engineer but to provide a useful memory store of all level settings and their position in time.

AUTOMATION

An automatic mixing facility will encompass the simultaneous recording on spare tracks of the master tape of digital control data derived from the analog signals being handled by the mixing desk. On replay, the control data resets the desk levels to produce the previously achieved audio mix. The system must be capable of being easily updated if necessary.

Control of faders can be achieved by one of two systems[1] (although it is believed that a completely different system will shortly appear, of which no details are available at present). The first is a servo-controlled conventional fader operated either manually, or positioned by a motor powered by a small servo amplifier. Two inputs to the servo amplifier would be a control voltage and the other a feedback control voltage to stabilize the servo system. The main advantages of this system are low noise and obvious operation, the fader knobs physically move in sympathy with the data input to the servo amplifier. Mechanical complexity and the need for a power amplifier requiring relatively high current tend to offset the advantages.

A second system uses voltage-controlled amplifiers as the faders themselves. Frowned upon in the past for their poor stability and noise performance, v.c.a's are now available with low inherent noise, low distortion, and high gain. An indicating system must be provided if the v.c.a. is to be used as a practical fader so that the effective position of the fader can be easily read at any time. Light-emitting diodes can be included to indicate the control voltage and hence the effective slider position. The advantages of the v.c.a. are low cost, low current requirements, a high level of stability, and a low level of complexity.

The v.c.a. circuit may not itself be complex, but the control unit must be. The fader control voltages are multiplexed and converted into a digital signal for recording on spare tracks. The control unit must then decode replayed signals, convert them back to analog form by sample and hold circuits and provide adequate timing facilities. Some form of error correcting code must also be supplied because sound recorders use tape that is inferior compared with computer tape, in that a higher level of dropout can usually be tolerated.

It will be some time before automatic mixing facilities become a common sight in recording studios. One possible problem may be the use of different methods for coding and decoding the control signals on different desks. This would mean that a tape recorder using one facility must have its final mix prepared using the same facility, and cannot be taken to another studio for the final mixdown, as sometimes occurs.

MULTITRACK AUDIO IN VIDEO RECORDING

Recording a video program and the associated audio signal on separate machines has this advantage: The audio tape can be handled by sound engineers using multitrack facilities until it is ready to be dubbed back on the edited video tape. The television industry has been slow to adopt this technique, although the film industry has used it for a number of years.

Problems of synchronizing separate video and audio recorders have only recently been solved by the use of an 80-bit digital code generated 30 times a second (for a 60 Hz field rate) known as the SMPTE code.[2] The main concern of the code is to provide a means of gaining some increase in quality and flexibility of the audio portion of the program by standardization of synchronizing codes and allowing electronic editing to be applied to the audio tape.

It was necessary to develop the facility for audio tape recorders to use the code in a manner similar to that handled on vtrs. The audio machines could also be used in conjunction with the automated editing systems for post-production work. As a result, a synchronizer code-reader is necessary to compare code signals on a frame-by-frame basis and then generate an error signal to the d.c. capstan servo that is a function of the difference between any source of master code (vtr) and the audio tape recorder.

The code-reader has two functions, the first being to demodulate and decode the slave (audio recorder) serial time code, the second to display the slave or master codes for manual parking of the slave and master machines. The reader's ability to read code from 1/5 to 40 times play speed makes it a useful tool in the "search" mode. The synchronizer contains the necessary circuitry for providing a control voltage to the audio tape recorder capstan servo circuitry. ■

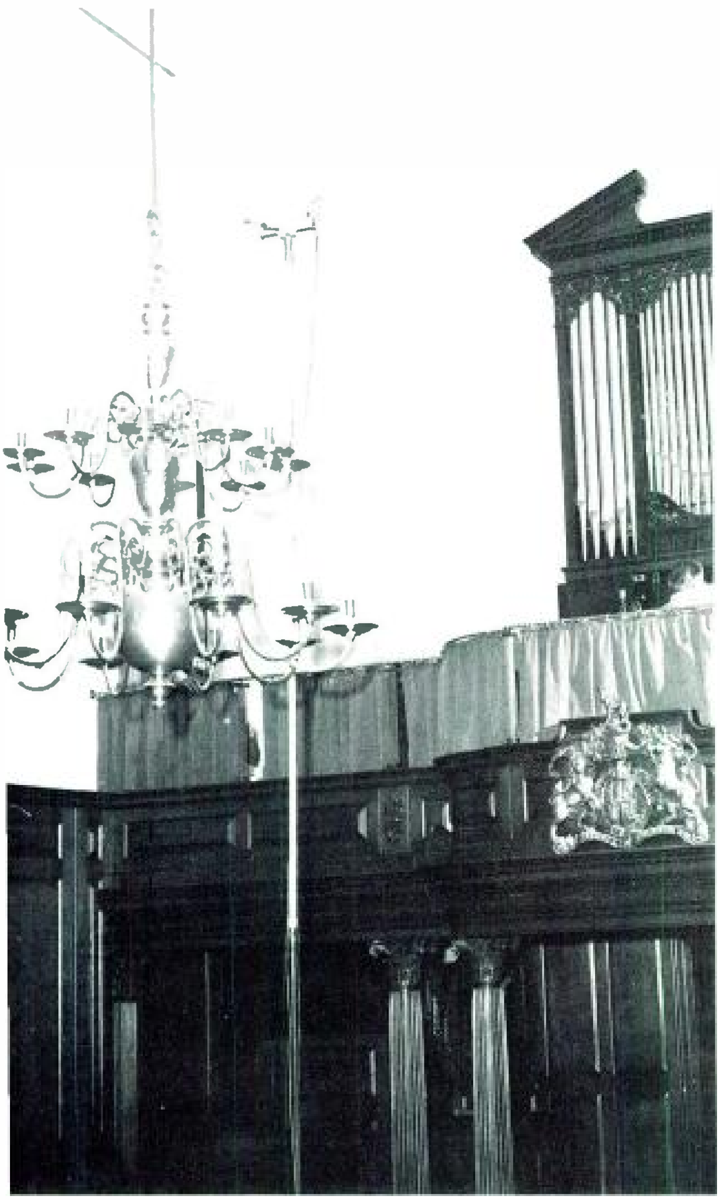
References

1. Curtis, M. S., "Designing an automated mixing console," *Studio Sound*, July 1973, Aug. 1973, p. 36, p. 24.
2. Nothaft, E. M. and Irby, T. W., "Multitrack audio in video production," *JAES*, April 1973, pp. 172-176.
3. Blakely, L., "Using Noise Reduction to Reduce Disc Surface Noise," *Recording Engineer/Producer*, Dec. 1973, Vol. 4, No. 6.
4. Swettenham, R. W., "Vehicles for Multitrack Sound Recording," paper presented to 47th AES Convention, Copenhagen 1974.
5. Jones, A. H. and Bellis, F. A., "Digital Stereo Sound Recorder," *Wireless World*, Sept. 1972, Vol. 78, No. 1443, pp. 432-435.
6. "Sound Recorder uses P.C.M.", *Wireless World*, Nov. 1973, Vol. 79, No. 1457.
7. Sato, N., "A new type of audio magnetic tape recorder," *JAES*, Sept. 1973, Vol. 21, No. 7, pp. 542-547.
8. "It is important that everyone in your studio should read this." *APRS* publication, 1972.

RICHARD B. TISDALE

Recreating Colonial Sound

Recording eighteenth century music in authentic surroundings is a special challenge in Williamsburg



Strange juxtaposition . . . modern electronic equipment placed near an eighteenth century chandelier. E-V RE-15s mounted on a speaker stand.

YOU WANT to make a recording of a recital on an eighteenth century organ installed in the chapel of the Wren Building at the Colonial Williamsburg restoration in Virginia. This is a logical inspiration, as part of the program of producing phonograph records embarked upon by the restoration's Foundation in 1967 in order to enhance the sense of history evoked by Williamsburg with the sounds, as well as the sights, of the past.

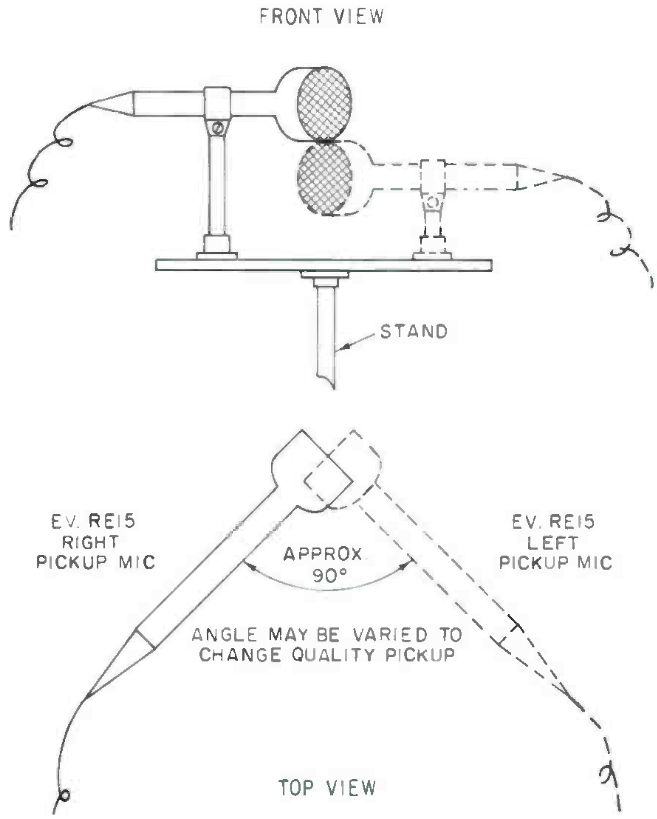
The idea is pretty simple, a recording of an organ recital. However, there are certain criteria which have been imposed by the Williamsburg recording program in order to achieve the main objective, authenticity.

The road to authenticity when making a recording of an eighteenth century organ in a seventeenth century building has several turnings and obstacles as modern technology mingles with colonial music. First of all, it had been determined when the project was embarked upon that all recordings must be made in the original buildings, which, while pleasing to the eye, were not exactly designed with twentieth century acoustical needs in mind. Then, the very fact that these buildings were being used as museums for a lively tourist trade created a practical condition, the need to use the building when it was cleared of camera-toting visitors with their attendant cacophony of footsteps, Junior yelling for souvenirs and Father pedantically and stridently instructing his young in American history. Therefore, the recordings were often made late at night, during those snatched intervals when the schedules of the musicians and the availability of the building merged.

The original instruments, of which the eighteenth century organ was one, also presented a challenge, as did the original musical scores, which could not be changed. Designed for eighteenth century ears and not for reproduction, the resultant sound cried for the usual electronic embellishments, such as echo, limiting or compression, etc. But that would interfere with the passion for authenticity. The only concession to such tampering was some minimum equalization permitted during mixdown or sweetening sessions.

The Wren Building, in which this particular recording was made, is the oldest original college structure in America (1697) and is exhibited by Colonial Williamsburg in

Richard B. Tisdale, Jr. is an audio engineer with the Audiovisual Department of the Colonial Williamsburg Foundation.



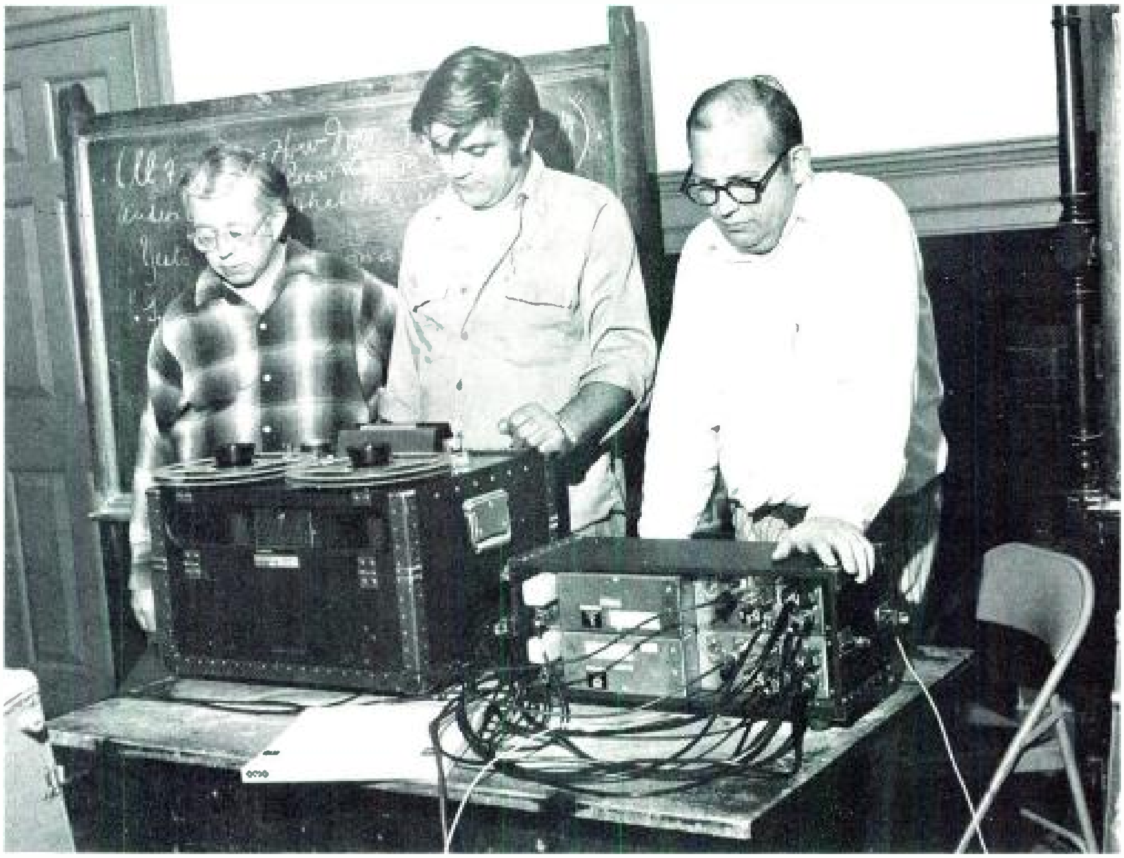
Angles of pickup.

cooperation with the College of William and Mary. My only previous experience with the Wren Chapel was recording motion picture dialog. The room has a three- to four-second decay time. During the movie experience, the director wanted a long wide shot with sync dialog and no radio mics!

Fortunately, long decay time is acceptable in recording classical organ music. My goal during the organ recording became one of simply controlling the level of the decaying sound. At first I was tempted to use two omni-directional condenser microphones placed close to the organ case. But this would have created an undesirable ping pong effect, hardly in keeping with our aim of achieving a sound as close as possible to that which would be heard by someone in an 18th century congregation.

Since I had never tried crossed cardioids on a stereo pickup, J. S. Darling, the organist and Colonial Williamsburg's music consultant, my assistant, Bill McAllister, and I went to the chapel one evening prior to the actual recording date to experiment with microphone positions and types. I was equipped with two E-V RE15 cardioids, two cardioid dynamics of another manufacture, and my trusty omni condensers. First, I tried the RE15s about ten feet in the air and about twenty-five feet back from the organ. The sound we heard was amazing, very lifelike, with good tone and feeling, nice stereo with depth, no ping pong. There was just a little too much overhang muddying up the sound and I felt we needed a bit more presence. The mics were moved in closer, to about fifteen feet from the organ pipes and at the same height. Everyone liked the results, so a thirty-minute demo tape was made to play for interested parties. Incidentally, we were so pleased, the other mics never came out of their cases.

The recording was done on an Ampex AG440-B, using their plug-in preamps rather than hauling in a mixer for just two mics. We monitored with Ampex 620 speakers. (I don't usually use these, but their portability was desir-



Listening to playback . . . l. to r. J. S. Darling, organist, Bill McAllister, R. B. Tisdale.

able during the tests and they gave us enough information to make decisions.)

When it came time to do the final recording, I decided to use the same setup—a quarter-inch two-track Ampex AG440-B with its own preamps and the 620 speaker/amps, but adding quality 15-inch coaxial speakers. The monitoring room was a classroom in the Wren Building which is still used today as a modern classroom. This room gave a bad flutter echo and a longer decay time than desirable, but we had no other options.

After getting everything set up, I played the demo tape and adjusted the 620 EQ control and the speaker positions for the best reproduction, as judged by Messers Darling, McAllister and myself. The monitor level was set as high as possible but nowhere as high as rock monitor levels.

The E-V RE15s were placed fifteen feet from the organ pipes and raised to about twenty feet above the floor, which put them about on a level with the throats of the pipes. This added some presence and helped reduce mechanical noise from the tracker system. Other than that, no changes were made. The complete album was recorded in about four hours.

The sweetening session was minimal. About the only thing I did was to dub the masters, adding two dB at 100 cps and two dB at 10kc (my apologies to the Hertz people but you can't teach an old dog too many new tricks) and putting everything in the correct order with leader tape.

By now I know many readers may be asking why didn't I use one of those condenser mics with the stacked cardioid capsules, one over the other? I hope you will agree that the following reasons are valid.

1. Our department doesn't have one (although, I could have borrowed one). We do have RE15s.

2. The stacked condenser mics are large and unwieldy and require an a.c. power supply.

3. Most important of all, they have no low frequency roll-off switch. Years ago when I was doing radio remotes I quickly learned that it is better to limit the low frequency energy picked up by the mic and not risk overloads and resulting muddy sound. If you want more bass, you can add some with EQ as long as you don't go overboard. Some people may take exception with me, but I happen to agree with Lou Burroughs, that sometimes you simply don't need all that bottom end. The RE15s seemed to fit the bill; I had previously used them for some music and quite a bit of motion picure dialog recording. That, coupled with the published curves, persuaded me to take a chance with them and it paid off. Incidentally, the backup mics I carried with me for the first test are pretty flat from one end of the spectrum to the other, although the omni condensers have three positions of low roll-off.

In mastering for pressing, I gave the studio no specific instructions other than to send me a test lacquer before pressing. We were very pleased with the results and ordered the plant to press our initial order.

I considered myself extremely lucky in that I was fortunate enough to have a room, the Wren Chapel, that matched the quality of the instrument—this is not always the case in restoration recordings—and microphones that achieved the quality for which we were aiming. But whether matters go smoothly, as in this case, or are beset with situations with their particular Colonial headaches, it is always a genuine pleasure to participate in the recreation of long-ago sounds, adding to the education and entertainment offered by our town-size museum—Williamsburg, Virginia. ■

RICHARD S. BURWEN

A Unique Recording and Reproducing System

The author describes a truly unique and quite effective room that has been built for both recording and reproducing sound of the highest possible quality.

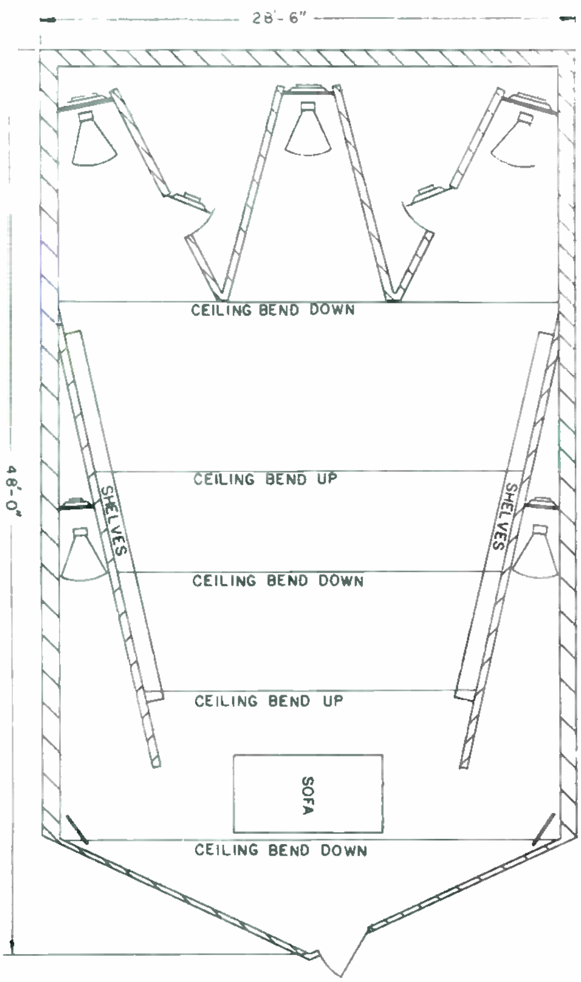


Figure 1. The basic room plan as described.

UNDER CONSTRUCTION for the past 11 years and now nearing completion is what I have intended to be the ultimate home-recording and reproducing system. This system which is built into a 48-ft. horn-shaped room, includes wide dynamic range recording and playback electronics now approaching 2,000 transistors and 2,000 operational amplifiers, an 8-kilowatt automatic color lighting system, and a 70-channel recording mixer. The sound emanates from five 13-ft. horns incorporating a total of 169 speakers driven by the equivalent of 20,000 watts.

My own objective with regard to the quality of the reproduced sound is to achieve maximum entertainment value. Judging from comments of musicians who have listened to their own recorded sound here, it seems possible to improve upon the original. Nevertheless, as a point of departure, the system should be capable of making small groups of instruments sound as though they were actually playing in the same room.

In regard to stereo, I have always felt that the appropriate number of channels is not two or four but either three or four. Accordingly, the system has been built with three speaker systems for the front channels to avoid a hole at the center while the two rear-channel speaker systems are designed to produce primarily reflected sound. All five speaker systems are utilized to various degrees whether reproducing one, two, three, four, or five channels.

ROOM ACOUSTICS AND LIGHTING

The room has been designed primarily for listening but is also used for recording without changing the acoustics. The shape of the listening area is essentially a horn with the listener positioned at the throat. At the front and along the sides are five conical speaker horns each 13 ft. long with a 64-sq. ft. mouth. Diagrammatically, the horn arrangement is shown in FIGURE 1. Three front horns are

Richard S. Burwen is president of Burwen Laboratories of Burlington, Massachusetts.

Figure 2. A recording session in progress. The East Bay City Jazz Band is situated in the three front horns.

shown in the photo, FIGURE 2. This room and a well-equipped electronic labortaory is built into the basement of my home in Lexington, Mass., and the house was designed and built around it. Because of the 10 ft. 6 in. limitation on ceiling height, the ceiling has been made wavy to diffuse the sound. There are no parallel surfaces and the room is unusually live, having a reverberation time at mid frequencies of approximately 0.8 sec. Because of the excellent high-frequency response of the system, the overall shape of the room, and the focusing and reflective effects of the horns on the middle and high frequencies, it is possible to utilize a somewhat longer reverberation time than in more conventional shapes.

The room is constructed entirely of concrete and cinder blocks for the walls and horns with extra heavy plaster on the ceiling. The only sound absorption is provided by records, tapes, books, equipment, three pieces of upholstered furniture, and two special 4 ft. x 20 ft. panels. These panels consist of a sandwich of hard plastic and acoustic tile mounted on concrete and are designed to absorb middle frequencies while reflecting high and low frequencies.

SPEAKER SYSTEMS

At the end of each of the five horns is an array of speakers consisting of two 16-in. woofers, a midrange horn with two drivers having 4-in. diaphragms, and 30 tweeters. In addition, the left front and right front horns have a pair of 24-in. woofers with feedback windings mounted on their side doors. The location of all speakers at the

Figure 3. A rear view showing the equipment racks. The author at the equipment rack.

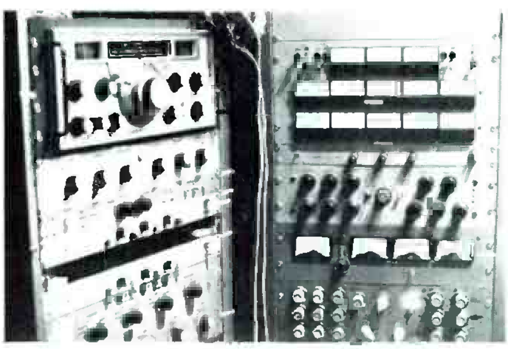


Figure 4. Racks with special peak vu meters.

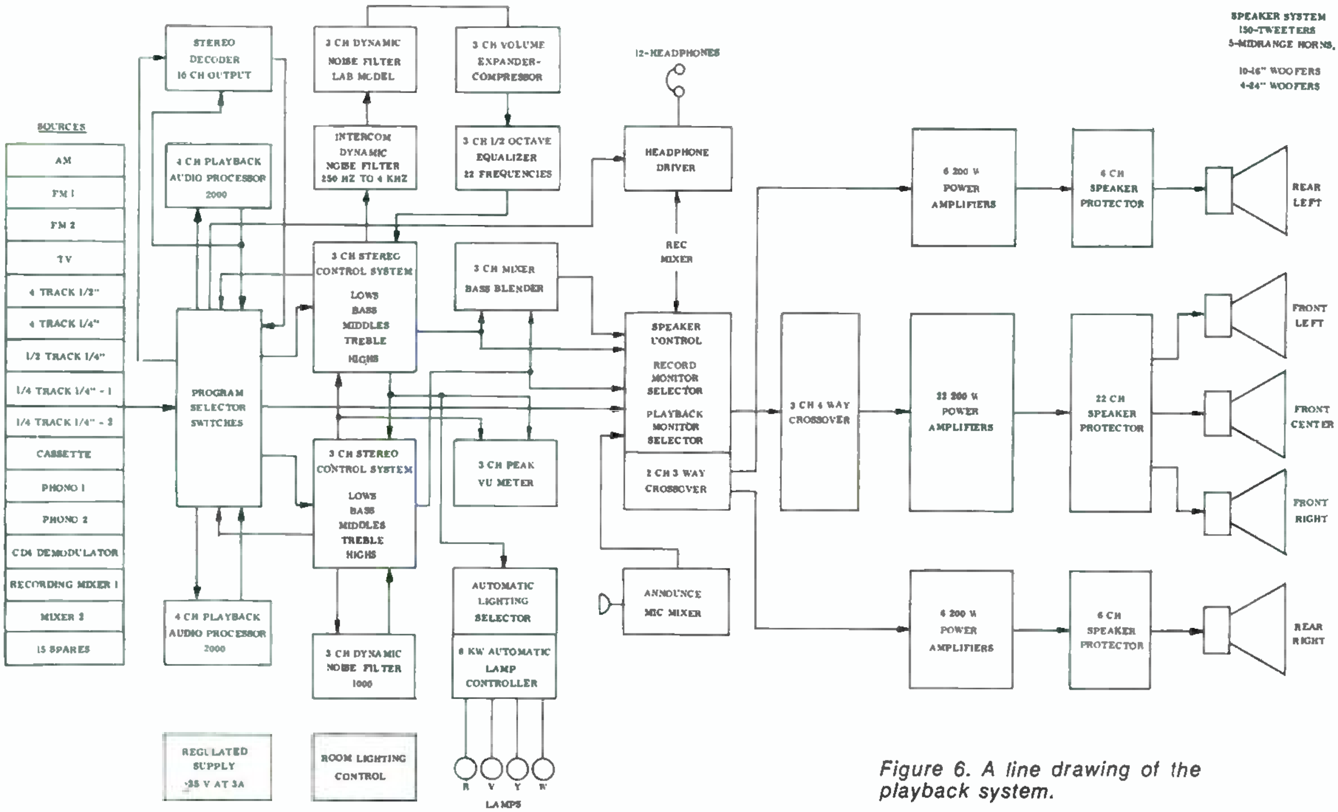


Figure 6. A line drawing of the playback system.

end of the horn permits the horn walls to provide reflected sound well into the high frequencies, and this contributes strongly to the blending of the music. The side horns do not face the listener but direct their sound to the slanted rear walls which reflect the sound to the listening area. Considering the reflection, the acoustic path length for each side horn is about the same as from the front horns to the listener. Therefore, no electronic time delay is needed when reproducing two or three channels through all five horns.

In addition to housing the speakers, the left side horn contains an electronic theatre organ console which plays and records directly through the electronic system, and the right side horn contains a 10-ft. grand piano.

The speaker systems are driven from a total of 34 200-watt amplifier channels utilizing an electronic crossover system. Crossover frequencies are 50, 400, and 6,000 Hz. Besides high- and low-pass filtering from 6 to 36 db/octave, the crossover network also provides equalization for the speakers. As a result of the electronic crossover, the system is capable of producing a sound level without clipping equivalent to that which would be produced by a single 20,000-watt amplifier.

The reason for all the power is to reproduce a live drum set at its original acoustic level from any one of the five speaker systems. Experiments indicate the system has 3 to 6 dB to spare before clipping. Most of the peak power goes into the tweeters which are driven by three 200-watt channels having a power gain at 20 kHz, 15 dB above the signal driving the midrange horn. Acoustical measurements using one-third octave noise bands and summing microphones indicate overall systems response flat within 2 dB from 16 Hz to 16 kHz. Fine adjustment of the system equalization was made in direct A-B comparison against the live drum set.

To prevent a 169-speaker disaster resulting from an inadvertent overload, the entire speaker system is designed to handle the maximum peaks at the clipping levels of the amplifiers. An elaborate protection circuit computes voice coil temperatures and disconnects the speakers from the amplifiers before reaching the point of damage. In addition, the cone excursion of the 16-in. woofers is limited electronically at 0.75 in. peak to peak and at 1 in. for the 24-in. woofers.

LISTENING EQUIPMENT

Specially designed and constructed, all the signal processing and automatic lighting equipment is built into 3½ ft. x 7 ft. high racks as shown in Figure 3 and 4. Signal amplifying functions are all performed by operational amplifiers using both modular and integrated circuit types. The development of several of these operational amplifier modules for this system resulted in the formation of Analog Devices, Inc. Similarly, the solutions to the systems' noise problems brought about the Burwen Laboratories Dynamic Noise Filter and the Audio Processor.

Signal flow through the playback system is shown in Figure 6. Sources for front and rear channels are selected by means of a set of 24-position three-channel selector switches. Although the signal sources utilize some purchased equipment, all of it has been at least partially redesigned. These sources include four channels on ½-in. tape, four channels on ¼-in. tape, two channels on ¼-in. tape, two quarter-track stereo machines, f-m, a-m and short-wave, cassette, tv sound, and electronic organ.

Most important in maximizing the entertainment value is the overall acoustic frequency response of the system and its fine adjustment for each individual signal source. As mentioned earlier, compensation for speaker response is part of the active crossover system. Equalization for stereo signal sources is provided by two separate sets of three-gang-tone controls for the front and rear channels,

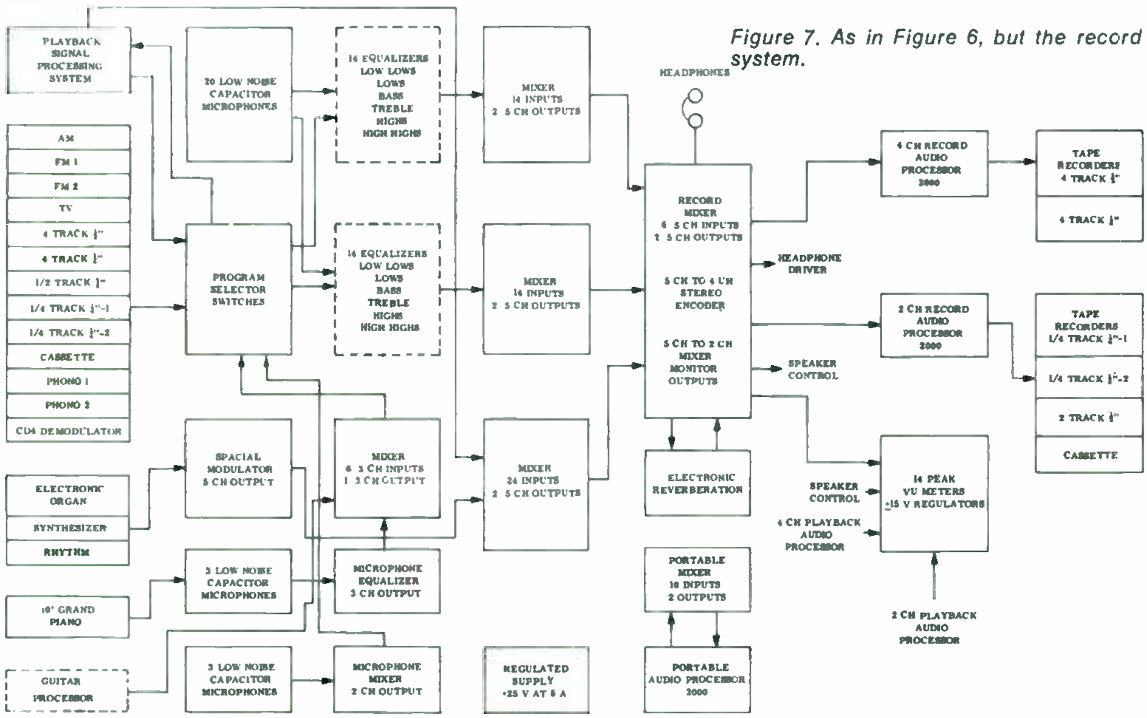


Figure 7. As in Figure 6, but the record system.

each operating on the lows, bass, middles, treble, and highs. In addition, 22 more three-gang controls operate at half octaves from 16 cps to 23 kc.

Disc record signals are generally processed through a stereo decoder which has 150 screwdriver and digital potentiometers on the front panel. This unit produces a five-channel output having any desired amplitude and phase from any of the matrix sources available as well as mono and stereo records. The signals are cleaned up using a Burwen Dynamic Noise Filter and its developmental predecessor. Occasionally, the dynamic range is expanded using a three-channel-wide, dynamic range volume expander-compressor.

RECORDING SYSTEM

Although this system has been used to make Burwen Laboratories' records, my main objective for the recording capability of this system is to make single generation tapes to be reproduced over the five speaker systems in this room. Mixing is, therefore, done live and, since there is no separate control room, monitoring is via headphones and peak vu meters.

The liveness of the room and the lack of acoustic isolation between instruments makes it necessary to operate all microphone channels at nearly the same gains. Recording convenience and efficiency have thus been sacrificed for fidelity and for the ability to immediately play back a recording in the acoustic environment intended for listening. Nevertheless, it is possible to make well-balanced recordings and to achieve upon playback through the entire system more than a 100-dB dynamic range.

As noted earlier, the system is designed for five-channel stereo, although my largest machines have only four tracks. Built into the recording mixer is a matrix encoder which makes four channels out of five with crosstalk between channels down 12 dB. The stereo decoder in the playback system for records also includes a five-channel decoder. Decoding is extremely simple since the four corner channels are the same as the tape signal while the front center channel is derived from addition and subtraction of these four signals. At this writing, the system records and decodes successfully but has not yet been tested with live musical instruments.

The 70-input recording mixer uses about 250 Burwen UM201 Universal Mixing Amplifier modules. Each channel provides master gain plus five more potentiometers for directing any microphone signal to any of the five speaker horns. The microphones can be turned on and off in groups and monitored individually as well as in various combinations. Twenty of the capacitor microphones used are of special design and deliver a 20-dBm line-level output for sound pressure inputs switchable to 140, 125, or 115 dB. Mixer noise, when using 12 or so microphones, is slightly below the microphone noise which amounts to 15 dB SPL A-weighted for each microphone. Seven other capacitor microphones have had their vacuum tubes replaced by transformerless f.e.t. amplifiers.

At the heart of the tape-recording system is a set of five Burwen Model 2000 Audio Processors (formerly Noise Eliminators) which make it possible to record and play back the 105-dB dynamic range of the mixer and microphones. In addition, I am equipped with a portable Model 2000 and a ten-input to two-output mixer for remote uses.

STILL UNDER CONSTRUCTION

At this point, the basic recording and playback functions have been completed. Still under construction is a set of 28 equalizer channels involving six tone controls each to be used on the various microphones during live recording. Also under construction is an analog-type electronic reverberation unit involving no springs, moving parts, or digital sampling. The electronic organ is being coupled to the sound system using an electronic modulation device to replace the rotating speakers in the organ console. To date, the system has taken about ten man years to design, construct, and test. Although the number of knobs and switches is approaching 1,100, playback of program material which has been completely equalized and processed for five-channel reproduction can be accomplished with one knob control. ■

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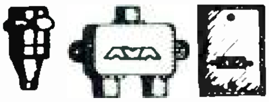

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MULVIHILL

● **Matthew J. Mulvihill** has been appointed vice president and general sales manager of **North American Philips Electronic Component Corporation.** Up to the time of his new appointment, Mr. Mulvihill was the company's western regional sales manager. Mr. Mulvihill succeeds **Allan L. Merken**, who recently moved to North American Philips Lighting Corporation as vice president, marketing.

● Several new executive appointments have been made at **Audio Magnetics Corp.**, Gardena, California. **John J. Kane** has been elected to the office of president and chief operating officer. **Peter Hughes** joins Audio Magnetics, coming from **Admiral International Corp.**, as senior vice president. **Mort Jacobson** has been promoted to senior vice president, engineering. **William De Mucci** has been named senior vice president, manufacturing. **Dale Humphries** has been appointed general manager, video project. **Stewart Scholsberg** has been promoted to vice president, sales, industrial products. **Dan Fine** has been named vice president, sales, consumer and a/v-educational products. Mr. Kane and a New York investment banking firm, **Gibbons, Green & Rice**, with a group of private investors, recently acquired Audio Magnetics from **Mattel, Inc.**

● **Gerald Landau,** formerly vice president, marketing, for **Acoustic Research** of Norwood, Massachusetts, has formed his own marketing consultant firm, **Hi Fi Marketing Consultants,** located at 65 Turning Mill Rd., Lexington, Mass. 02173. Mr. Landau will continue to advise Acoustic Research on its marketing and advertising programs. **Jason Farrow** has recently been appointed director of public relations at Acoustic Research. **Peter Dyke** continues as national sales manager for the company.

● The ninth **MIDEM,** the International Record and Music Publishing Market, will take place in Cannes, France at the Palais des Festivals, from January 18-24, 1975. In addition to the customary exhibitions, entertainment is being planned, to be held in a 5,000 seat tent set up on an esplanade near Palm Beach. Information may be obtained from: Midem Information, 3 rue Garnier, 92200 Neuilly, France. Tel. 747-84 00.

● **Leon A. Wortman,** author of **Closed Circuit Television Handbook,** has been named manager, distributor product sales, for the audio-video systems division of the **Ampex Corporation,** of Redwood City, California. Mr. Wortman is western vice president of the **Audio Engineering Society** and a member of the board of governors.

● **William R. Krehbiel** has been appointed vice president and general manager of the **Scully/Metrotech** division at Mountain View, California of the **Dictaphone Corporation.** Mr. Krehbiel was most recently executive vice president of **Bactomactic, Inc.** Earlier, he had been with the **General Electric Company.**

● U. S. marketing of a number of products, most notably Audax speakers from France will be handled by the **Neosonic Corporation of America,** of Westbury, N.Y. Principals in the new company are **Joseph Longin** and **Joseph N. Benjamin. Jules Rubin** will serve as sales consultant for the Metropolitan area. Sales representatives covering most major U.S. areas have also been appointed.

CUNHA

● **Capitol Magnetic Products** is the new operating title for **Audio Devices, Inc.**, a division of **Capitol Records,** of Los Angeles. The division is responsible for the manufacture, distribution and sale of open reel, cassette, and 8-track cartridge tapes. **Anthony P. Cunha** has recently been elected executive vice president and chief operating officer for Capitol Magnetic Products.

● **Altec Sound Products Division** of Anaheim, California, has relocated its eastern warehouse in Elizabeth, New Jersey. The new warehouse is conveniently situated near Newark airport, facilitating shipments by air freight. The company has also recently enlarged its manufacturing facilities.

● In order to intensify CD-4 promotion in cooperation with **JMF Associates,** the president of the cutting center of **JVC America** of Maspeth, N.Y., **Katsuya (Vic) Goh,** has relocated his office to Hollywood. His new address is: **JVC Cutting Center, Inc.**, 6363 Sunset Blvd., Hollywood, California 90028. Assistant engineer **Gene Yamamoto** will remain at the present New York office.

● The manufacturers of the **FRAP** transducer have set up a new laboratory at 759 Harrison St., San Francisco. Their mailing address, P.O. Box 40097, San Francisco 94190 remains the same. Other actvities at FRAP include a guitar clinic and a paper given by **Arnie Lazarus** for the **Audio Engineering Society.**


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