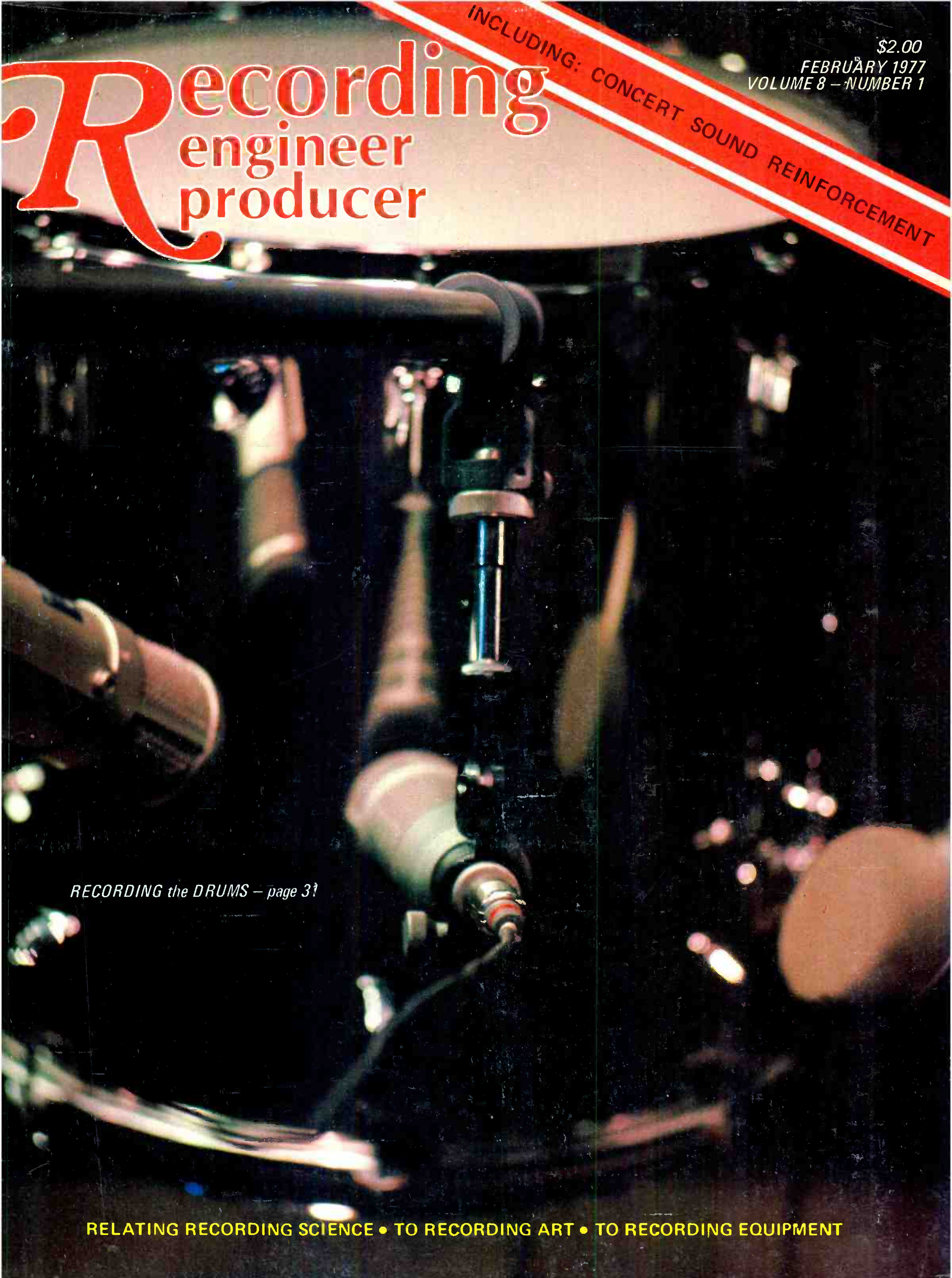

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RECORDING the DRUMS – page 31

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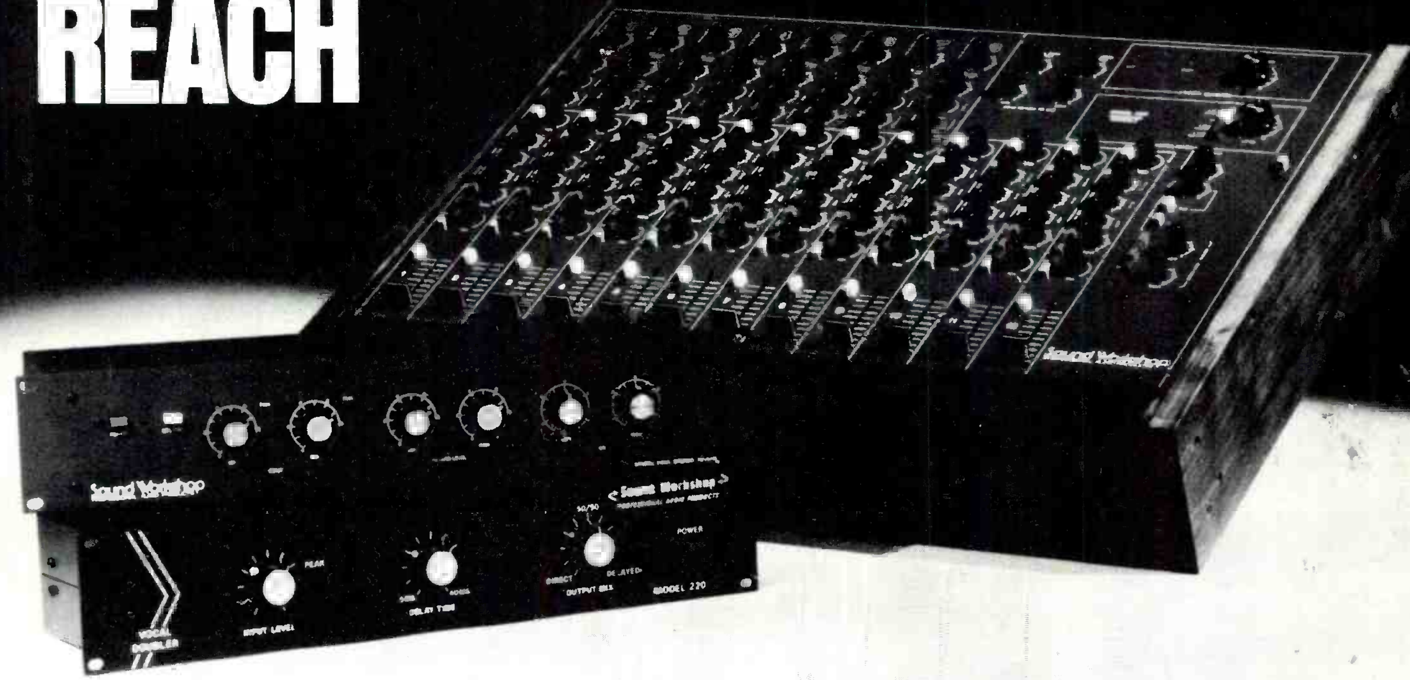

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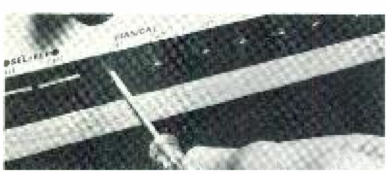

Bias can be re-optimized in seconds.

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Performance Features: Headroom is 19 dBm, a full 15 dBm over the switch selectable fixed output of +4 dBm. This standard reference level output can be rear panel switched to −10 dBm to drive a PA system or power amplifier. S/N ratio is NAB weighted 69 dB full track, 68 dB half track, and 65 dB quarter track. Crosstalk is greater than 60 dB half track. Outputs are 600 ohm balanced (standard on half track) or unbalanced. Line input and output connectors are XLR.

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See your nearest Otari dealer for the full story or contact Otari. And, if it's multichannel you need, ask about the standard-setting four and eight channel versions of the MX-5050.

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acoustics. Robert Schulein of Shure Brothers will discuss Microphone Use, and Juergen Wahl of U.R.E.I. will review Sound System Electronics, Don Keele, Jr., of Klipsch Associates will end the morning session with an Overview of Loudspeaker Systems.

Sound Reinforcement System Applications is the subject of the afternoon sessions. Presentations will be made by Stan Miller of Stanal Sound, who will speak on Sound Systems for the Performing Artist; Mahlon Burkhard of Industrial Research Products, Inc., whose subject will be Creating the Concert Hall Outdoors With Sound Delay; and David Klepper of Klepper, Marshall, King who will speak on Sound Reinforcement in the Theater.

Don Davis of Synergetic Audio Concepts will discuss the relationships among various industry service segments that work to create quality sound reinforcement and what they can learn to improve the satisfaction of the system user. After his presentation, Davis will moderate an audience participation panel discussion on Future Aspects of Sound System Design. Featured speakers at the conference will serve as panel members.

Further information on attendance at the conference or participation as an exhibitor may be obtained from Harry O. Saunders, 225 West Randolf 24A, Chicago, IL 60606 – (312) 727-4331.

CLEVELAND ORCHESTRA RECORDS FIRST SYMPHONIC DIRECT-TO-DISC RECODING OF MODERN TIMES

Kenneth Haas, General Manager of the Cleveland Orchestra, and Jack Renner of Cleveland's Advent Recording Corporation (not related to the Advent of Massachusetts), have announced that The Cleveland Orchestra, Lorin Maazel conducting, has made a phonograph record using the direct-to-disc recording technique during sessions on January 16 and 17 in Cleveland's Masonic Auditorium. Robert Woods produced the recording for Advent, which will release the record on its TELARC label. The repertoire includes works by Berlioz, Bizet, Falla and Tchaikovsky.

This is the first commercial recording made by a symphony orchestra using dir-

nductor Maazel in the foreground, with pro-er Jack Renner at the Auditronics console. he operator Dave Ellsworth in the rear.

For additional information circle No. 7

In session at the Masonic Auditorium . . .

ect-to-disc techniques since the development of the long-playing record. Producer Woods commented: "The Cleveland Orchestra and Lorin Maazel were invited to become involved in this monumentally difficult project of recording unedited performances of complete recorded sides, recorded without pause and cut directly onto the lacquer disc without the intermediate use of a tape generation, because of their proven ability to perform efficiently and effectively under the extreme pressures involved in this kind of recording. This method of recording is much the same as that used in the early days (before tape), and greatly increases the risk factors and requires optimum conditions during every facet of the project."

REPORT ON "NEW CREATIVE AUDIO EFFECTS FOR THE LIVING THEATRE" SYMPOSIUM

On January 22nd and 29th, California State University at Northridge (CSUN) and Cetec Audio conducted a two day symposium on "New Creative Audio Effects for the Living Theatre." The event was hosted by Dr. William Bellman of CSUN, and Mr. Bob Slutsky of Cetec.

The symposium was conceived to develop a framework for sound in the theatre, and then on that basis, present an example: a scene from Ionesco's *Rhinoceros* (which will be presented in full starting February 25th, 1977 at CSUN).

In addition to the high quality of the presentations, the symposium was exceptional for at least two reasons. First, while the idea of theatre sound immediately suggests theatre sound *reinforcement*, this symposium primarily concerned theatre *sound effects* and the implications of new electronics technology on effects systems. Second, instead of presenting a theatre sound "clinic" with a few speakers and an audience, CSUN and Cetec held a true *symposium*, where *everyone* participated. There were a large number of creative suggestions from the "*audience*" concerning possible ways to overcome the challenges of producing the sound effects called for by Ionesco in his *Rhinoceros* script.

The play *Rhinoceros* takes place in a small town in France. The plot concerns the sudden appearance of several rhinocers' (rhinoceri?) in the town, and the eventual (literal) transformation of everyone in the town into a rhinoceros with the sole exception of one man. While in the play, the transformation is literal, the rhinoceros' are supposed to represent Nazi-like conformity. Part of Ionesco's purpose in writing the play was to demonstrate that the potential for this rhinoceros-like conformity exists in each member of the audience as well as the cast.

In order to help build the mood for the play, Ionesco wrote liberal sound effects cues into the script, calling for "tramplings", "snufflings", "trumpetings" and even "musical trumpetings" from the rhinoceros'. Significantly, there is no mention in the script of methods for producing these effects, and few guidelines as to the extent that the effects should be used. The symposium then, became a search for effects that an audience would accept, and that could be produced with the avilable equipment.

This last limitation turned out to be less a limitation than a matter of "overchoice". A Cetec Series 20A console with their "Pre-set Distribution System" controlled the system. There were eight output channels each with a separate power amplifier and speaker system, and the speakers were set up in all corners of the theatre, as well as back stage and in the orchestra pit, to perform multi-channel effects. A Vega wireless microphone system was used on the stage, and some of the effects were recorded on a Scully 280B-2 two-track recorder with variable speed electronics. In addition, there were a whole array of special effects devices including a MICMIX Time Warp, two model 102 Lexicon and one 9430A Altec digital time delays, a Lexicon Varispeech 27 (able to vary either the pitch or the speed of a program independently), a Putney VCS3 analog synthesizer, a 527A and a 529A UREI equalizer, and an Orban/Parasound parametric equalizer.

Not all of the equipment will be available for the final presentation of the play. At the symposium, however the equipment provided the means for a number of startling distortions of an actor's voice (as the actor turned into a rhinoceros on stage), as well as excellent "background" effects that were entirely electronic, and some interesting demonstrations of the ability of digital delay to "expand" the apparent size of a room, or to "localize" certain sounds without the need for panning techniques.

While this was probably not the first time that some of the new electronic technology has been applied to theatre sound effects generation, it was certainly an eye-opening experience, and the implications to the world of theatre and to the sound world are significant.

Sound effects have a long history in films, where they can be recorded and re-recorded in the studio with alterations and timing changes as needed. In live theatre, however, sound effects have been traditionally produced by as simple means as possible (often mechanical methods) for two main reasons. First, sound systems, for effects (or for reinforcement) have traditionally not been as reliable as they might have been, and are considered "untrustworthy" by many theatre personnel. Second, until the recent development of digital audio devices and synthesizers, most sound system effects were produced mechanically and recorded on a tape machine for playback at the appropriate cue. This works fine for simple, short-timed effects, but for effects that occur over several minutes or more, a tape machine forces an actor to conform to its time schedule, a very undesirable limitation in the flexible world of theatre.

Now however, new equipment and techniques can overcome these problems, making the sound effects system a viable addition to live theatre. Sound effects can be used to create illusions, to enhance the emotional effect of a scene, or to manipulate audience attention. In some cases, the sound may even be an actor unto itself. The power this can give to a director is obvious.

The implications for the sound contractor or engineering consultant should also be obvious. Potentially, a sound effects system can be as important to a theatre as a stage lighting system, even in a theatre too small to require a traditional sound reinforcement system. And because the potential exists, it's only a matter of time before such systems may become commonplace. The CSUN/Cetec symposium was worthwhile because it not only was a good introduction (or review) to theatre sound in general, but that it also helped create an awareness of this "other side" of theatre sound, and its probable future development.

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market comment:

The Resurrection of BOP and BRAHMS

by Kim McKenzie and Carl Plumgarden

Once upon a time there were two tremendously popular forms of music in America: Classical and Jazz. Although the two forms never really met eye-to-eye with the audience, they met eye-to-eye on the sales reports. During the late 40's and early 50's when the LP was in its infancy the number of small labels springing up to service this need was something short of an epidemic.

But then something happened. The audience became fickle, the prices went

"BOP and BRAHMS" continued —

up, the artists were being lured away by fatter contracts from the larger labels, and for a period of time from the late 60's until just recently the two forms seemed to have all but vanished from the American music spectrum. The *American* music spectrum, but not so everywhere else. In Europe the tradition is still alive and flourishing.

Although the cause for the two mediums to succumb to such passive oblivion is relatively different for different reasons, the fact that they are beginning to get back some of their notoriety after neglect for many years is important. Although a handful of people are doing something about it, the audience is slowly coming back to what was always there.

For the classical market, the dismal periods came when such former giants as RCA and London, trying to tune in to a progressive rock audience created ludicrous packages serving only to scare people away with their poor packaging, poor pressing, and poor concept.

The Jazz market, on the other hand, became so bad here in the U.S. that it literally forced many musicians to take sanctuary in Europe, where their craft was appreciated and preserved.

Of course, one must realize that the Jazz scene *does* still flourish here in the U.S., but is somehow subdued when compared with the goings-on in Europe.

With the new emphasis on the disco market, however, many jazz musicians have found an easy escape hatch from relative obscurity and found acceptance in the new form. Whether or not this is valid is highly disputed amongst record company execs and musicians alike.

One of the healthier aspects to the whole thing has been the appearance of various labels deeply concerned with quality product and quality acts. Since its inception in the late 60's, ECM Records, in Germany, has shown that consistent quality has many benefits. Its stateside appearance, through Polydor, has substantiated it. But meanwhile, there appears to be a trend beginning in America towards this same concept; giving the selective, quality conscious music buyer what they want. Pablo Records, begun in the early

Miller listening to a play-back with Jazz All-Star ROY ELDRIDGE

70's by Norman Granz with Eric Miller, have had stellar careers in the past in the Jazz field, and their contribution to music and the selective buyer has been far reaching. With beginning releases of Jazz greats of the 50's, many of which were previously unissued, Pablo made an immediate impact on what was once a sagging industry. During a recent interview Miller reflected on the success of Pablo: "Timing is everything. We came about at a time when it was sorely needed. We gave and are continuing to give the audience something they really want. It's as simple as that. And it really is a simple thing; but for some reason maintaining simplicity on the large-scale corporate level has always been difficult."

An interesting sidelight: Pablo, like ECM, is distributed throughout Europe via Polydor; but distributed in the U.S. by RCA, making for a rather interesting set of bedfellows.

It is important to point out that with the resurgence of the independent label, taking on the responsibilities of issuing quality product and nurturing acts, a sense of freedom for the artist is established. Knowing there is an audience who will accept and enjoy what a particular musician is doing, he can explore many more varied fields, sometimes at the expense of the exec's nervous system. "I was over at RCA's studios a while back with Dizzy Gillespie", Miller points out, "and he was doing a disco session. I couldn't believe it. Here is this guy with all this history behind him, one of the true greats of jazz, and he's in there doing this *disco session!* I was horrified at first, but that's what he wanted to do, so I got out of there and just sat in a bar down the street for the rest of the day."

The need for creative freedom on the part of the musician has been a struggle for as long as anyone can remember. And achieving that has opened up a whole new burgeoning scene for the musician, the record company, and the audience to work from.

On the classical side, the success of labels such as Pablo are based on entirely different circumstances. Most production is still being undertaken in Europe, simply because there are more halls in which to record, musicians are considerably cheaper to hire, and one very touchy, but important point, the unions are easier to get along with. Although I have not been able to substantiate this rumor, I have heard that if a certain number of American musicians play as soloists, or in a chamber capacity for a recording, and should their orchestra or union know this is going on, the company responsible for the recording must pay for the use of the entire orchestra although barely any of them are used. As I said before, this has only been a rumor circulated through a number of sources. So with those problems besetting the classical label, what is the solution? A number of

. . . listening to a take with the incomparable Dizzy Gilesp

smaller labels, such as Nonesuch, which is referred to as the biggest-smallest label in the U.S., simply issues a large quantity of small works, electronic works, or pieces leased through larger European labels which have not found release over here. Other labels have taken on the responsibility of issuing historic recordings, previously issued on either 78's or through radio transcriptions of the 30's and 40's, and most have met with a certain degree of praise, raised eyebrows or, as in the case of a certain release by Turnabout/Vox, a court injunction by RCA.

So it is still left up to the larger labels, with larger budgets, to bring out new items for the classical market. One recent revelation was the go-ahead for Westminster/ABC to begin domestic production of classical recordings. The other was the debut of Erato, through RCA, and Seon, through ABC, as well as the bolstering of lesser-priced lines such as Angel/Seraphim and CBS/Odyssey. These trends have made it possible for the neophite classical listener to purchase and experiment in his purchases, while building up a quality library. And this in turn has created, at last report, a *real* market.

But back to Jazz and Pablo and Eric Miller. What is the level of success an independent label such as Pablo is having? A consistent number of Grammy Award nominations and awards, a stepped up endeavor to bring the music to the audience who wants to hear it, and some nuances as well. Recently, Pablo, through RCA, issued a first-anywhere recording of Gershwin's *Porgy & Bess*, exquisitely done by Cleo Laine, Ray Charles, and conducted by Frank DeVol. A lot of avant-garde Jazz affecianados will undoubtedly scoff at this as not being real Jazz, but the real point to be made of all this is: it was done, it was an idea that needed care and patience, it was masterfully executed; and above all, it worked.

And because it works nobody jokes about independent labels anymore, and nobody jokes about Jazz or Classical music being dead.

The music industry is looking twice and seeing what they've missed all along.

Late News continues on . . . Page 82 . . .

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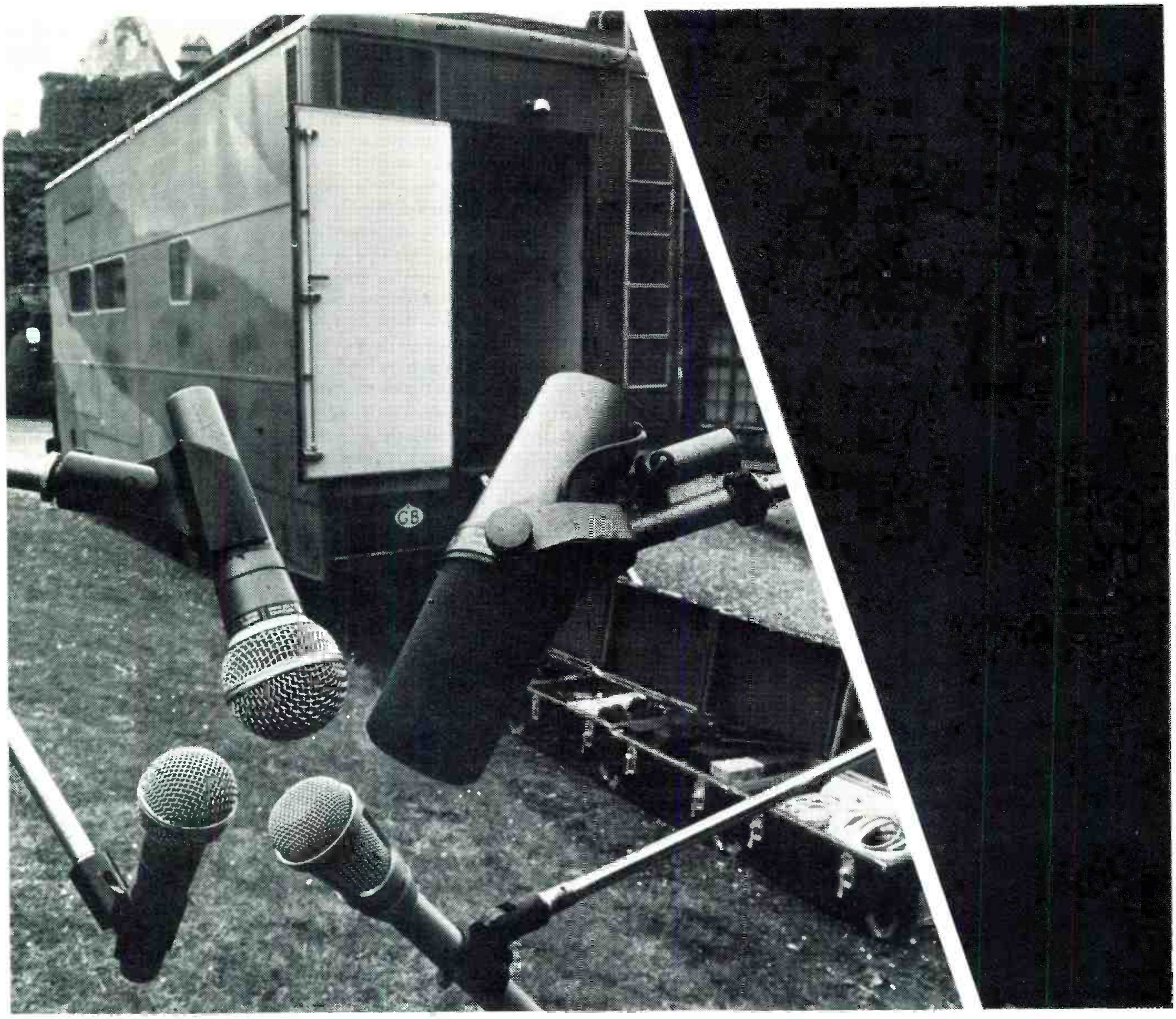

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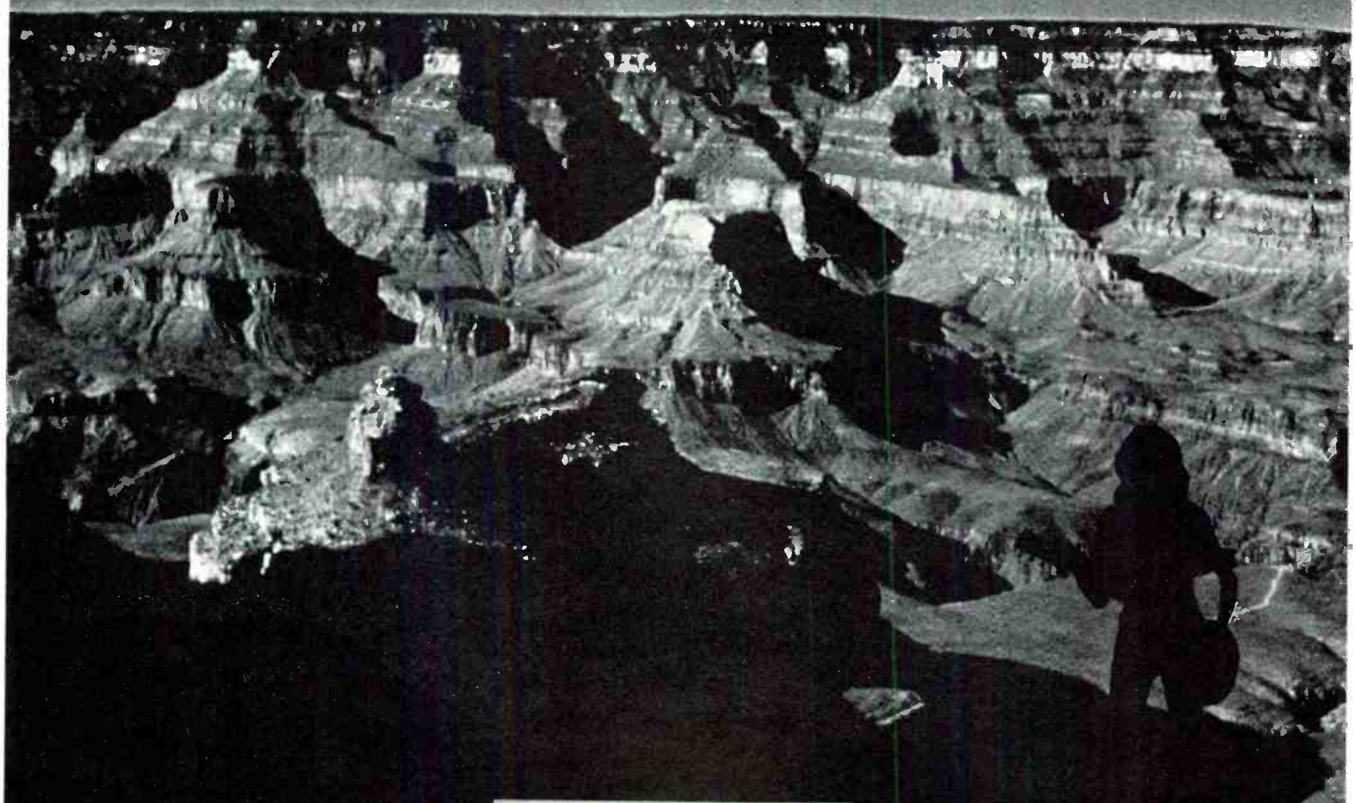

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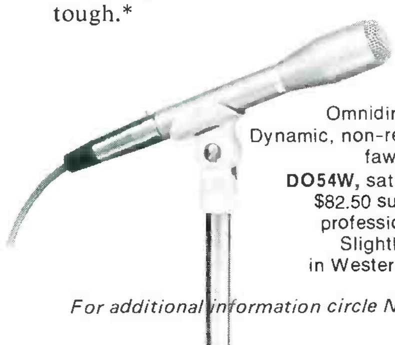

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interfacing the portable sound system with the permanent sound system

by

Chris Foreman

Today's touring concert reinforcement companies probably wouldn't exist if it weren't for the fact that few permanent systems, whether they are in auditoriums, theaters, clubs or elsewhere, are ill equipped to reinforce popular music. Many permanent (house) systems were designed for speech reinforcement, not for music, and may be underpowered, lack headroom, and have a restricted frequency response typical of paging/background music systems. Often, their coverage is poor, and, sad to say, many house systems are in a state of disrepair.

On the other hand, there are a few house systems, designed by farsighted engineers, and purchased by equally farsighted organizations, that are well planned and versatile enough that a touring sound company may be able to use them exclusive of any portable equipment. More important to the concert sound reinforcement company, however, is the fact that there are *a large number of house systems that are capable of complementing the portable system*, if they are utilized properly. This article explores some of the techniques for making the interface between these systems and the portable system, and in addition, details some ideas for anyone who may be designing or upgrading a permanent system for popular music.

The Portable/Permanent Combination Is Useful:

A. When there are seating areas that the portable system can't easily cover such as the areas under low balconies, the rear of long, low buildings, outdoor areas or any area where the acoustic situation indicates that the additional system could help. An example of this last situation would be an indoor arena or auditorium where the location of the portable system might aggrevate an already bad echo problem. In this case, the portable system might cover the near stage areas, and the house system could then cover the outer areas.

B. When the system is restricted by a failure, or by size, weight, financial or political factors.

One good way to avoid failures and to foil Murphey's Law is to plan ahead for the possibility of catastrophic failures. (i.e., if you want to avoid accidents, pay your insurance premiums). Part of such a plan for the sound company should be the ability to supplement whatever's left of the portable system with the house system.

Other limitations come more often than failures. The height of the portable system's speaker stacks may be limited to avoid blocking seats, or the sound company may have to air freight a small system to a show to meet a deadline. Sometimes, the "powers that be" may require removal of some of those "ugly" portable speakers to avoid an esthetic conflict with the stage set.

Incidentally, one very good way to solve a lot of these problems is to have a system that can be "flown" (a portable hanging system). Unfortunately, there are many places where even this solution won't work, because of weight restrictions on the building support girders, or because something else is already hanging in the only usable spot (a scoreboard, or the house speaker cluster!). In these cases, the house system may be able to provide that extra, needed reinforcement.

C. There may be times when the house system is good enough that the portable/permanent combination is desireable even though it's not needed (in a strict sense), to reinforce the existing portable setup.

Systems like this do exist, and the touring company should be ready to take advantage of them. For example, the Indianapolis Convention Center has a unique distributed system* using 192 Altec Voice of the Theatre speaker systems which can be arranged electronically using a specially designed time delay system for any stage location. This system is used for many different types of shows. For some shows, a pair of small stage speaker stacks are added for psychoacoustic effect (as discussed later).

Another example of an excellent permanent system is the system at the New Grand Old Opry in Nashville (Re/p October, 1974). It was conceived, designed and built for popular music.

It's useful to point out that few, if any, of even these high quality permanent systems are capable of reinforcing "hard core" or "acid" rock and roll music. Since most rock sound companies will not be interested in using a house system, and few houses will permit it, these combinations are not considered in this article.

Not all good systems are located in large auditoriums or theaters, however. Many of the systems in resort area clubs have been continually upgraded, and are quite usable. Many of the artists that play these clubs carry only their musical instruments, and a minimum amount of portable equipment.

There are probably enough other systems worthy of mention to fill several pages. The point is, that the sound com-

---

** A "distributed" system is characterized by a large number of full-range speaker systems located in the ceiling, which point down towards the seating area. A "cluster" system is characterized by a group of high frequency horns and low frequency woofers all in the same place (or in discrete groups), aimed towards the various portions of the seating area.*

pany, should be ready to use a good house system, and the house can benefit from the realization that the house system may be constructively used to improve the sound quality of a performance.

The Electronic Interface

In order to make efficient use of a house system, in conjunction with a portable system, there must be a fast, workable interface between the two. Usually the portable system is used as the "main" system, and the house system receives a single input directly from the portable system mix. When there is a choice of inputs to the house system, bypass as music of the system as possible. In other words, if there is a choice between a mic and line level input, choose the line input (to bypass the mic preamps). By bypassing as much as possible of the house system, noise and distortion are reduced.

Both the sound company and the house technician need a good assortment of audio adapters and patch cables, in-line pads, and transformers. It's a good idea to have a couple of "ground lift adapters" and "phase inverter adapters" made from a Switchcraft S3FM as shown in Figure 1, A&B. Another handy device is a Shure M63 Audio Master coupled

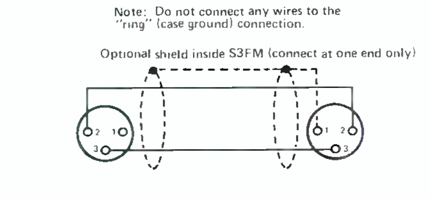


FIGURE 1A

GROUND LIFT ADAPTER MADE FROM SWITCHCRAFT S3FM

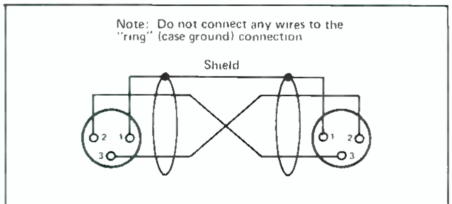
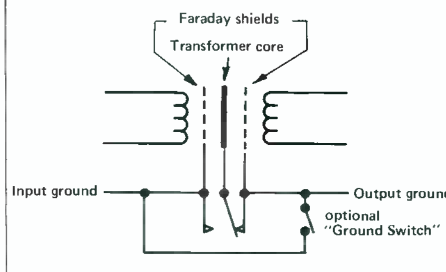


FIGURE 1B

PHASE INVERTER ADAPTER MADE FROM SWITCHCRAFT S3FM

with a Shure A68M "side car" mic pre-amp. This combination will accept almost any kind of input and convert it to almost any kind of output while including an ample amount of gain, a VU meter, tone controls and variable high and low pass filters.

Making connections between two systems will often result in noticable grounding noises -- hums, buzzes, and so on. Sometimes the "ground lift adapter" shown in Figure 1-A will break the offending ground loop on a balanced line. Other times, the capacitive coupling between the two grounds will cause some noise even when the ground is interrupted. In this case, an isolation transformer with a faraday shield should help solve the problem (Figure 2).



NOTES:
Mount the transformer in a metal box, connected to the output ground.

Use a 1 : 1 ratio transformer of the appropriate impedance.

Avoid contact between the box and other system grounds (such as amplifier chassis, conduits, etc.).

FIGURE 2
ISOLATION TRANSFORMER

View from the real world: HOW IT WORKS at the TEXAS A&M THEATRE COMPLEX

According to Steve Hodge, Manager of the Theatre Complex at Texas A&M University, "When a touring system is used, we *suggest* that at least part of the house system be used with it. The shows that have followed this suggestion have been quite successful."

The system in the Texas A&M Auditorium is actually two systems, a sound reinforcement system and a reverberation system. The reinforcement system has three bi-amplified channels consisting of a bass box with four 15" speakers and three horns hung in clusters above the stage. The reverberation system consists of a reverb chamber, digital delay and twelve 200-watt power amplifiers driving 74 speaker systems in the ceiling. While the reinforcement system provides direct sound to the audience, the reverberation system enables the sound technician to artificially extend the reverb time of the hall. The reverberation system is also used to lend a "surround" feeling to a performance.

All the equipment is Altec, including two custom-built mixing consoles using Altec components. There is a stage monitor system and the required cue systems. There are 1/3rd octave equalizers on all principal systems. These equalizers are set for the desired system response rather than for maximum gain before feedback, and they remain in the circuit even when the house system is used to complement a portable system. The only system limiter is built-in to the digital delay, but Steve Hodge reports that to date, portable system operators have been realistic in their expectations from the house system so that limiters haven't been necessary.

A house sound technician is present for every use of the hall, whether or not a touring system is being used. The technician monitors the signals coming from the portable system as well as the house system output.

As shown, the system has versatile input capabilities. It can accept a mike level, line level or speaker level signal from a portable system. Each of these inputs is transformer-isolated and includes an attenuator. One signal from the portable system feeds the house reinforcement system. This signal allows the house system to provide better coverage in the rear of the balcony (unused portions of the house clusters can be shut off). If possible, another signal (from the portable mixing console's echo buss) feeds the house reverberation system.

Most recently the Texas A&M Auditorium system has been used in conjunction with a portable system for two touring shows: Sergio Mendez and The Captain and Tenille. On one show a signal was fed to the reverb system to provide the "surround" effect. For the other show, one signal from the portable system fed the house system speakers covering the rear balcony, while another signal fed the house reverberation system. The portable system operator had a good "feel" for the reverberation system from past experience with spring reverberation units, and the results of using the house system was impressive. Both shows were considered highly successful.

Steve Hodge notes: "It is our feeling that our job and the job of the touring sound company is to provide the best possible sound. If this is best done by using parts of both systems (theirs and ours) then every effort should be made to make this possible."

Equalization

Bypassing the house equalizer can result in a very noticable improvement in many permanent systems when they must be used to reinforce music. Many of these systems are equalized for speech reinforcement (and many of them have been severly over-equalized), and this one action can improve frequency response and headroom, while lowering noise and distortion. In addition, the combination of the portable system with the house system may change feedback frequencies, so using the portable system's equalizer allows different equalizer settings, if necessary, without changing the control settings on the house equalizer.

To help protect the house system, avoid excessive boost on the portable equalizer, and use both a high pass and a low pass filter, such as the Shure M63 mentioned earlier, or some other type. Many house systems cannot handle either low bass or high treble frequencies. The high and low pass filter frequencies can usually be set by noticing the settings on the house equalizer (or the house high and low pass filters, if there are any), and by applying good engineering judgement. If the house system is primarily speech oriented yet does have low frequency speakers, use a high pass filter at between 150 and 200 Hz, and a low pass filter at around 12 kHz, both at 12 or 18 dB/octave. If the system does not have any low frequency speakers, it probably should not be used to reinforce the portable system except in an emergency. Even if the system is planned for musical performan-

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Limiting

If possible, the portable system should include an extra limiter, to use as the final device in the chain feeding the house system. If there is a house limiter, either use it, or adjust the portable limiter to provide the house system with the same degree of protection. As a rule of thumb, determine the signal level necessary to just clip (overdrive) the house system, set its input signal (from the portable system) for an average (nominal) level that is at least 10 dB below the house clip level, and adjust the limiter to *prevent* the feed signal from reaching the house clip level.

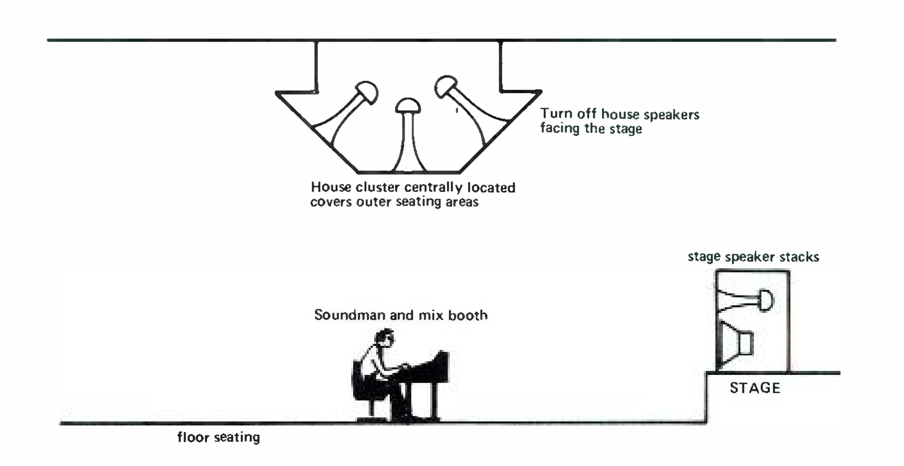
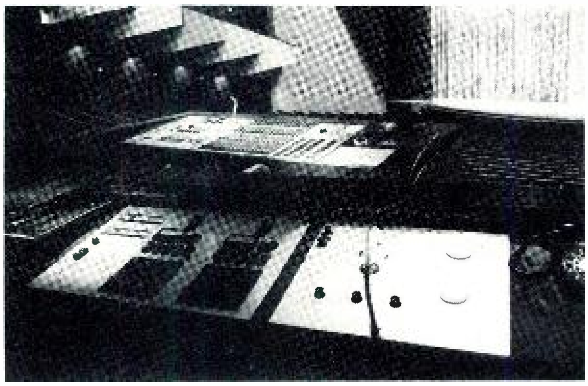
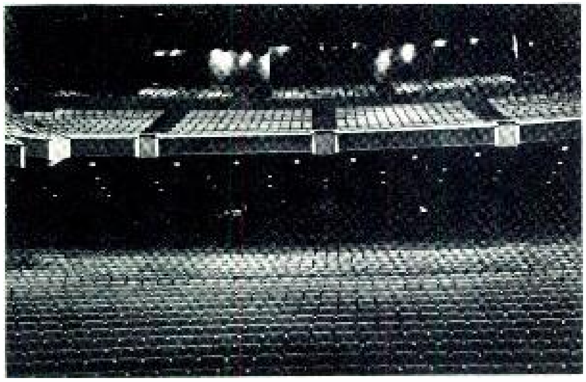


FIGURE 3

The combination of high and low pass filters, careful level setting, and the use of a limter should provide reasonable protection to the house system, while allowing it to perform up to its peak capabilities.


TEXAS A&M Theatre Complex Auditorium. Equipment room and mixing console . . .


Effects and Mixing positions, looking toward the stage . . .


View from the stage. Mixing position barely visible to the left of center, under the balcony . . .

Time Delay

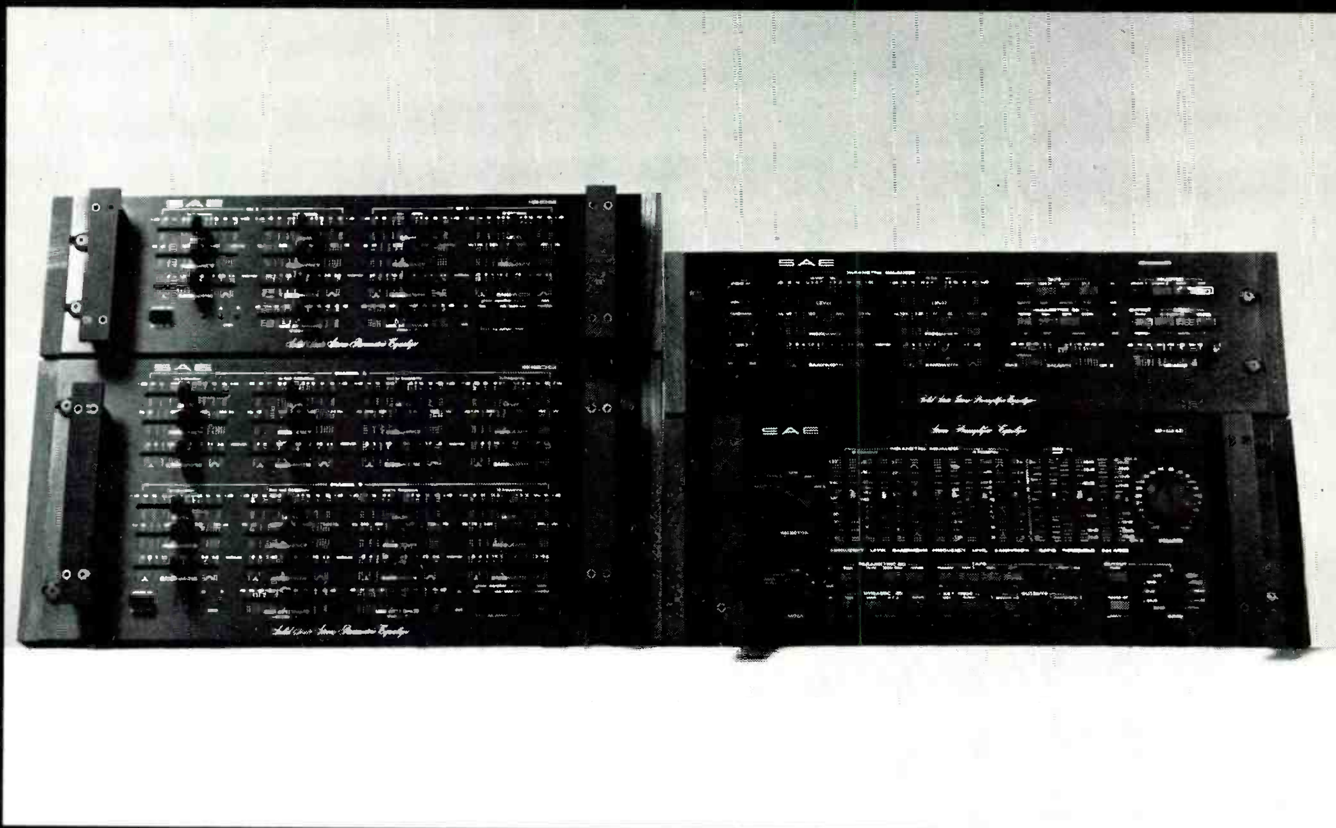
To make effective use of many house systems, the sound company should carry a high quality audio time delay device. For example, the house cluster in a large sports arena may be in the center of the hall, yet the stage may be set at one end. Assuming that the house speakers facing the stage can be turned off (Figure 3), this system may be able to provide valuable extra reinforcement for the portable system, when fed a time delayed signal. Without the time delay, however, the acoustical time delay between the portable stacks and the house cluster will cause an irritating artificial echo.

A similar situation exists when the house system feeds speakers under a low hanging balcony (Figure --). The chances are that there will already be a time delay device incorporated into such a house system, but if not, the sound company can be prepared by carrying their own.

Time delay is useful for psychoacoustic effect as well. For example, if the portable system feeds a house proscenium cluster, sending it a signal delayed by a few milliseconds will help create the illusion that the sound is eminating entirely from the portable system's stage stacks, while there is still the advantage of the added coverage provided by the house cluster (Figure 4). On the other hand, delaying the stage stacks a few milliseconds, creates the illusion that the sound is coming directly from the center of the stage.* Delaying the signal feeding a house distributed system takes advantage of the increased coverage while retaining

** Our ears can easily tell whether a sound comes from the right or the left, however, they cannot as easily discriminate between sounds coming from above or below our line of sight. Therefore, a central cluster system (proscenium cluster) can produce sound that seems to come directly from the center of the stage, even though the speakers are above.*

the SAE Revolution



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the illusion of sound eminating directly from the stage area (Figure 5).

The Program Signal To The House System

One of the most critical decisions to make when using a house system to reinforce a portable system is what program signals to feed the house system. The decision must be based on a number of factors. First, if the house system is being used to provide extra coverage (for example to the back of a long hall), it should receive the signals which will not carry to those areas by themselves. Normally, bass frequencies will travel to the back of a long hall readily because of their longer wavelength (which allows them to diffuse around barriers), and because there is less likelihood that they will be absorbed by acoustic materials (and people) in the house. Therefore, the house system may be primarily used to reinforce mid and high frequency material. This can be accomplished by the judicious use of a high pass filter, or by feeding the house system with a mix that includes only mid and high frequencies (vocals and higher range instruments). Besides avoiding over-reinforcement of bass frequencies, this provides the house system with an extra measure of protection against too much low frequency energy. In addition, many house systems, which are not designed for those low bass notes, will benefit in terms of lower distortion and increased headroom when the low frequencies are reduced.

On the other end of the frequency spectrum, unless the house system has exceptionally good high frequency response, there won't be much gain in coverage or quality by feeding it large doses of super high frequency material (cymbals, vocal sibilants or synthesizers with a high harmonic content). If the house cluster cannot reproduce these highs, they will become heat energy in the voice coils of the speakers, causing increased possibility of distortion or failure. Since it is hard to cut the instruments producing these high frequencies out of the mix feeding the house system, a low pass filter, as recommended before, will do the job.

One very effective method of dividing the signals between the portable stage stacks and a proscenium cluster, a distributed system or a delayed central cluster, is to feed a mix of instruments and vocals to the house system, letting the instruments (those with their own stage amplifiers) predominate slightly over the vocals. Use the opposite blend in the stage stacks. Since the instrument amplifiers themselves will cover the seating area immediately in front of the stage, the stage stacks do not need to reinforce them to any great degree in that area. Since the instrument amplifiers don't carry farther back in the house, the house cluster reinforces them. This mix method provides an even blend of instruments and vocals throughout the entire seating area.

In addition to these uses, if the mix is multi-channel, the house system may be able to provide some good effects capabilities. Use the house under-balcony speakers in conjunction with a rear channel mix, or the house proscenium cluster for a center channel mix.

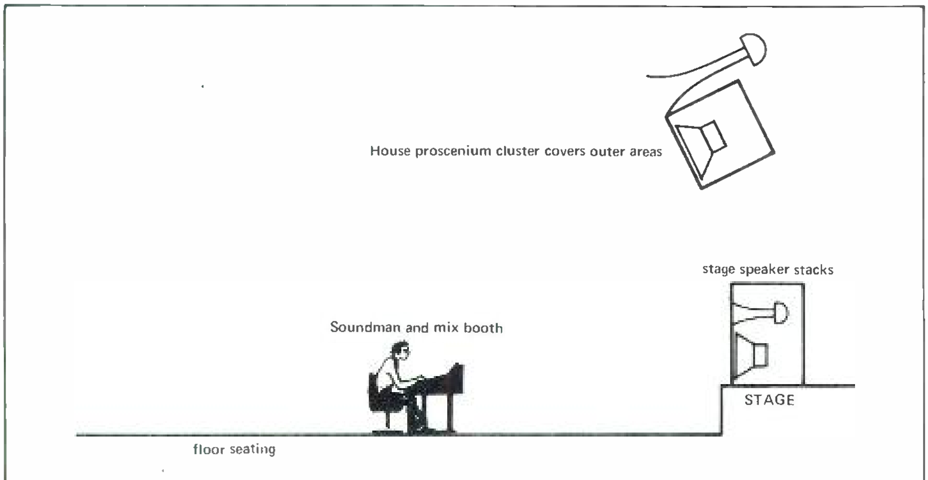


FIGURE 4

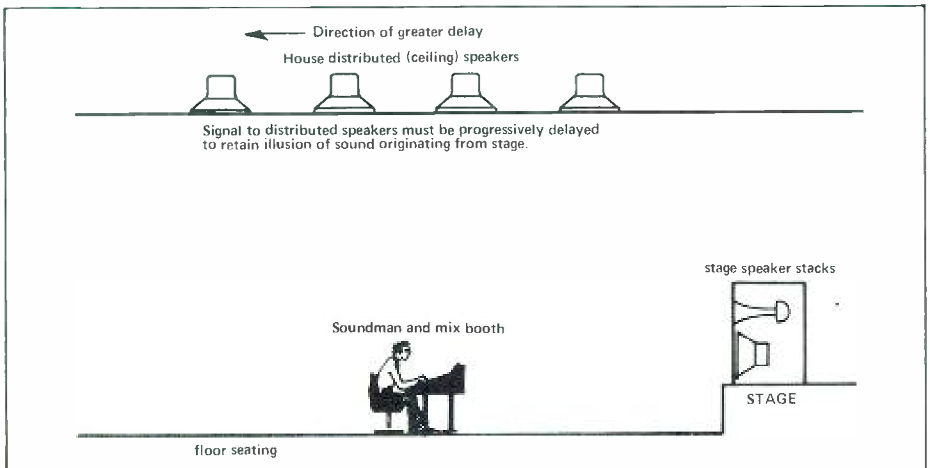


FIGURE 5

Using The Better Systems

Many resort-area clubs, such as those in Las Vegas and Lake Tahoe, and some music halls and theaters have brand new systems, or systems that have been continuously upgraded. These systems can often be used exclusive of any portable equipment. On a small stage, or in an area where large stage stacks would block seats, this is a distinct advantage. In some cases, house rules may even require use of the house system. Performers who regularly play the club circuits in these resort areas know which clubs have good systems, and often they will carry a minimal amount of their own equipment (instruments, mikes, a monitor system and possibly a mixing console).

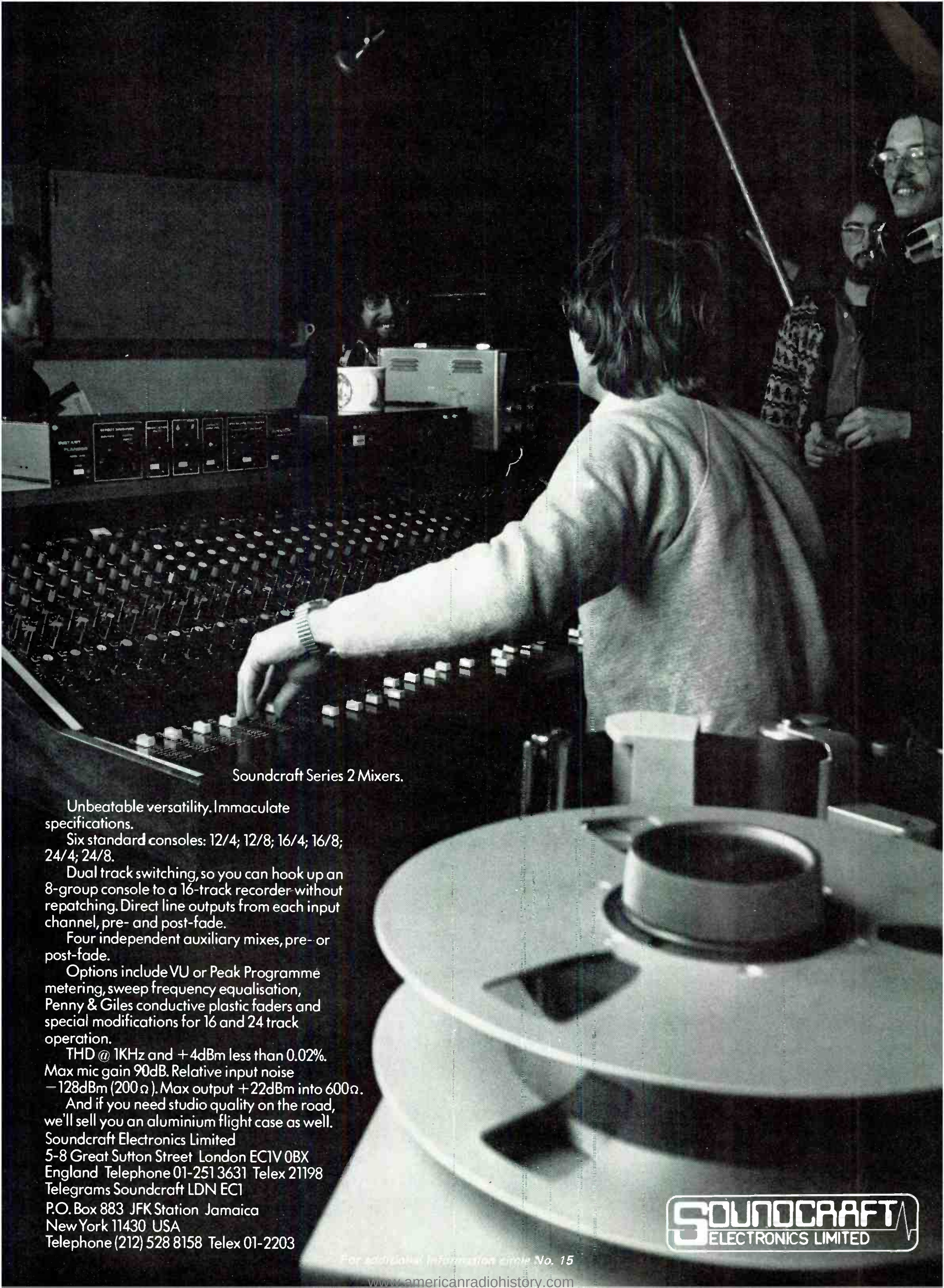
High quality systems are sometimes found at seasonal outdoor theaters or music halls. For example, the New Greek Theater and the Universal Amphitheater, both in the Los Angeles area, rent large, complete portable systems from a concert sound company for their operating season. And, as mentioned before, a few large halls and indoor theaters have well designed systems oriented towards musical performances.

To the sound company the most important things about using one of these systems is to know beforehand what its capabilities are, and what the house rules are concerning its use with portable equipment.

FROM THE OTHER SIDE

The operator, or owner of an existing permanent house system, may understand that it could be used to augment the portable systems that the sound companies bring in, yet still be somewhat concerned for its safety.

The best thing to do to clear up any difficulty is to notify the system designer that there is an interest in allowing the house system to be used in conjunction with a portable system. If the designer feels that the system is capable of being used, at least partially, to reinforce a portable sound system, he or she will probably recommend a number of safety precautions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(which may be similar to those mentioned earlier in this article).

In addition to adequate protection schemes, try to provide a variety of inputs for the signal from the portable system. One ideal input would feed the signal from the portable system through a set of high and low pass filters tailored specifically for the house system, and a limiter set to severely limit incoming signals that exceed a predetermined point (to prevent possible overdriving of the house system). Especially if the house system uses any unconventional connectors, keep a supply of adapters on hand (carefully labeled as to their rightful owner).

If possible (and appropriate), provide a means to switch different segments of the house speaker system so that it can reinforce coverage in selected seating areas, no matter where the stage is located.

The best way to protect the house interests, yet provide this service, is to keep a qualified technician with the house system before, during and after the show. The technician can aid in the electronic interface, and operate the house system as well as answer any questions the sound company may have.

Designing New House Systems

The following ideas are by no means inclusive, but are presented as an overview of some of the concepts that make a permanent house system usable in conjunction with a portable concert system.

General Hints:

A. If possible, hire an engineering consultant who understands popular music and rock and roll sound systems, as well as the traditional acoustical and engineering concepts.

B. Design the system for *musical* usage. Frequency response should be as wide (and flat) as possible. Dynamic range is another important consideration. While it's obviously important to keep hum and noise to a minimum, adequate headroom is also critical. (Headroom is the amount of power the system has left above its average level for musical peaks.) For a musically oriented system, 10 dB is a minimum amount of headroom. If it is economically possible, the system should be able to produce peaks of 100 dB SPL, or higher in the audience area (85 to 90 dB SPL average level).

C. Generally, a speaker cluster system is probably a better choice for musical performances than a distributed system. If there is a choice, a speaker cluster will probably provide higher fidelity performance per dollar spent, and can add to the illusion that the sound emanates directly from the performers. A good distributed system, if properly designed, can be very effective and highly versatile, but may cost considerably more than a cluster system.

D. Know where to stop. Unless the system is planned to singlehandedly cover any perfomance to come, now or in the future, keep in mind that some sound companies won't be interested in using it. Because of its mobility, the sound systems used by many popular entertainers have become a part of their performance, and in many cases, the performance *depends* on certain aspects of their own personal system. The system may be used for effects, or its frequency response and distortion characteristics may have become associated with the entertainer's particular "sound". In other words, the system has become another musical instrument, albeit a large one. The point is, that there is little reason to design and build a system big enough for any conceivable concert unless it will be used for *every* concert; the economics should be clear. The idea is to design a **permanent system that can stand on its own for smaller performances, yet complement a portable system when the performance becomes larger and more complex.**

Microphones and Inputs

When choosing microphones for a new system, concentrate on the rugged professional types which are designed especially for musical use. If there will be a large number of microphones and inputs choose a variety of dynamic and electret (or standard) condenser types (usually cardioids); otherwise, choose only two or three types of microphones, including at least one cardioid dynamic type, and one electret condenser type.

Avoid sinking microphone connectors into the stage floor. Besides gathering dust, spilled drinks and floor wax, they are never in the right place when set-up time arrives. Either install them in a permanent box near the side of the stage and keep lots of long extension cables, or use a portable microphone "snake" cable and connector box.

Some performances will require several "splits" from each microphone to feed the main sound system, a monitor system or a recording or broadcast truck. Splitter transformers are available from Sescom, Jensen Transformers of Hollywood and other transformer manufacturers to aid in grounding isolation and impedance matching for a microphone splitter system.

The Mix Booth

Locate the mix booth so that the soundman can hear the sound system and see the performance directly, just like the audience. In fact, an audience location is by far the best choice. Place the mix booth in the audience seating area about half way back and somewhat off center left to right, and if necessary, sink it a few inches to avoid blocking seats behind it. If the booth is exactly in the center of the audience area left to right, and a portable system with split stage stacks (one stack of speakers on each side of the stage) is used, the soundman may hear the results of acoustic phase cancellations from the two stacks and have trouble doing an accurate mix. If the booth is too far back in the audience*, the soundman will hear primarily reflected (reverberant) sound, which again is an impediment to an accurate mix.

Unless there are plans for a large multi-channel mixing console, which will be kept updated, provide some place to put a sound company's portable console (which may be much larger than the house console) and submixers. And no matter how many microphone line connections are provided from stage to the booth, the day will come when that's not enough. Therefore, it's a good idea to install a large (about 8" diameter) conduit from stage to booth with a pull rope for the sound company's microphone snake cables, and another one (which probably can be smaller) for the cables from the console outputs to the stage amplifiers and monitor system. Separate the two conduits by at least 12 inches to avoid electronic crosstalk

** Past the so called "critical distance" where direct and reflected (reverberant) sound levels are equal.*

or oscillation.

Even if the building is designed for a flexible seating arrangement and stage location, making a permanent mix booth unfeasible, it's still a good idea to install conduits to several possible mix booth locations to avoid cables strung across the floor for every show.

General specifications for a mixing console should include: multiple inputs and outputs and subgrouping capabilities; inputs for submixers; flexible channel equalization; easy to use, human engineered control and meters. The console should be reliable, yet provide easy access for servicing.

Equalization

Equalization can be a very effective tool for a concert reinforcement system. It can help avoid system feedback, and smooth system frequency response for a more natural sound. If a house system is used for speech reinforcement as well as musical performances, it may even have two overall equalization systems so that equalization can be switched back and forth depending on the current usage.

Don't depend on an equalizer, however, for problem solving when there is a deficiency elsewhere in the system. For example, trying to compensate for a poorly designed speaker system by severe equalization will probably result in a severe loss of headroom. Similarly, acoustic problems are best treated acoustically, even though equalization may sometimes seem to be a simpler solution.

If at all possible, provide a method of bypassing the house equalizer. Equalization, besides providing a tool for the technician is also a tool for the soundman/artist. One soundman may want the system flat within ±0.5 dB. Another may put in peaks or dips to create a special "sound". Provided that the house system is adequately protected by a limiter and high and low pass filters, bypassing the house equalizer won't hurt anything, and it will make the sound company happier.

Other Electronics

In addition to limiters, and high and low pass filters, a time delay device can be very useful, and other "effects" devices such as artificial reverberation may be desirable. By incorporating these devices in such a way that they too can be bypassed, and patched into the system at different points (install them at the mix booth for easy access), they become much more valuable to the touring sound company.

Careful cabling, level and impedance matching, and proper grounding cannot be overemphasized since the house system will be connected to a variety of different portable systems, and if there are any problems with the house system, they will always show up when the interface is made.

Speakers And Power Amplifiers

In many cases, speaker components designed for musical sound systems can provide similar fidelity to the best commercial sound components and they are usually more rugged. Still it's a good idea to install protection capacitors on the high frequency drivers, and plan plenty of headroom on both the high and low frequency speakers. While a three way system will most likely improve frequency response, a good two way system can be more than adequate, and it's better to buy higher quality components for a two way system than to stretch the budget for a lower quality three way system.

The newer professional grade power amplifiers, when stable, are a good choice since they have extremely low distortion and hum and noise figures, and their high power can provide a system with improved headroom when compared to older, low powered commercial grade amplifiers. On a watt per dollar basis, these amplifiers may cost less than commercial grade units, and they are generally well built and reliable.

Monitors

On most large concert reinforcement systems, the monitor system is actually a completely separate system. It may have its own mixing console, taking splits from the main system microphones, and sometimes it may even have microphones of its own. It has separate power amplifiers and specially designed speaker systems. In the economic sense, a monitor system is like the main system: unless the system will be used for every show and will be expanded as necessary, it's a good idea to simply provide enough system for smaller performances while *staying flexible enough* to accomodate a concert sound company's monitor system.

Plan a spot for a monitor mixing console, usually off to one side of the stage, backstage where the monitor soundman can hear and see the performers. Provide a splitter arrangement for the microphone snakes, and extra cabling to feed the monitor mixing console. If there will be a moderate to extensive monitor arrangement, make it flexible. Have several different types of portable monitor speakers -- slopes as well as large side fill monitors, and possibly some permanent hanging monitor speakers. Keep in mind that the monitor system is often as powerful as the main system. At least one Las Vegas performer is noted for clearing out the first few rows of audience with the sound pressure level from his *monitor* system!

Politics

For the sound company:

A. Write a contract option clause requesting permission to use the house system at your option after inspection.

B. Call ahead of time to okay the use with the house management and to discuss the system and the interface with the house sound technician.

C. Be aware of house and union rules regarding the use of the house system, and decide who will pay any extra charges for its use beforehand.

For the house:

A. Protect the house system, and the house interests, but compromise whenever possible realizing that the house system may significantly improve the quality of a performance.

B. Plan new systems around the idea that they may be used to augment a portable system.

C. Have a qualified technician available before, during and after the show to help the concert sound company and to operate the house system.

Other Systems

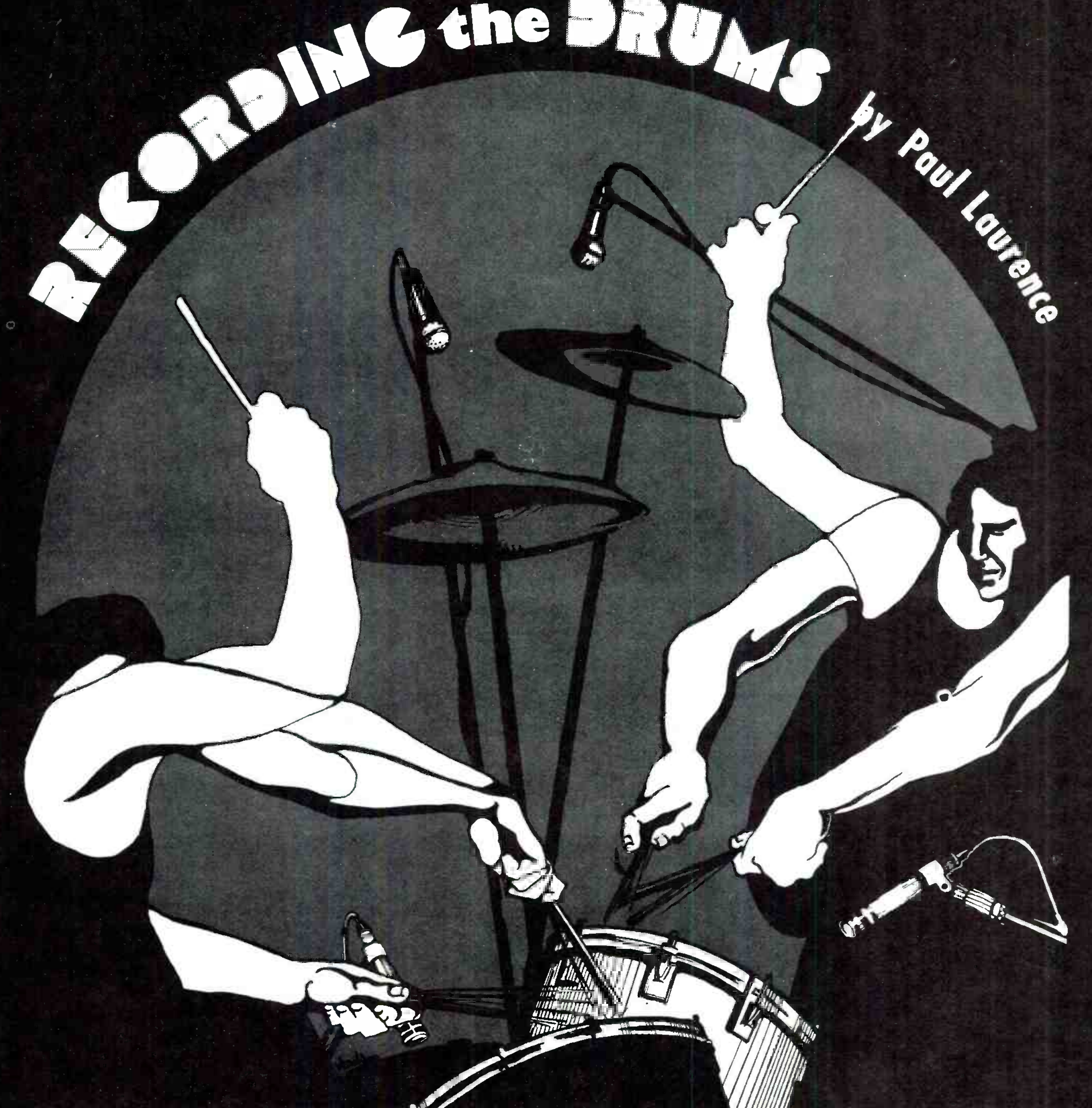
Many smaller clubs are now installing high quality systems and requiring that performers use them. This policy can result in consistant sound and a more attractive stage. However, by designing the system to be connected to an outside system as well, the club system can merge with a portable system when an occasional group needs more system than the club can provide. This concept, if carefully considered, provides a way for the club to meet almost any situation without having to lay out large sums of money for a system that is only partially used most of the time. Many sound companies keep portable equipment on hand that can serve a club system if the group needs extra equipment but their own is inadequate.

Even church systems can be designed for the occasional rock mass or special performance. By making the interface with a portable system easy, the church system can be designed for economy yet can be used for more elaborate purposes.

Summary

Especially for musical concerts, the synergetic combination of portable and permanent systems could happen more often if sound companies were aware of the potential, and house system owners and operators were aware of effective protection schemes.

The large scale portable systems were developed in answer to the general lack of house systems usable for a modern concert. These systems have now made large scale house systems unnecessary in many cases, giving the designer (and purchaser!) of new systems a welcome option.



The drumkit is possibly the single most challenging instrument to record. Not really one instrument, but an odd assemblage of drums you kick, drums you hit, cymbals, and often other things like tambourines and cowbells, all arranged in an irregular shape and occupying over 60 cubic feet of three-dimensional space. The permutations of just drumkit sound alone are practically limitless, and through the miracle of multimiking, there are probably equally many possibilities for the basic drum picture. Each kit can generate a wide variety of different textures, and over an extremely broad bandwidth, too – from under 20 Hz. on the kick drum to over 20 kHz. on some cymbals. Drums are by far the loudest acoustic instrument, and many parts require that they even be played loud. And lurking within each and every kit are the twin demons Leakage and Phase Cancellation.

continued overleaf . . .

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Considered as a whole, recording a set of drums presents the same challenges, on a smaller scale, as making a traditional classical recording (i.e., live-mixing a live performance to stereo); decisions must be made at the time of recording with due regard for not only tone, but (relative) level (of the individual components), space, and panning of not only the internal components but also the kit as a whole, all within the limitations of how many tracks you've allotted for the drums and whatever combining you've done thus far.

So it's fun. And as with everything else in recording, people have different ways of doing it. Because of the nature of the beast itself (the recording of drums necessarily entailing making a great number of interrelated decisions), many people have definite ideas and some even rigid "beliefs" about how it should be done. When you get right down to it, probably almost everyone concerned has at least a fairly systematized way of recording the drums.

What is presented in the following paragraphs, then, are the views of three prominent engineers, notable for their artistic relationship with an identified drum talent: MARK SMITH with Robie Bachman (Bachman-Turner Overdrive), KEN SUESOV with premier sessionman Hal Blaine, GEORGE MASSENBURG with Earth, Wind & Fire's Fred White.

MARK SMITH

Mark Smith very rarely limits drums, doesn't believe in damping, and is often nothing less than obsessed with achieving a perfectly symmetrical perspective. His influences are many and diverse, having played trombone in the symphony orchestra, been a mastering engineer for RCA of Canada, and having recorded almost every type of music imaginable, from Dixieland to Heavy Metal. He has amassed 27 gold and platinum records from the U.S. and his native Canada, in working with artists like Bachman-Turner Overdrive, Pure Prairie League, Rick Springfield, Paul Anka, Little Richard, and Gordon Lightfoot.

Philosophies Of Recording

Some of the greatest records have been made by very naive people, you know. There's another thing that you have to understand, and this is that you can get very carried away with all this stuff that we're talking about. And you can miss the whole point. Like for instance, you know there is the *artistic* recording engineer, and there is the recording engineer that can sit there and quote you serial numbers of transistors and things like that. I would much rather go with a recording engineer who doesn't really know that much technically but is very artistic, than a guy that sits there and analyzes technically and electronically everything that's going on. Because then you end up getting a very sterile kind of sound, as opposed to a "feeling" kind of sound.

I go for a very British sound. I learnt my basics from Phil Ramone -- in my opinion he's the best there ever was. He taught me to *understand* what you're doing. You just don't use something unless you know why you're using it. You

continued overleaf . . .

KEN SUESOV

Ken Suesov relates in terms of how it sounded to him from the trumpet section when *he* was playing trumpet in the symphony orchestra, does believe in damping, is disturbed by what he perceives as a general return to more of a "room" drum sound, and thinks of himself as the conductor while mixing. He has very specific notions about booths, and says that being a recording engineer is something that he would probably do "even if he didn't get paid for it" as a hobby. He has worked with Dr. John, the Mike Curb Congregation, Jud Strunk, and Jimmy Osmond (on whom, incidentally, he used a U47).

Philosophies Of Recording

My main job as a recording engineer is to try to find out what a producer wants in his record, and deliver that to him. You know, it's easy for me to say, "Well, this is how I think the record should sound", or something like that, but when I find out what the producer wants, then I can expand the sound that he wants in that direction.

I like music, I like being *around* music. Being a recording engineer is something that I would probably do, even if I didn't get paid for it, as a hobby.

My whole background as a mixer is not as Electronics Engineer. I'm not at all. I don't claim to be and I don't want to be. My whole background is music, per se. Playing trumpet, that's what my main instrument was. I got my B.A. in music and a teaching credential -- I was a high school music teacher for several years -- and I was in the process of working on my Master's in Composition when I made the transistion out of teaching and into working in studios.

I'm really into the musical aspect of

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GEORGE MASSENBURG

George Massenburg comes to audio engineering via an extremely technical background in electronics and hardware, did not play in the symphony orchestra (but did play bass and synthesizer for a bluegrass band), believes in damping, and is a meticulous student of every aspect of drums. His recording experience is both deep and wide: As a young engineer on the east coast, he was hired by a local poet to go to a pig slaughterhouse and do a remote recording of pigs "at the moment of their demise", which he later edited into a fatback beat with "a fat pig as a bass drum, a medium pig as a snare, and a little baby pig as the high hat". Among the acts he has worked with are Little Feat, Earth, Wind & Fire, and Mike Auldridge.

Philosophies Of Recording

I came across this last night. This is a book review about Blackett, who was a great British-American scientist and geophysical experimenter and worked for some of the heavies in his time -- Rutherford and Frank and some of the others. But this applies to recording -- this is on a paper called "The Craft of Experimental Physics" -- and one thing reads, ". . . With such varied manual and mental skills does the experimenter go about his work in the laboratory. An amateur in each alone, but unique in commanding them all". So my philosophy is that all of us are really amateur engineers and maybe amateur musicians -- you know, we're really in the dark. Compared to what scientists are doing on Mars, we are in the Dark Ages.

I've approached all this as an amateur in hardware, as an amateur musician, as an amateur ear . . . an an amateur diplomat. The thing is that the requirements

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use it for a specific purpose. Which makes so much sense, and it's helped me a great deal.

Too many people copy other people's techniques instead of understanding *why* they are done. I did a session once, and at the end of it I had to move some mikes I had on the drums to make room to move some big boxes. The next day I came to the studio early, and there was a session going on, so I stuck my head in and saw that the engineer was miking his drums the same way I had left them the night before. Left where I had moved them to make room for the boxes! I was amused but a little sad at the same time.

I make records like they make movies. Instead of duplicating exactly what is there, I take my own interpretation of it and try and make it sound bigger than life. I don't believe in arriving at a natural sound just by recording "naturally". In other words, running a microphone off the floor using no equalization, no limiting, no anything, and right onto a machine. I don't believe that means you're going to get a "natural" sound that sounds *good.* That's like making a movie to scale. I don't believe that you would get the full impact out of a movie if you were to see a guy like George C. Scott in "Patton" or something to scale on a screen. I mean, it would look ridiculous to see a guy who's six feet tall on a screen and see him six feet tall. That would look absolutely absurd.

The recording process is unnatural to begin with. There's nobody that can tell me that putting a microphone in front of something -- in an almost anechoic room -- and feeding it through wires, through walls, and then through faders and transformers and transistors and onto a magnetic piece of tape is "natural". It's not a natural process to begin with, so why for the sake of saying it's natural should you just put something in front of a set of drums or in front of a guitar and say "Okay, let's record that" 'cause that's "natural"? I mean, the fact that you are hearing music coming out of pieces of cardboard and metal is artificial, so why go so far and then disregard the most important thing, and that is the actual *idea* of recording.

Even *balance* on a record is unnatural. There are no pop records made that have a natural balance. There are no vocalists that can, without amplification, sing in balance with his or her band. This is a little extreme, but you get my point. "Reality" or "naturalness" on a record has to be arrived at by un-"real" or un-"natural" means.

Approach To Drums

I really don't believe that a recording engineer should be at the mercy of the studio, or should be at the mercy of the sound of the instruments that the musician has. If you have anything to do with it. I don't believe that if a guy comes in with a bad-sounding set of drums that that's the way it's got to be on the record.

Air is very important to have between drums -- otherwise you might as well overdub everything. You see, the drummer doesn't play *one* instrument. I don't look at it as one instrument -- I look at it as nine or ten instruments, you know? And so you treat each one of those things as a separate entity, even though in the final result, it's got to all sound like a properly balanced set of drums.

Usually, drummers, when they sit out in the studio and they play, and then they come into the control room, are absolutely amazed that the sound I'm getting in the control room has absolutely nothing to do with the sound that they are hearing live off the drums. But I've never heard a drummer ever complain about the sound of his drums in the control room.

I try and get as lively a drum sound as I can, but also as deep as I can. A lot of "weight" to the sound of the drums, and a lot of depth, but clear. Solid . . . all of those things.

Kit Placement In Room

There are so many reasons why you place drums *anywhere.* A lot of people will set drums up in corners, for instance. You get a lot more bottom end out of the drums in a corner.

Where you put the drums can depend upon the size of the room, the decay time in the room . . . I personally try and put my drums in the livest part of the room. I *hate* dead-sounding drums -- I can't stand them. So the livelier the better for me.

When you walk into the room, you know where the dead spots are, or you *should* know. You know from experience. I mean, you know when you walk out of your bathroom into your living room that there's a difference -- of acoustics -- right? If you can tune your ears very fine, you'll know that in your living room there are deader spots and there are liver spots.

You know by the construction of the studio and all that stuff before you even walk in. From the control room you can look in and you can see where the live parts are and the dead parts are just by the way they built the rooms. And then you go out and you walk around and you talk and you stomp your feet and you hit instruments . . . And you find out where it is. What I'm saying only takes a couple of seconds to do, you know. It's a very obvious thing in most studios.

Tuning

The tuning of the drums is the most crucial point in getting the whole sound of the drums together. And if you know how to tune drums properly -- to get the proper balance and depth and punch out of a set of drums -- then you will get the sound you want.

It depends on the sound you want to get. Sometimes you want a very deep drum sound, you know?, especially if you've got a lot of high-sounding things on the track. And other times, if you've got a lot of very low-sounding things on the track, you want them tuned high but still sound big.

I usually like the tom-toms down to where they start to give a little ripple in the skin and then tighten it up from there just a little bit.

Every drummer tunes his drums differently. I usually end up re-tuning them for the studio, but they're usually kept in the same ratio as the drummer has them. And it's also very important to make sure that the skin has the same tension around the rim.

I think something should be said about the way drums are built. For the studio, the diameter of the drum is not nearly as important as the length. I believe that drum manufacturers should be building small-diameter but very deep drums. The difference is amazing. Nigel Olsson, one of my favorite dummers, has a special set built like that and I just love the sound I get from him. Just listen to the new Rick Springfield album and you'll see what I mean.

Damping

I don't believe in putting all kinds of stuff covering the heads. I've seen

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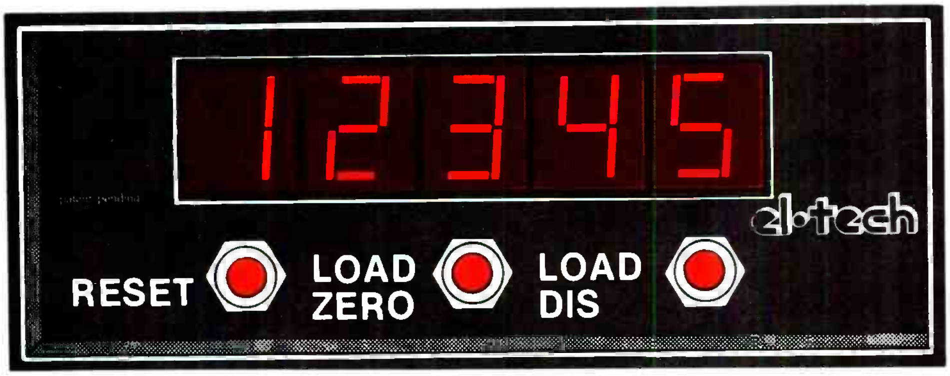

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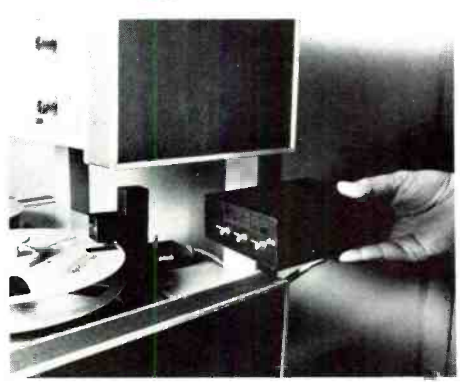

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guys drape cloths over sets of drums, and I really can't understand that. Of course, I'm only talking about my own preference. The next guy you interview can probably sit here and say, "I can't stand that ringing". Hey, listen, if it works for them, that's great, you know? It doesn't work for me. I don't like my drums sounding like cardboard boxes, and I don't like the snare drum sounding like a two-by-four.

General Miking

The number of mikes I use on the drumkit depends on the drummer, and it depends on how many drums he's got. If he's got a lot of tom-toms, I don't use a lot of microphones. If he only has three tom-toms, then I'll use a mike on each tom-tom. It depends on how he plays his drums, too. There's so many variables.

There was one session where I used 14 mikes on a set of drums. I mean, it depends on what you can get away with, but if they start causing problems, you have to start and analyze what is causing it.

I use as few mikes as I can get away with. If I could get away with using one mike on the drums, I'd use it. Then you start to add the other microphones in.

Now if the drummer plays fairly evenly -- that is, the ratio between his tom-toms, his snare drum, and his cymbals -- then you can get away with very few microphones. Of course, the bass drum is always miked.

Drums Booths And Rooms

I like drum *rooms*, okay?, as long as they're nice and big. And lively. I would much rather have to deaden down a drum room than to try and make it lively.

Booths usually are too dead-sounding. I would much rather have a drum *room* in which there's a complete isolation, and it can be from very lively to very dead or somewhere in between, if you want.

Kick Drum

I don't limit the kick drum. I've never limited or compressed drums. You see, most people do that to get punch out of it. To me, limiting the drums will give you a certain amount of punch, but it also takes a lot of the size away from it -- makes it sound small, very concentrated. However, you can still get the punch out of the kick drum and keep its size, just by the tuning of the drum and by mike placement and by equalization, if needed.

Snare Drum

I played trombone in a symphony orchestra for three years, and if you were to listen to any symphonic recording that has a snare drum on it, you hear a super bright, crisp, snare-y sound. That, to me, is the epitome of "The Snare Drum". Now I'm getting back to a purely natural -- and I mean "natural" -- snare drum sound in a symphonic hall. Especially from the trombone section. I'm talking about me sitting there. You gotta remember that all these things subconsciously prey on your mind, okay? My balance of string sections all come from playing in the symphony orchestra, from the trombone section. And the sound of the snare drum in a symphony orchestra I have never forgotten. And that is the kind of snare drum sound that I go to -- always. It's usually very, very deep, very snare-y, and very crisp on top. "You Ain't Seen Nothin' Yet" is a pretty good example as far as that is concerned.

I've had producers tell me that to get more punch, let's say, out of a snare drum, you should limit it. Now that to me is naive. I don't believe in that. If you want to get a lot of punch out of the drums, you've got to do that out on the floor. You have to. With miking and with the tuning.

Toms -- Left And Right Sides

I think of the drums as one whole instrument with how many other little parts to it, but they all must sound as though they're part of the same instrument. In other words, I don't like tom-toms that sound as though each one is overdubbed, you know what I mean? A little hit here, and a little hit there -- you know, very separate. I like a lot of air and I like them very lively.

Cymbals

I don't mike BTO's cymbals because he plays them so hard. But other guys, of course, depending on how they play them, sure.

I never mike the high hat -- never. Because I like a lot of top end on the snare drum, and that usually brings in enough high hat for my liking.

Overhead Mikes

I like a very, very bright drum sound. *Very* bright, and when the room mikes need to be brighter, instead of bringing in the cymbals, I keep lowering the mikes to keep the same amount of brightness but less cymbals.

Most people put overhead mikes over the drums. I don't know why, but they do. A lot of people put them in the middle pointing out, people put them at the sides pointing in, people use them from the back . . . I use them on the sides of the drums, from sitting on the floor to one or two feet off the floor. I also like shotgun mikes if the studio has them.

Distant Mikes

Yes, I have used them, but under very, very controlled circumstances only. Only when I can get a lot of separation from the other instruments on the very distant miking of the drums. In other words, if they were in their own room -- in a big room by themselves -- then I would use distant miking on them.

I've done that on most BTO things. On the "Four Wheel Drive" album, on "Hey You", the drums were in a separate drum room that had a slate floor and mirrored walls, and about a 20-foot ceiling. And I put microphones on the floor -- 87's on the slate floor -- and way, way up the walls -- 10 feet over the drums, to pick up the resonance of the room. And you can hear that on the album. And I also did it on the "Head On" album, in another studio, in which it was a drum *room*. The floor was linoleum and the walls were tile. And I used microphones in the four corners of the room. You gotta position them almost mathematically, so that you get a center snare drum and a center bass drum.

Phase Cancellation

Using different kinds of microphones within the same proximity of the drums'll get you in trouble. Let's say you use a couple of Shure microphones plus a couple of Neumann microphones. Now when you've got a set of drums, the mikes can't help but be close to each other. And then you start getting different phase cancellations and different colorations coming into one microphone that has a different frequency response than the other microphone. You start hearing swishy sounds coming off drums, you start hearing all kinds of different cancellations coming off of upper heads as opposed to bottom heads, and if you do a roll, all of a sudden one of the drums turns into sounding like a cardboard box 'cause there's a phase shift between the two microphones and they're cancelling each other out.

I get my drum sounds in mono. If there's any phasing going on on the drums, they will immediately show up in mono. In stereo, the phasing gives you a "spread-y" kind of sound.

For instance, if you took a bass drum and you split that bass drum onto two faders and you put them 180 degrees out of phase with each other, the bass drum will appear to come from the outsides of the speakers. But in mono, that bass drum will cancel itself. However, when you're listening to it in stereo, it sounds like a huge bass drum, but as soon as you go to mono, the thing disappears. So that's no good. And you know, all AM radio is mono, and FM radio, even though it's broadcast in stereo, a lot of people pick it up in mono. So a lot of those things will cancel out.

You have to be very, very careful about the phasing and the balance, especially if you're going into stereo drums. Stereo drums could be problems for a lot of reasons. If you get a lot of presence on tom-toms, and there isn't a great deal of air between the tom-toms, and you place them on the outsides of the speakers, you could get a great deal of left-right excursion on a disc, which drives disc-cutters crazy.

There are very few sets of drums that

I've ever heard on any record that doesn't have at least some kind of cancellation on it. Especially when they're using more than two or three microphones.

Tracks Of Tape

With 24-track I've used up to 12 tracks just for drums. But that was strictly for quad. It usually works out to six tracks -- bass drum, snare drum, stereo tom-toms, and external mikes. When I was doing 16-, it was four tracks -- snare, bass drum, and drums left and drums right.

You should try, if at all possible, not to mix down drums on a multitrack machine. If I know ahead of time that I can't have a lot of tracks for drums, I'll do the mixdown right from the floor. I would spend more time getting the sound and balance right and have it while the drummer is there. So I could be more flexible with miking and tuning. If you have to mix down drums off the machine, you *must* transfer the entire set whether you mix down the entire kit or not. You will get terrible phase-shifting if you don't.

Panning

I use the full spread because, when I record them, I try and get a full spread off the floor, but an all-encompassing sound rather than a tom-tom away off to the right and one away off to the left. When he hits upper tom-tom, instead of it being away off to the right as you see the speakers, It's kind of like out to the right but also right into the center, you know? It's pretty much always a full spread unless there is a lot of other instruments going on in the stereo. Then I will either put them in very close -- left-of-center and right-of-center -- or mono.

The high hat doesn't just go into the snare mike -- it goes into all the other mikes, too, so it gets pulled over -- about right-of-center, and that's just by leakage, from the upper tom-tom mike and the right external mike.

ROBBIE BACHMAN (Bachman-Turner Overdrive)

Recording BTO is kind of like a game as to how loud can you get the record without bending the needle. There's nothing really artistic about the engineering on any of those things -- it's just a matter of getting as much energy/sound onto a disc as you possibly can.

The drummer, Robbie Bachman, plays very, very hard and very frequently on his cymbals. And that's always been a nightmare for me. From the very first time I started recording him till now, it's still always a nightmare. Listen to their records. It's just all high hat and cymbals, very, very hard.

I don't mike any cymbals or any high hat at all -- ever -- with him. And he plays light on his tom-toms and snare drum and very heavy on his cymbals.

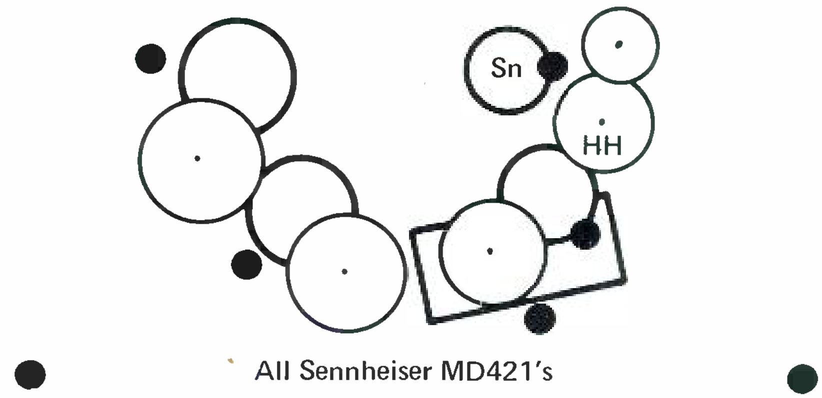

FIGURE 1
Mark Smith's miking of Robbie Bachman's drums for Bachman-Turner Overdrive's *"Not Fragile"* at Kaye Smith Studios, Seattle.

And so you have to go to very tight miking on the tom-toms and snare drum.

We did one album on which he wanted to use the front skin on the bass drum, okay? Now he had read somewhere that John Bonham does that, so he wanted to do that, not really realizing that their music is entirely different. Their instrumentation is so different -- BTO's is like crash from the word go until the end of the record, and they rely very heavily on punch and things like that.

But he wanted this "round" kind of bass drum sound with all this crashing going on, and there was no amount of arguing that would make him change his mind, so we went ahead and did it. I put a mike outside the bass drum in front of the front skin, and, of course, the sound of the bass drum was so "pillow-y" and very "soft" . . . It was "round" all right. If you listen to the drums by themselves without any other instruments, the bass drum sounded terrific. But then you start adding the guitars and the bass and all that stuff, and the bass drum was gone completely. I added as much EQ as I could. But when you start adding top end to the bass drum . . . if the microphone could think and say, "I don't want to hear this, I just want to hear the bass drum", that's terrific. But when you start with a lot of equalization on a bass drum that should actually be fixed on the floor instead of in the control room, you're starting to pull in cymbals. I had more cymbals on the bass drum track than I had on the rest of the kit!

For the drums on the "Not Fragile" LP, I used all Sennheiser 421's -- on everything. On kick drum, too. That was at Kaye-Smith, in Seattle. There was a mike on the bass drum, he uses one overhead tom-tom and two floor toms, and so I used one on each of those. He's got cymbals, lots of cymbals, all over the place. There's about three inches of upper tom-tom skin peeking through the cymbals up there, and so you have to tight-mike the upper tom-tom. He also has his cymbals very low. For instance, he's got a cymbal that's about two inches off the upper skin of the closest floor tom. And then I run a boom underneath a cymbal, underneath his high hat, and onto the snare drum. Pointing as far down as I can, to utilize as much of the front-back as I can off the microphone.

The 421's happened to work. In that particular case, that was the most microphones of one kind that they had. And it worked terrific because they were all the same and I had very little phase-shifting going on between the microphones.

To use just these microphones wasn't sufficient to capture air, so I had to use -- I don't like the term "overhead microphones" . . . let's say "external microphones" -- outside of the perimeter of the drums.

In this particular case, I couldn't get them very high, so what I did was I used two 87's, and I put them on the floor. *Sitting* on the floor, okay? Now this took quite a bit of time. I had a guy out there moving these microphones around, and I was just monitoring through these floor microphones. And all he was playing was his bass drum and snare drum. You try and figure out how on earth you get the snare drum -- which is usually off-center to the bass drum -- equidistant from the two points. It was a matter of inches -- just keep moving and moving and moving until finally the bass drum and the snare drum were in the middle. And there was all this air . . . I mean, it was really amazing. Working the microphones on the floor, I was able to pick up certain transients going through the floor, plus I was getting air. And the floor, by the way, was hardwood, so it was lively and I put the microphones in omni-. And it worked. I didn't get a great deal of cymbals through the things, but I got a lot of the bottom end when he hit his tom-toms.

There was one thing on the "Not Fragile" album in which the two floor toms on Robbie were sealed -- the holes were sealed up -- and hoses were put in the holes. And one of the roadies had to lay on the floor, and when it came time for the drum solo, had to blow into the tom-toms to give like a rising and falling effect when he hit them – almost like tympani. It was pretty funny.

Ken Suesov

engineering -- I'm not into the electronic part of it. I kind of think of myself as a conductor. I try to make my mixes sound like music, as opposed to . . . I imagine myself as sitting at the console as if I'm conducting. I don't necessarily have a score that I'm reading, but I kind of imagine the score as it would go by -- you just try to put it together musically. In other words, a conductor when he's conducting may want to hear a certain section of a symphonic orchestra louder or softer dynamic-wise. I think an engineer is really an extension of the conductor's ears and hands -- in other words, he's putting on record what a conductor would put to an audience.

Approach To Drums

I think of drums as the backbone to whatever you're working with -- it's like the skeleton.

When I mix, I don't mix my drums first -- I usually put them in after I have a lot of other instruments going. I usually start with maybe a piano and guitars -- get a good, cookin' rhythm track going -- and then try to bring my drums up so that they *push* that track, to whatever it needs to be doing. I don't do it that way every time. But I like to try to do it that way if I can, like on a Top 40-type thing. I don't know, it just seems to work best for me.

I try to bring the drums right up front, but not cover anything. If it's a disco thing, the drums would be the #2 volume right behind the voice. Get it up so that it's there and it's "Present", and you can feel it, you can hear it. Try to get as fat a sound as possible, but not let them cover . . . what the rest of the

track is doing.

I like a fairly dark sound, as far as snare and toms and kick go. But I do boost some 10k -- I boost at maybe 100, 200 cycles, but I also boost around 10k to get some presence back into it, to get some snap in the drums. To make 'em bite. But at the same time, I like to keep a lot of life in my cymbals, too. It's not like I want a real dark, thuggy sound to it, but I like them to snap when they hit, and I like the cymbals to ring.

90% of what you're going to come up with in a studio as an engineer is what the guy's drums sound like and who's playing them.

Kit Placement In Room

I usually look for a pretty dead spot. You can usually tell just by walking into a room -- walking around clapping your hands.

Number one, it would have to be in a location where everybody can see him, he has to be in a location where you can control the leakage . . . It's hard to say because every room is different.

Let me put it like this, I guess: It's probably not so much where you put the drumkit in the room, but it's where you put everybody else around it, too. You don't put the drums right next to the acoustic guitar player, unless there's some kind of baffling or something like that. You know, when you're there and there's a specific location and you know what a specific setup is, you know what you can do with it. But when you just say, "Well, where do you put a drumkit?", well, I have no set place. It would vary every time.

Drum Booths

I like to work with drum booths, but I don't like them in the next room either. I like them where people can be fairly close to a drummer, still hear the drums in the room, but at the same time . . . Before I built the one that I use, I did a lot of studying of what I liked and what I didn't like, and came to the conclusion that there was no way that I could ever get what I want and still keep the drums out of the room sound itself. So I've tried to construct a booth where the leakage that's in the room is good leakage -- it's not the type of leakage that's an annoyance.

I like booths to be 50% padding and 50% wood, and at least four inches thick of fiberflass insulation. On the one I built at Wilder Bros., I used two layers of 4-inch thick insulation. Now the drum booth can either be pie-shaped or kind of V-shaped and squared off on the back and the front of the V — almost like arrowhead-shaped, I guess. I would say pie-shaped — like a quarter of a piece of pie — would be more ideal. And there's a point right over the drums where the drummer would sit — his head would be like the central point — and at that very highest point probably about 8 or 8½ feet high, and then it equally slopes in all directions towards the front of the booth down to about six feet high.

I don't know if it would be unique. I've never seen another one like it, but I've seen things similar to it.

As far as isolation goes, it's not totally 100% isolated. There is some leakage coming out of the booth into the room, but the leakage is not the very highs — it's kind of mid-range leakage. And it's usable leakage. It's not a leakage that has such a high-pitched transient response that it's gonna cover up your other instruments.

That's what I've found with the projects I've worked on. It gives you a little bit of drum leakage into some guitar tracks or piano or something that doesn't hurt you, as long as it's good-sounding leakage. You know, if it's a real high, bad sound, then naturally it's gonna sound bad. But if it's a good-sounding leakage, you can use it and you can use it to your advantage sometimes to get more of a room *sound*, too. If you want a more lively effect.

Most booths are either too live or too dead. I guess this one is kind of a compromise between the two. I've seen some booths that are *so* dead there is no life to anything . . . It sounds like the drums were all filled with water or something. You know, the cymbals don't have any ring to them, and they're just kind of "toing" . . .

Tuning

I think drum tuning is about 90% of what you can do with drums. It's more-or-less up to the drummer or the guy who's tuning the drums, as to what those drums are going to sound like. Most people have good-sounding kits -- there's probably not very many bad-sounding kits themselves. I think about 50% of what the sound is, is the heads that they're using and the other 40% is the way they're tuned.

Most drummers I work with have to re-tune for studio playing. But I don't know if I could say how much down they have to go. See, that all depends, too, now -- if a guy is used to playing like a rock & roll type of thing, his drums are usually tuned a lot higher and a lot crisper-sounding than a guy who does primarily studio work and then does casuals. And he tries to make *his* drums sound in a club like what they sound like in the studio, and a rock drummer wants his drums to sound in the studio like what they sound like on stage.

I find it makes a pretty big difference on whether drummers use the butt end of the stick or the regular tip of the stick. Seems like a lot of young drummers like to use the butt end of the stick when they play -- they feel they don't have to hit as hard. But I find that drummers, if they use the tip end of the stick and play a little bit harder usually get more snap on their drums. You know, not a distorted snap, but just a more bite-y-sounding, crisper sound to 'em.

Damping

On snare drum, yes. Well, then again, it depends on that particular snare drum. Some guys can come in with a real old 4-inch snare drum and it just sounds great -- you know, it has a big, thick, rich sound to it. When I walk in, I see a guy setting up his drums, I'm hoping he's pulling out an 8-inch snare drum. And sometimes they will and it just sounds awful. A lot depends on the head of the drum. I can't really say how do I damp, because every drum is different. I usually try to put something on the top. You know, I'll go as mild as maybe a strip of tape across it or as heavy as a wallet -- you know, have the guy lay his wallet on it. It depends on what that particular drum sounds like.

General Miking

My technique for miking drums is pretty much the same. Of course, it varies slightly with each person's drum set and what kind of sound we're trying to get, too. My preference would be the 664 on kick with that particular console (at the Wilder Bros. Recording Studio) -- for some reason that just seems to be a good match -- and an SM57 on snare. Usually I use the Sony C37 on all toms and two of the 452's as overheads for cymbals, and a 452 on the high hat.

I try to keep my miking as simple as possible. It usually depends on the number of toms. I like to give a kick its own mike, a snare its own mike, the high hat its own mike, and every tom its own mike, provided that he's not playing a big Roto-tom type set.

I'm not real picky as far as what mikes I use. I'll tell you, it would be very easy for me to say: I like a dynamic on kick and snare, and I like condenser mikes on toms and overheads. The dynamic on the kick and snare seem to have the more thick sound to them, and the condensers seem to have the more warm, live sound to them. The combination of the two and the way that they're recorded and mixed, you can come up with just about any kind of sound you want.

I did a session once with a local group who wanted to do a demo thing, and the guy was using like 13 tom-toms and two kicks and a countless number of cymbals. I guess they had recorded at some pretty big places, and he insisted that he knew exactly how he wanted that kit miked. We used a total of about 16 mikes on it. We both thought it sounded awful, so we wound up using I think six mikes in the end for the best sound.

Kick Drum

I hardly ever limit the kick. I pad it

very heavy and I probably don't push my level any higher than about -5 to -7 on it. I pad it with everything I can on the console.

I like the kick to be punchy, but I don't like it to "slap" at the same time. If you get up in your higher frequencies -- say about 3k on up, to maybe 10k -- if you add maybe six or seven dB in any of those frequencies, you can hear it start to "pop" more when it hits. And I try to get the "pop" out of it and get more of a "thud" sound to it.

I'm hearing a lot more of the "pop" sound on records lately. Drum sounds are kind of going back to a more room sound, as opposed to a real studio sound, and so I'm hearing more and more of what I've been working for all these years to get away from. It's kind of creeping back to being what people want now.

Snare Drum

Instead of a really thick, bottom-y snare sound, people are going for a little bit thinner sound, and a little "bite"-ier sound. They add a lot more highs to 'em, they use a lot more overheads in the actual drum mix, shotgun mikes, room mikes -- that sort of thing.

If they want a real Top 40/Pop type of sound -- if the people are that oriented, if they're from that market -- I'll go with a good dynamic mike on the snare. If they want a real live -- what I would call "rock" -- type of sound, then I usually use a KM84 on the snare, which is a condenser mike.

The sound I like on most snares now is the SM57 -- it has a real thick sound to it. I'm not a big person as far as saying "You have to use this mike . . .". I really believe that the guy's drums is 99% of what you can do with them. You know, you can't make a very live set sound very dark, and you can't make a 4-inch snare drum sound like an 8-inch snare drum, you can't make an aluminum snare drum sound like a fiberglass or a wood snare drum. Everything is different.

Toms -- Left And Right Sides

I like to try to keep them as individual as possible, but then I put them back and forth, so that when a drummer hits his toms, it doesn't necessarily go from a stereo effect like *left* to *right* -- it'll go maybe left, right, and back to the left again.

Usually if a drummer's playing a break of some sort, if they're spaced left and right, you can usually push them a lot harder and they're not as pounding as far as getting in the way of a mix. It's not like they're extreme left and right and they just come out of nowhere.

I like to hear a tom strike here but kind of work its way to the middle. The toms themselves are usually spaced extreme left and extreme right, but I usually run my overheads more towards the middle of the kit.

There's a lot of controversy on how do you mix for AM radio and still get a great stereo sound and that sort of thing, and this is what I like to do: I like to bring my overheads in kind of left-center and right-center, so that when you're hearing it on the radio, you're still hearing . . . everything is still biting through right in the middle, but when you're hearing it on a record on a good stereo system, they kind of bite at the sides and kind of work in at the same time. They hit here and kind of creep in -- that sort of thing.

Cymbals

Whether I mike the cymbals or not depends on the drums and how the drummer sets up his particular set of drums, but 99% of the time they're miked. I usually try to mike them with some AKG 451's, or U87's.

I usually add 3 or 4 dB at 10k on most all the cymbals.

When I mike the high hat, I don't mike the edge of it -- I put a mike right directly over the center stem of the high hat, and point it straight down at it at about four to six inches above it. And depending on what mike I use, I usually add a lot of top end on it. It seems like we're doing a lot more disco-type things now where they want that really high, sizzly sound on a high hat. It seems like more and more stuff we're doin' is in one

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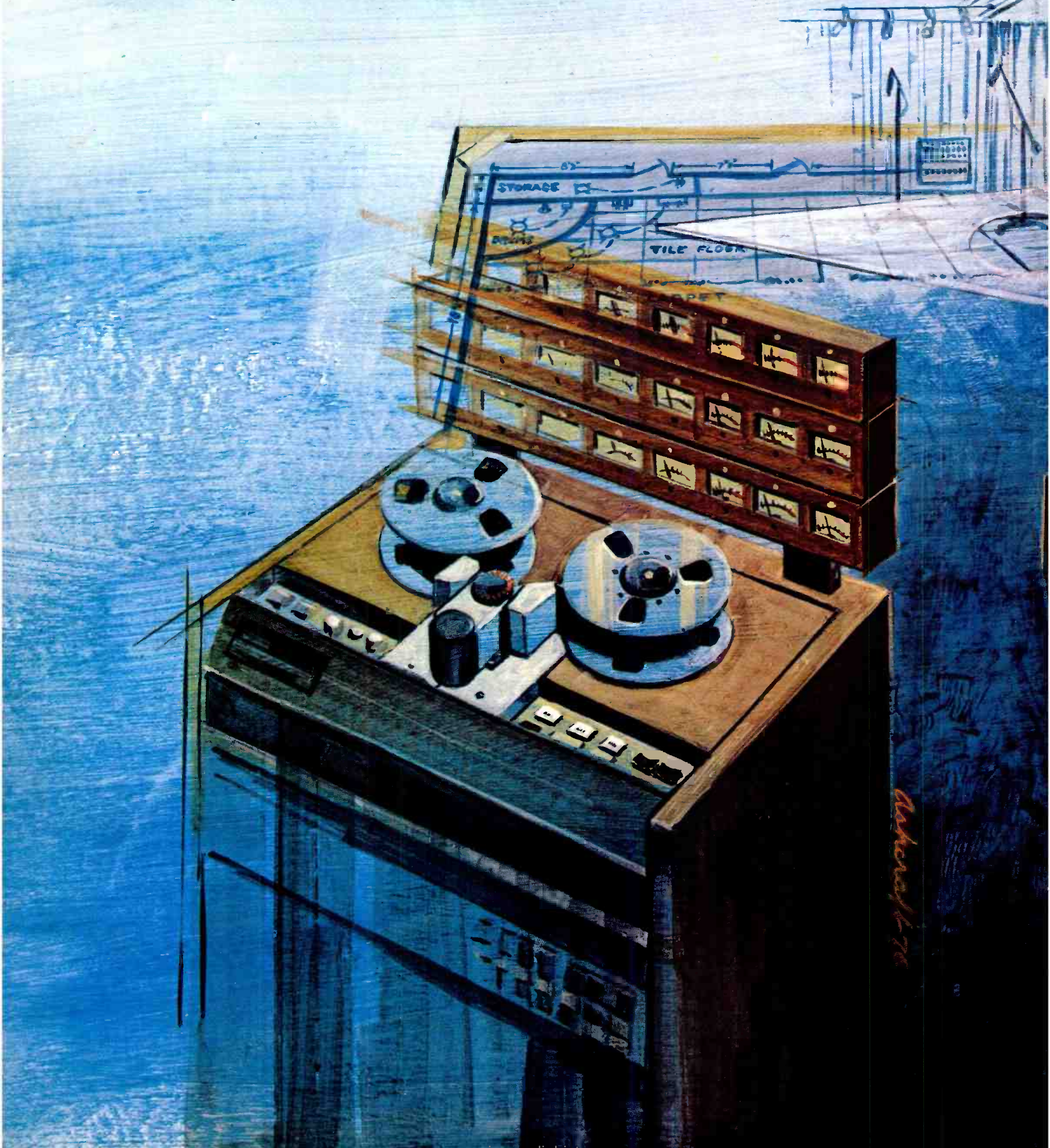


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of those two classifications, and they want that real "tssst" sound to it, as opposed to a "splash" sound. I'm changing a little bit, in some of the mikes that I use, and I'm adding a lot more highs. I'm adding sometimes 8 or 10 dB at 10- or 12,000 cycles. If I'm really straining, and I'm trying to pull some highs from somewhere, I might use 12k. Most of the time it's 10k.

Overhead Mikes

If the guy has cymbals on both sides, I use two overheads. I put mostly cymbals on my overheads, but at the same time I always put my overheads where they can pick up some toms, too, so I can get that "center" sound out of my toms. I always mix them in with the left and right sides.

I normally use two overheads, for stereo cymbals and toms. You know, I close-mike the toms, too, but I use some overheads to pick up some kind of distant tom sound, plus my cymbals. They're mainly there for cymbals, but also to pull the toms into the center a little bit. The overheads are what bring the toms to the middle.

Distant Mikes

Well, that's a whole other ballgame. I find that people usually come in two classes: they come in with people who are with groups, or they're producing an artist and they bring in studio players. The groups are usually a little more rock-oriented than people who are working with a single artist. So when you're doing that, you're getting into more of a live sound. That's when I use my overheads up real high, right up next to the top of the drum cage. I almost don't get any left or right sound out of 'em, but I get more of a total drum sound. And when I'm doing a mix, I'll usually run those on separate tracks – my two overheads by themselves – and I usually bring them up as my drum sound. And then my drum left and right tracks and snare I bring in to fill up the gaps whre I need 'em. And if there's not enough snare, then I can push the snare on the snare track itself for a little more snare.

Phase Cancellation

Well, the more mikes you use, the more chance you're taking of that. I know the studios I work with so well – I know what I can do and what I can't do as far as where I can put mikes – that I really don't worry about phasing on drums. If I'm working with somebody who I work with a lot, I probably don't check it. When a group comes in – say it's somebody I've never worked with before and they bring in a new drummer, yes, I'll check.

I really don't worry about phasing in my drum sound because I don't hear any of it, and I've never had a comment from anybody saying, "Hey, I hear something out of phase in yours drums".

I don't run into the problem too much. Unless, you know, I happen to be using a cord that's out of phase or something.

Tracks Of Tape

I usually only use four tracks. Like a kick, snare, and a drums left and a drums right track, but when I'm *recording* the drum left and drum right tracks, I'm very careful that I'm getting plenty of overheads in there, too, so that they're still getting sort of a middle sound out of them.

Panning

I don't like to spread just for the sake of spreading. I like to full spread like a piano left and right, but the way I usually mike a piano, it doesn't sound like "left" and "right". It sounds like left-center and right-center.

The smallest drum spread I would use would be completely mono. It depends on the type of song you're doing, and the type of mix you're going for. If you're going for a strictly AM radio type of thing, I may put them as close as . . . almost mono, maybe a 40%/60% type of thing. If I'm that close, that's almost considered mono. You know, if you're trying to get a mono type sound so it sounds good on AM radio – not for like an album mix – I generally keep my drums in closer.

I like my kick and snare right at 50% – they're right at the middle. I usually put the high hat at about 25%. I like to keep it away from the snare so I can get a distinct and separate sound on the high hat itself, as far as if he strikes the high hat and the snare at the same time, I want two different sounds. And so when you spread one of 'em a little bit, you can get that, and you put them right together and sometimes they almost sound like one.

HAL BLAINE

I used to do a lot of work with Hal Blaine, and I generally never used more than five mikes on his whole set. I'd use a kick, a snare, two overheads – a right and left overhead, positioned like right in the middle of his toms facing out, sort of V-shaped – and a floor tom mike. Using five mikes on his set was some of the greatest drum sound I ever got.

He's very easy to record. You can hardly go wrong. He's very consistent in what he plays, and you can get an overall level and not have to worry about it for the rest of the session. And his sound . . . His drums always sound great.

I was using a Beyer M160 on his kick and snare. And he usually sets up in a way that you can get a fairly good position between the snare and the hat, and you can work one mike as a balance between those two.

He uses a special set of toms. It's a custom-made set, and he has like a whole set of toms . . . I believe he calls them Roto-toms. They're on two stands that he wheels in, and I thknk there are three toms on each one.

I used a set of Synchron condenser microphones as overheads, and they were placed kind of right above his head and in front of him. With this setup, I got a great tom and cymbal sound, but I usually didn't get a whole lot of floor tom unless he hit it really hard, so I usually ran a C37 on his floor tom. So five mikes on it mostly. And I've seen other mixers come in and work in that same studio with the same guy and you know . . . They're using up to 12 microphones on him and they don't come up with nearly as tight a sound as I came up with, I think. It was a very clean, tight sound.

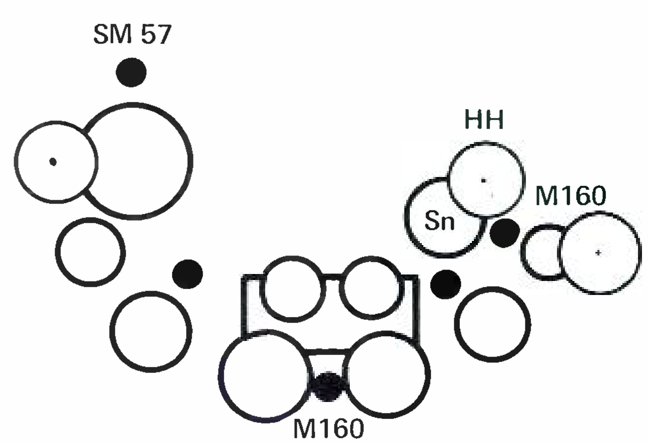
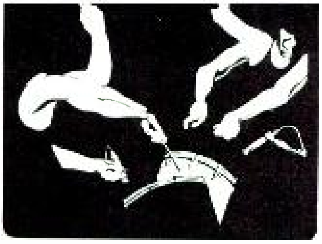


FIGURE 2 Ken Suesov's miking of Hal Blaine's drums (Dubbington Downs Studio A, Hollywood)



George Massenburg

continued from page 33 . . .

of each of these categories have to be satisfied. And too many people, I feel, approach it just from the sound – "Make it huge", you know? And to me, the real timely recordings – the things that *I* listten to – were at some point archivally correct. Tom Dowd is a great case. The cat was good. For "King Curtis Live at Small's Paradise", he took a 351 that he'd converted a little bit out to Small's Paradise, and did a great remote. Consummate album – and my favorite saxophone player bar none. He works efficiently. I don't know whether he discussed this with you, but it was important to Atlantic Records when they were releasing five records a week. And the executives at Atlantic liked Tom Dowd because he could get it out of the band quickly. That's also a good idea musically because things do get stale if you keep going over and over them.

You want to have what you want to have when you want to have it. A clean sound, to me, is the hardest thing to go for – it's easy to shake something up and screw around with it and make it "dirty" or "huge" or "big" or "thick" or whatever negative kind of processing you want to do. But for a very clean sound, every single component in the chain really has to sound like a straight piece of wire.

I'm later dissatisfied with everything I do. I think most engineers are. But the professional side of it – the side that does keep you working–is even though you're dissatisfied and even though things aren't going right, that you can still handle what you're doing professionally and objecttively.

As an engineer, I feel that my first responsibility philosophically is to represent the music properly. If you're a musical producer like Phil Spector, you can make your sound and screw the musicians and whoever's out there . . . Well, that's okay, that's his thing, but I don't think that stuff is archival stuff, by any stretch.

Approach To Drums

Drums are very interesting, because they're one of the last citadels of live recording. Pretty much everything is electric now. So when you're talking about live recording, you're talking about drums, they're one and inseparable. You're talking about "The Lost Art of Live Recording".

I love space. I really love that full, fat quality that comes from having a tight sound on the bottom and a good spread on the top. The English really have it down. I like Bill Schnee's better than anybody else's. The penultimate drum sound – it was *huge.* Another good one Andy Johns did, and I'm surprised he didn't bring it up. He did a thing for Gary Wright called "Extraction", with Alan White the drummer. Terrific drummer – the best English drummer going, I think. That was my favorite sound at the time.

Kit Placement In Room

I work in five or six different studios, and it's different in each studio. Now in Heider 3, you can put the drums in one place, and there's a dead spot about three feet away where you can put a vocalist and have very little leakage. You put the drums back here and it's muddy as heck. But it's a room to get used to. If you move the drums around, you get different kinds of sound. Ray Thompson is the man that knows the room best, and he works the room like he's playing an instrument.

Where I place the drums all depends on the group. *Everything* about drums depends on the group – everything.

Drums Booths And Rooms

Again returning to Heider 3, they're building a big booth off to the left-hand side in between the studio and the office, and they either haven't had enough time or enough energy to finish it, so all it is is dry wall. It's irregular shapes because

GEORGE MASSENBURG --
(continued)

of the nature of the building -- and it's the livest room at Heider's. Now on these last couple of projects, I've put congas in there for something, just to cop that sound on one instrument. Or maybe put a mike in there and leave the door to the studio open.

The rooms in any studio can make the most profound contribution towards the character of the sound. Most studios have enough different rooms adjacent to the control room so that a room sound might be selected more live or more dead. The Sound Factory, on Selma, has three different rooms -- three sizes. The medium-sized room sounds better than most drum booths, and the isolation is useful. Hollywood Sound Recorders "B" Studio is really one rather large booth -- very dead. They then have three excellent live echo chambers, and you can make up a sound to order. Kendun has a new room that can be partitioned with glass. That's provocative. Westlake is really a lot of booths, and not meant for live recording, per se. But you can put the drums in the hall, the synthesizers next to the desk in the reception room . . . and so on. We've used the hallway at Sunset Sound for an extra booth for the last two weeks, sounds fine.

I'm now engaged in designing a studio complex -- two studios and a mix room. I'm planning to build my main room more along the lines of the classical acoustical designers with some unused modern theory added. I use a set of room coefficients pointed out by . . . L. Sepmeyer in the Journal Of The A.S.A. Using computer vector analysis, he found four sets of coefficients for room height, length, and depth to optimally distribute the modal response. I'd like to see a reverberation time which is, although short enough for modern studio recording, somewhat variable. Also quite diffuse and very linear. And there certainly is a difference working a room with solid -- and I mean concrete and hardwood and plaster -- walls and ceiling. It makes a great difference in the character of the low end in particular, not to mention the fact that it can more easily be made quiet. Our main room will be a 50 to 60,000 cubic foot space, which should suffice for live recording.

I should add that it's not often that you can mike the room in recording rhythm tracks these days -- you can't tell what will be added and subtracted in overdubbing and mixing. Consider the flexibility of The Burbank Studio's Music One -- a studio with a volume in excess of a million cubic feet! I feel that at least the *orchestral* sound -- like strings, horns, percussion -- is better there to warrant its consideration.

Tuning

I'm a tuning nut. As early as '69, I had a strobe tuner in the studio. And tuned the musicians up.

I like to see how drummers tune their drums. I mean, there are some *good* drummers in this town. The best! Jim Gordon is a big man, and he can really smack drums, but knows how to tune drums, too. Like, he tunes the bottom head a fifth above the top head and tunes 'em to good notes -- Hal Blaine does the same thing with the set that he's got. Except I don't think he uses bottom heads. But a lot of drummers have not the slightest idea how to tune drums.

Drummers have got to tune it as they hear it, but sometimes I can help them. I have it in my repertoire to help them out. Bernard Purdie tunes his top head tight as a mother . . . He's maybe the best rock drummer ever. At least he's had more licks stolen from him than any other drummer, short of the guy that played in Iron Butterfly -- "In a Gadda Da Vida". Which is one of the least imaginative lines of all time.

The kick is really important, but it's not tuned to a note -- it's tuned *close* to a note; usually one can only hear it in perspective in the control room.

It's funny, but you don't build your tuning from any particular point. It seems to come together a couple of different ways. You get the toms in tune first. Fourths, fifths -- whatever's convenient -- relative to the root, I'd say, of the tune. If you're doing a weird tune, you might try to distribute the tuning so you have more of a thud than a note. Hal Blaine's got a set that he can almost tune chromatically.

My favorite drummer these days, for a combination of vitality and expertise -- mostly he's a very, very funny person -- is Jeff Porcaro. Excellent, excellent drummer. And he knows how to tune toms. He can also whip up a different snare drum sound and lock in right on the snare drum sound that you're asking for. And that's good. Jim Keltner pretty much has one snare drum sound, and it's great so you never ask questions.

You gotta go by ear. In the middle of a session, if you've really been going at it for hours and you've been listening to the music, you can develop a sense of near-perfect pitch -- you really can -- that allows you to feel the note that you're getting on a tom, you know? Just feel it, not really hear it. I guess you could say endurance is important . . . patience.

Damping

It's optional. It all depends on the tempo of the tune and what you're trying to do. Unfortunately, you're at cross purposes -- if you're trying to get a fat snare, then it depends on that low-frequency resonance, you know? You may want a tight snare, but you still want some snare sound on top -- you *want* some top on it, some hang-over. And if you damp the low end, you're gonna lose the top as well as the bottom. What I used to do with the snare was to try to build inside a resonator of sorts with a circular piece of Micro-coustic ¾-inch, but I don't have time for that these days. It took out the heaviness in the low end -- it distributed the low-end resonance -- but still gave a full drum sound. More often these days, I'll use a parametric equalizer (not an Orban) to remove the resonance.

You're trying to damp that low end in the snare, unless you're going to use it, like in the big Yes tune . . . "Roundabout". *Awful* snare sound. It worked, 'cause it was in some kind of relative tuning. But it was the worst snare sound in the world. It was proper, so I guess it's the best snare sound, and it sold a million copies so I guess it's the very best. But aesthetically speaking, it was terrible.

The best thing I've found for bass drum is like a feather pillow from Holiday Inn. You cut a hole in it, get half the feathers out -- you know, real crummy old pillow -- and get that just right in the bass drum . . . The thing is to make it loose -- not so packed. And you can experiment a good deal if you're on the road. If one doesn't work, you just leave it for the next patron.

Generally a roll of gaffer's tape and some medical gauze takes care of the toms, if necessary.

General Miking

The old school of putting up one microphone to cover the drums was safe and expedient. Not to mention the fact that we couldn't put more because we didn't have them! And if we did, we ran out of inputs. When boards expanded to more inputs and more than one pan-pot, engineers could begin to indulge drummers, who had been asking for more mikes all along. Multiple and tight mikes mean more problems, but in a way we're locked into the concept. The big problems we face now are phase cancellation and distortion, among others.

As a normal setup these days I use about eight mikes: a kick, three toms, two overheads, a snare and high-hat. Sometimes I'll use all eight. But my approach is to get the basic sound in perspective with the overheads and the kick – moving the mikes physically until they sound best – forgetting the tight mikes for the time being. Then I'll set the high hat the same way, then the toms. The snare is last and I always end up putting it on another track like some sort of a misbegotten son!

I don't use one-inch mikes on drums as a rule, though I've tried to on occasion. The U67 and 87, U47, U47fet, M251 are all large capsule mikes, and I find lacking in transient response and high-frequency response too. I rather perfer KM84s, Sony C37s, C451s, and so on, by comparison.

However, because different microphones color the sound differently, I'll use any one that fits the circumstance. It is, after all, a *sound* that the artist is going for . . . I change the snare mike with the seasons.

I most often look for the spread. The snare starts in the middle and fills out after the hit. The tom's start here and end there. The high-hat isn't way over there relating to nothing at all.

I've been using an M421. Where I place it all depends. Sometimes a little loose. If you want it real tight, you damp it, and right up against the head and tune for that. If you want it a little loose, you can just move it back and forth in the kick and come up with a different sound.

I don't limit until mixing, usually. It's hard to use a limiter on drums and have it controlled when you're trying to think of so many other different things, because drummers I work with generally will at least play a ltttle louder as they get into the session. Using a limiter on the kick? The kick mike picks up a lot of snare drum, to the point where you almost always have to reverse the phase of the kick. As a matter of fact, I do. With a standard-phase microphone, you reverse it on the kick, so the feed from the snare drums comes back in phase. It sounds a lot better, too, 'cause the kick is getting into the overhead mikes. That's an important point, I used to do it a lot, but it wasn't till I saw some other people doing it that I really felt good about it.

Dave Hassinger at the Sound Factory is a nut on phasing, to the point where if you've got bass guitar leakage in the room, he can show you that changing the bass phase relative to the drum phase makes enough of a difference so that you should consider it. The purpose is to keep one note from ringing out more than the others. You know what happens when two things are out of phase – you lose a note, then the next note'll be heavy, then you limit the bass more and more and it'll still happen, and you're going a little nuts. It's not a cure-all, but what you could do is throw the bass out of phase relative to the drums.

I use a C500 sometimes, unless the guy plays loud high hat, I use a C451 with a 20 dB capsule pad on the snare sometimes. It's a pretty linear mike – it' got a funny diaphragm and they're not consistent. It's a bright mike. And, It's a great acoustic guitar mike with an omni capsule.

It sounds best with an omni- capsule. Sometimes that doesn't help much. Although few cardioid or directional mikes remain so at low frequencies, the high frequency response is directional enough to warrant their use under certain circumstances. Lou Burroughs' book is a good starting point for those who aren't aware of the concept of the 'directional mike' which all too often isn't directional at all, or is directional in the wrong place for its application.

I've used -- and still do -- a lot of different snare mikes. Recently I'm using a KM84.

GEORGE MASSENBURG — (continued)

Toms -- Left And Right Sides

I use C37A's for tom-toms when I can get them, and I'll go all over town to get them. Now a lot of people — Donn Landee and John Haeny — use C37A's. The C37A is a tube mike, and it's a very linear mike from low levels to high levels. It's not particularly flat, but it does have a pleasing response — you know, smooth dip in the mid-range and kind of up on the high end — and It's nice on acoustic guitars. It's a good all-around mike. The thing about a C37A is it doesn't break up on that transient — you know, when you smack a tom hard, it doesn't seem to crap out as fast as a condenser.

I like the attack to come from a certain spot, wherever it's panned. But you have to understand that in the mix, you can pan these stereo overheads together and put amatrix onthem and bring the center down and you have a very nice mono track, too. With a lot of air. So you can use it either way. I'll play you a tune that Dave Hassinger and I worked on where the panning of the tom was really important to the perspective of the guitar vs. the tom. And it was very interesting.

Cymbals

I've been asked to mike the cymbals, in jazz sessions. Surprisingly, some of the drummers that I've worked with in jazz sessions have asked for more mikes when you would assume that jazz would indicate fewer mikes. I mean, Hal Blaine supposedly has sneered at engineers when they put up more than three mikes! We did a jazz date where I was miking the front and the back of the cymbals, and top and bottom of the toms. I guess every engineer's tried it. Ken Scott is a master of that but . . . I think he uses it subtlely. At any rate, I got a pretty good sound: but I wouldn't play it for my grandchildren.

Again, I use 84's.

Overhead Miking

For overheads and high hat and many other things I use KM84's. It's the best mike I can find short of the B&K 4148 or 4136 . . . the capsule, that is. Like most of the Neumann microphones, the KM84 has a one-transistor pre-amp. It's cheesy . . . a true masterpiece of economy . . . very compact. It sounds terrible! Could it be that this is part of the reason for the resurgence of interest in old mikes? I think so. Right now I'm using a two-transistor pre-amp — much better FET, tighter specs, flatter transconductance. And I'm not operating it at a lot of gain so the high level distortion is low. Of course, the output is lower, but in the recording studio this hasn't been a problem — certainly not on drums. I'm working on an esoteric front end for condenser mikes, and it certainly won't be phantom-powered with all the power it'll consume. Whatever it takes to make it sound good. Bud Wyatt, at the Mastering Lab, has been working along these lines. They used a Bud Wyatt line amp to buffer the output of M251's on the last direct-to-disc thing. You've got to hear the comparison to believe it. Getting rid of transformers has been like a crusade for both of us . . . and there is one in almost every condenser mike!

Distant Mikes

It's the same to me as miking the piano or miking the room. If I've got an acoustic piano in the same room, I'll set it up to where it picks up good drum leakage. I don't usually distant-mike the drums, first 'cause I don't want to mix it in with my master tracks with your getting in trouble for overdubs, and second because I don't want to use another track.

Phase Cancellation

Everybody knows it's happening, but there's not much you can do to get away from it. It's very simple in an anechoic chamber, but it's very difficult in a live room. One of the factors obviously is keeping all of the sources as close to the mikes as possible. Of course, that's not possible on drums. But you figure it this way: Since you are in a live situation, the highs are really random-phase -- it's hard to call out the phase of your high frequencies. It's also hard to say what is a high frequency in a live situation. It's a soft curve -- you know, it doesn't happen -- bang! -- you're out of phase troubles. Your main problem is low frequencies, and with two mikes far enough away, you could say, I guess, that your low frequencies are out of trouble. Because low frequencies have long wavelengths. Say you're talking about 200 cycles. It has a complete wavelength of some six feet. So maybe you're out of danger if the two mikes are 5.65 feet each away from the sound source, or your problem is minimized if the source is inches closer to one of the mikes, but God help you if one is at the half-wavelength and the other is at the full wavelength.

You can't just look at a miking setup for a drumkit and see that there could be problems with phase cancellation, nor could you say that it won't sound good. Because maybe -- just maybe -- you've happened on a setup where the phase cancellations occur at unwanted frequencies and aren't occurring at those very strong frequencies.

With phase cancellation your toms will sound weak -- you'll say, "There's no bottom on the toms. Give me more tom mike, or make it closer". That's a dangerous one -- miking the toms a little too tight, if you're going for a real full sound.

I very seldom boost the bottom. Few engineers go for hard boosts on the bot-

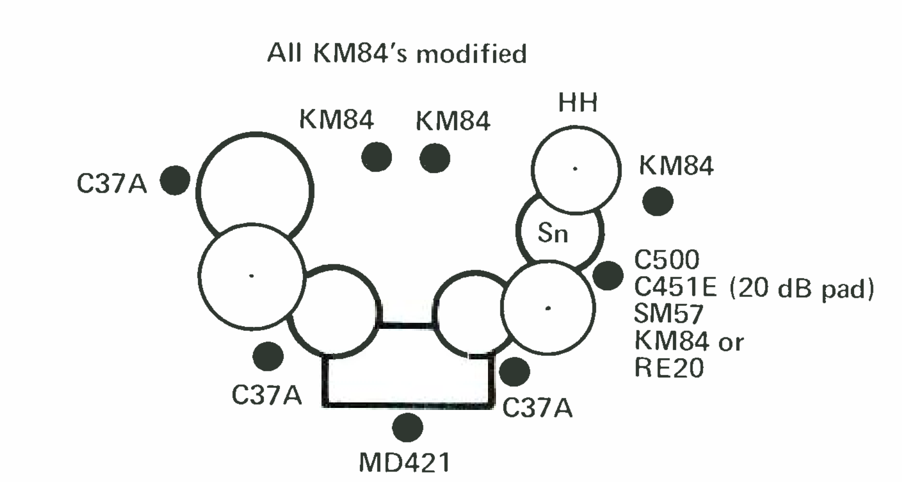


FIGURE 3 George Motion's miking of Fred White's drums (same setup for all LP's and at all studios)

tom, for a coupe of reasons I guess. One of which is that it tends to become muddy, because equalizers have phase shift, too. You know, there's so many factors coming into effect that it's hard to say, "Well, I don't do this because the phase numbers are such-and-such".

Of course, when you're in the studio, you don't always think, "Well, I want a live sound", take out my calculator and check the phase, get out a ruler and say, "You must stand there", and put an "x" on the floor, windscreen in front of the mike, and a dummy mike to make sure that he's . . . You know, it's pretty ridiculous. You gotta go for a sound.

Tracks Of Tape

The way I put drums down on a master is kick, snare, and everything else in a stereo set. Four tracks on 24-track, three tracks on 16-. Used to do it on two tracks on 8-. One track on 4-.

Panning

I usually run the high hat right, snare dead center, kick dead center, and toms spread. I'll pan the overheads left and right, but I'll move them to get the correct perspective, or the kind of perspective you need.

FRED WHITE (Earth, Wind & Fire)

With Earth, Wind & Fire, you have a group of five excellent drummers. Phillip's a good drummer; Maurice was an old Motown drummer and a standard fixture at Chess, was Ramsey Lewis's drummer for years; an accomplished jazz drummer and a very accomplished rock drummer. And you better believe he has his idea of what a sound should be. It's not the kind of sound that I go for by myself -- it's a hard crack on the snare and not a lot of that warm low end, and not a lot of open space.

Freddy White is so steady that it's beautiful -- the guy's just steady, all the time. He has a very consistent hit on the snare drum, so once you got your sound, it doesn't change a hell of a lot. You know, if a drummer hits it harder or changes the angle of his stick, if you've been in the studio for 36 hours, it drives you *nuts*! You know, you're saying, "What have I done? The drum sound has changed", and it's the drummer.

Fred's got a good hard kick, and all of the mike techniques apply except that maybe you'll go for a lower snare drum sound, because the guy can approach drums from that perspective.

There are two drum tracks on "The Way Of The World" -- kind of a jazz/shuffle/lot-of-cymbals track, and that was Ralph Johnson. That's a straight-ahead track. The overdub was Fred White and Ralph Johnson, Freddy playing a snare and Ralph playing claves miked with an RE-20 and about 10 feet away with a C451, both mikes taken into a special compressor that I built. It has four attack time true RMS sensing circuits that allow for variable crest factor. It just tightens it up -- it squooshes the snare down real hard. But still gives you some little attack on the onset. In other words, it lets a little bit of click from the claves through and a little bit of the attack from the snare through. And then releases real fast, without so much distortion. Since it's true RMS, and I do a phase rotation system, I really am only attacking on the actual energy envelope, and I get very low distortion values out of it for extremely fast releases. And what it does is it brings up a lot of room sound -- makes it sound real thick.

"Shinin' Star" has four or five instruments Kepexed off a snare. Handclaps, organ, and two guitar tracks were Kepexed off of one overdubbed snare drum.

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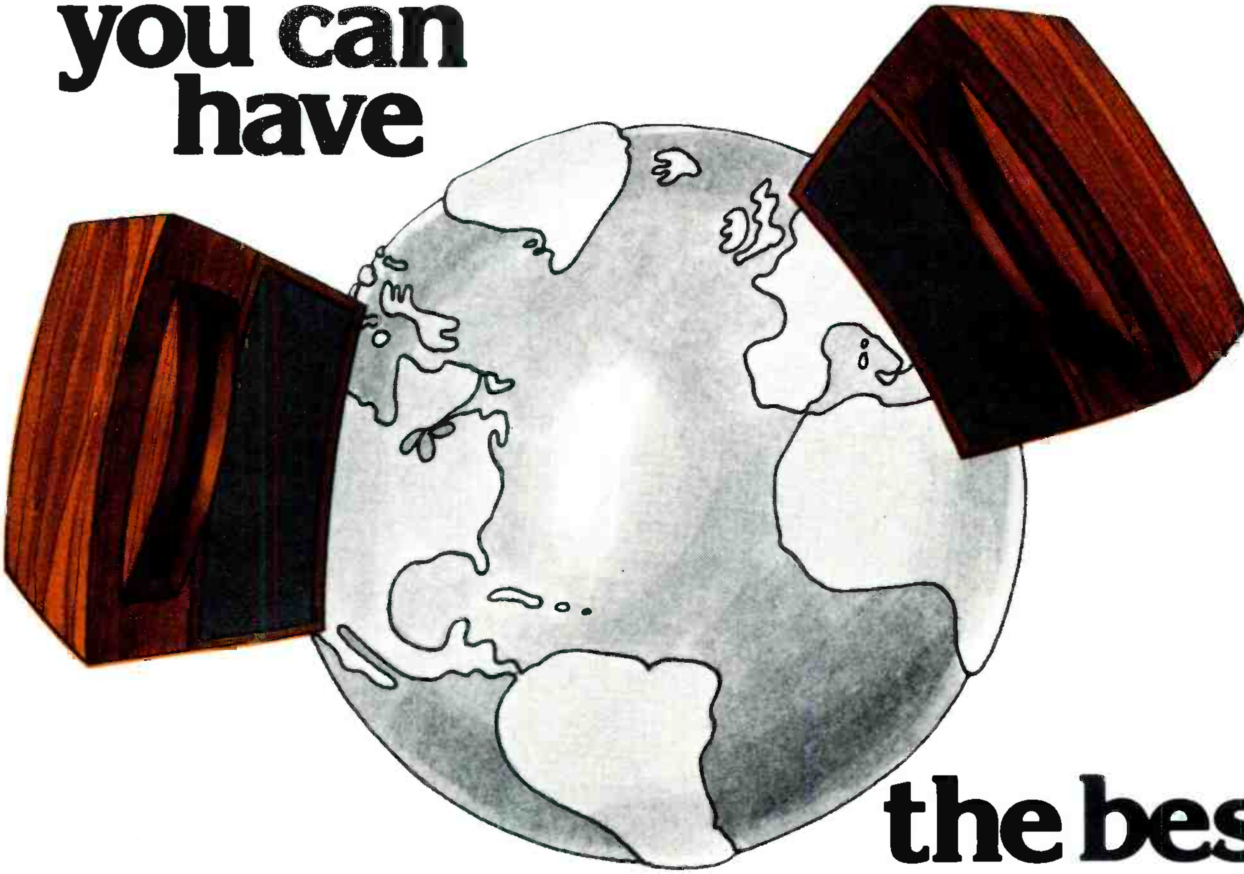

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(non creative) **PHASING effects**
in multi-track magnetic recorders

by
ANTHONY DEAN
Senior Field Service Engineer
Ampex Corporation
Nashville, Tennesse

As a Field Service Engineer for a major U.S. tape recorder manufacturer, I am frequently asked questions, and become involved in discussions with studio recording personnel concerning what is commonly referred to as "Phasing" in regard to multi-track tape recorders. The phenomenon is no more than signal delay between channels of a multi-track tape recorder. The consequences of this time delay are most frequently observed as differences in the phase relationships of a phase-coherent signal recorded and reproduced on the same or different machines.

Although this term is something of a misnomer, but as it has now entered the language of the industry, and most of the readers of this article will have some understanding of the term and its meaning. I shall continue to use the term throughout this writing and I hope that the reader will gain a more full and complete understanding of "Phasing", its effects, and how to cope with them.

The "Phenomena" is in fact caused by the following:

STATIC SKEW
DYNAMIC SKEW
GAP SCATTER
HEADS NOT PERPENDICULAR TO TRANSPORT

I will attempt to give detailed definitions of these tape transport inadequacies, and their effects on "Phasing". The phasing problem manifests itself when an attempt is made to mix high frequency material, which is coherent in nature. If the material is music, the phasing problem will be most evident when the instruments have a large amount of "highs", e.g. strings, sizzle cymbals. I have seen engineers attempt to make a mono-mix of the 15 KHz tone of an alignment tape, the results of which are usually a disaster. Fortunately, the industry is in the business of recording music, and a competent engineer can, for the most part, avoid situations where the phasing limitations of a multi-track recorder will cause major problems.

The normal "Phasing" characteristics of a well designed machine can be compounded if there are any *PHASE REVERSALS* through the system, this is not limited to the tape machine, but should include all the peripheries. I cannot over-emphasize the importance of checking for phase reversals in both radio stations with production facilities, and in multi-track studios. A new tape machine should be checked for phase reversals before being put into service, and when repairs cause major parts of the signal system to be replaced, such as installing a new head assembly, such a test should be a mandatory part of the checkout procedure.

A simple test for detecting phase reversals, which requires a minimum of equipment, is quickly accomplished in the following manner: Place a diode across the output terminals of the oscillator, which will of course clip one side of the signal from the oscillator. Record the clipped signal from the oscillator, at normal level, and a frequency of approximately 500 Hz. Connect the oscilloscope across the signal input to the recorder and determine which side of the signal is being clipped. Now reconnect the oscilloscope, this time to the output of the recorder. The reproduced signal should be clipped in the same direction as the recorded signal (See Fig. 1 for details). If a reversal is found it must be located and corrected. This is sometimes easier said than done. A helpful tool here is a tape on which all tracks have a clipped signal recorded on them, the direction of which is known. By viewing the output of each channel in both the Play and Sel-Sync modes, it is easy to determine if a reversal is in the record or the reproduce circuitry. You have now reduced the problem by 50%. The problem can now be found using conventional trouble shooting techniques.

Before we can continue in our examination of "Phasing" it is necessary that we do have a thorough understanding of *STATIC SKEW, DYNAMIC SKEW, GAP SCATTER*, and *HEAD PERPENDICULARITY.*

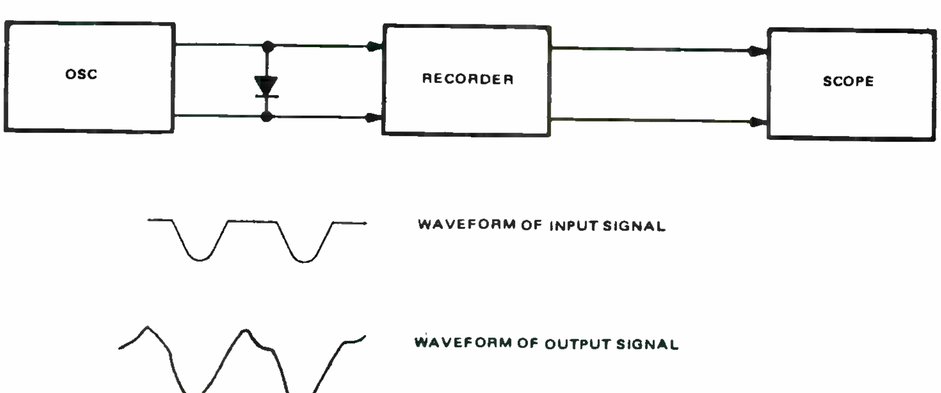


FIGURE 1:

STATIC SKEW

When tape travels across a tape recorder head assembly it appears to move in a path perfectly parallel to the base plate. However, it does in fact follow a somewhat oblique course due to mechanical tolerances in the guiding .system. These include, but are not necessarily limited, to the following: Imperfections in the flatness of the top plate, due to machining tolerances, or by the fact the plate is not of heavy enough material to give the required stiffness; tolerance of tape guides; head not perpendicular to base plate; scrape-flutter idlers either not perpendicular or worn out. On a closed-loop transport a very frequent culprit is the "turn-around" idler which may be out of perpendicularity or be worn out.

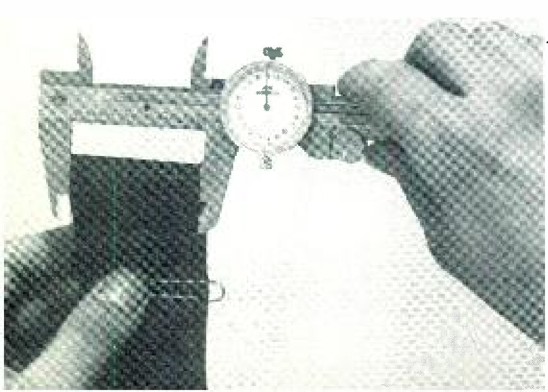
The actual tape being used is also a factor of no small importance. Here, it is a matter of tolerance in the slitting, and also how well the manufacturer maintains the slitting mills from the point of view of producing a product which is not "cut on a bias", i.e., equal tension was not applied to both sides of the web when the slitting took place. This can, and usually does, produce a tape which will handle very poorly in drive modes.

The mechanical condition of the tape is a matter of such importance that it deserves a few more words here. Badly slit tape can be determined in several ways. First, hold a five to seven foot length of the suspect tape up so that it dangles freely above the floor. In the absence of drafts, the tape should hang in a straight line with no tendencies to curl. Any deviation from the ideal should be cause for suspicion that the tape has been damaged due to stretching or non-ideal slitting.

Poorly slit or deformed tape will generally not wind smoothly onto a take-up reel in normal play mode. Inspection of a reel of tape that has been allowed to play through for a few minutes may reveal a rough, uneven texture to the tape pack rather than the smooth, even surface normally observed.

Tape widths are specified to be within +0.000, -0.004 inches of their nominal width dimension. Tape that has been slit wider than spec will show up by jamming as it passes through the machine tape guides. Casual visual inspection will reveal deformation typical of this condition with little difficulty. Detection of tape that has been slit *under* specification may be accomplished by measurement of the tape with a micrometer or dial caliper as shown in Figure 2.

FIGURE 2: Tape width may be measured using a dial caliper or micrometer. The tape is held in a loop as shown. The jaws of the caliper are opened wider than the tape width and gradually closed until the edges of the tape no longer pass through. With practice the method will measure to 1 mil increments. A strategically placed paper clip aids the process.



An interesting point to note here is that in my personal experience I have found that back-coated tapes generally guide better than the non-coated variety. (See Figure 3).

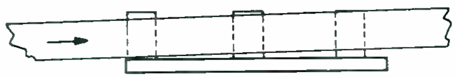


FIGURE 3: STATIC SKEW

DYNAMIC SKEW

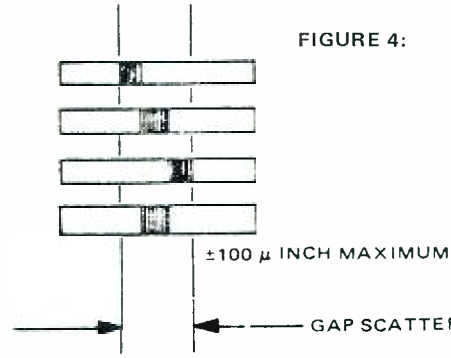
In defining static skew it was stated that the tape did not follow a path parallel to the base plate, but instead at a slightly oblique angle. This angle, however, is not a constant and is subject to random variations. The tape tends to make fractional up and down movements in the guides. The quality of the guiding system is a prime factor, from the point of view of the machine, in holding dynamic skew to a minimum. The guides must be held to extremely tight tolerances, and must be fabricated from a very hard material. Guides made from stainless steel or synthetic gems are often used on high quality transports. If we were to have a transport with all the tolerances falling in such a way as to make the guiding path perfect, the transport would still have dynamic skew. The reason being that tape is always slit with a minus tolerance to permit it to travel through the guiding system of a theoretically perfect transport. As stated earlier, tape which has been slit on the bias will cause very high levels of dynamic skew.

GAP SCATTER / HEAD PERPENDICULARITY

When head stacks are assembled from the individual heads, and remember there can be up to twenty-four in one stack, the mechanical tolerances are extremely difficult to maintain. Each head *must* be in the same azimuth relationship to all other heads in the stack. The gaps of the heads should be in line with each other. Errors in alignment of the gaps is *GAP SCATTER* and a manufacturer should maintain this to less than 200 Microinches (See Figure 4).

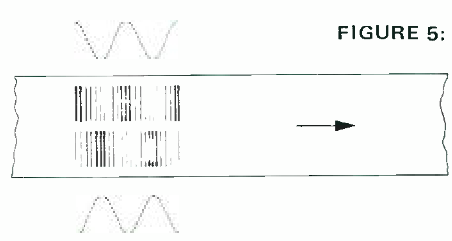
The end user has no control over head perpendicularity on multi-track machines with fixed heads, and relies on the integrity of the manufacturer. In the case of multi-track systems where the end user can adjust head azimuth, a perfect head would produce optimum high frequency response when the head is perfectly perpendicular to the base plate; however, as on any precision device, there are tolerances. On non-adjustable heads the perpendicularity is held to less than ±.15 minutes of arc.

FIGURE 4:



In further consideration of "phasing" consider a signal of 15 KHz recorded on four tracks of a ½-inch machine operating at a speed of 15 ips. If the transport suffered from none of the inadequacies mentioned in a previous paragraph, the outputs of these channels could be observed on an oscilloscope and they would all be in the same phase relationship. We know that this is not possible. The transport *must* suffer from the shortcomings of Skew, Gap Scatter, etc. If the 15 KHz is generated by a common source, the signals applied to the record head will be in phase with each other, and any phase differences caused by the electronics are so minute that they can be neglected. If this tape was developed with Edi-vue or some similar compound and we were to examine the point of minimum flux density on all tracks, which would be the cross-over point of the recorded signals, it would be observed that they would not be perfectly aligned with respect to each other (See Figure 5). This mis-alignment would be caused by the transport imperfections already discussed. When this signal passes to the reproduce head the problem will be compounded, and it may be said that we have a "phasing" problem.

FIGURE 5:



From what we now know, it can be seen that the phasing problem will become worse as the recorded frequency increases, e.g., the wavelengths become shorter. This gives us a clue to one simple solution, which, although not a panacea to our problem, will give a great deal of relief . . . increase the wavelength of the signals recorded on tape. We don't want to limit the highest frequency recorded, so: As wavelength is equal to speed divided by frequency . . . use the highest tape

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speed that is practical. Note that 15 KHz recorded at 15 ips will give a wavelength of .001 of an inch. If the speed is doubled, this will be increased to 2 Mil.

On 1" and 2" tape transports, the use of outside tracks for recording high frequency material should be avoided, for here the effects of head perpendiculatity will become more noticeable. On machines equipped with adjustable heads very careful alignment is of the utmost importance.

The following alignment procedure has given me very good results. This alignment is for a ½-inch four channel machine, but is quite valid for other multi-track systems. The tape path must be thoroughly cleaned and all the guiding elements must be in good condition and adjusted per the manufacturers instructions. Be sure that there are no phase reversals in the tape machine and that all test cables are free of reversals.

After de-gaussing the heads, load the machine with a full track test tape corresponding to the speed at which machine will be operating when in normal service. Obtain the required test equipment listed below in Table 1, and connect to the machine under test as shown in Figure 6.

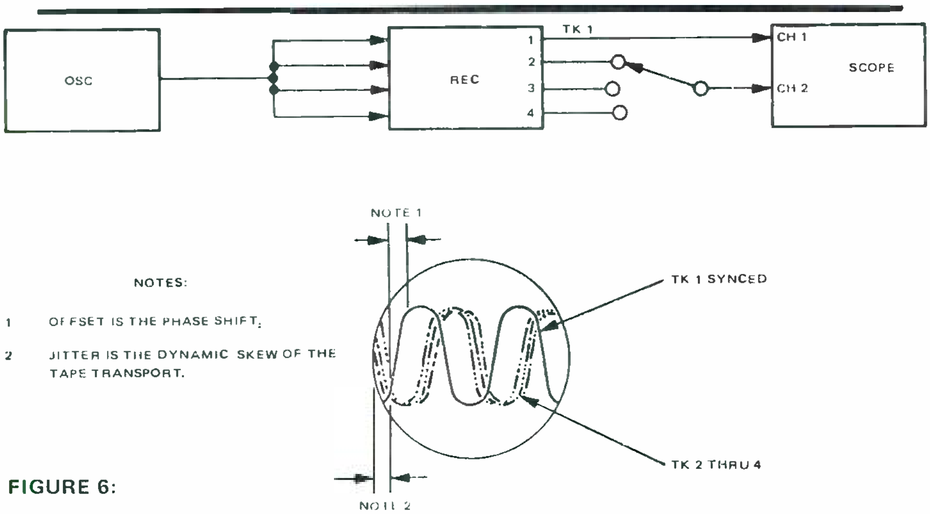


FIGURE 6:

TABLE 1

Alignment Tape: Choose a full-track alignment tape conforming to the flux-frequency response, tape width and reference level appropriate to the specific application.

Oscillator: As available.

Oscilloscope: Tektronix 453 or equivalent with two vertical input channels with the capability of syncing to one channel to the input signal while the second channel derives synch from the same source as the first channel.

Test Cables: As required.

Blank Tape: As required.

When the alignment tape is threaded on the machine, be absolutely certain that all channels are in the safe mode to prevent the possibility of erasing the alignment tape. The first tone on the tape will be a reference tone at 700 Hz. Set playback level to zero on the VU meters. With the sync being derived from track one, adjust the oscilloscope so as to display tracks 1 and 2, which, at this frequency, should be displayed on top of each other. Making no change regarding track 1 look at tracks 3 and 4. They should also be on top of track 1's display. Now repeat this procedure in the sync mode. If any of the tracks displayed are displaced with respect to track 1, the head is either out of perpendicularity or has excessive gap scatter.

Return to the play mode and monitor the second test frequency, which is 15 KHz. Observe tracks 1 and 4 and adjust the play head azimuth for a maximum reading on the tape machine's VU meters. This will give a display of the two traces on the oscilloscope, both of which are sync'ed to track 1. A very slight adjustment of the azimuth on the reproduce head will now reduce the time difference between the two waveforms displayed on the oscilloscope. The phase-shift between the two wave forms is proportional to the amount of time delay between tracks 1 and 4 in the play mode. Now, take a look at tracks 2 and 3 in reference to track 1. The phase shift should be essentially the same as when looking at tracks 1 and 4. *CAUTION:* When adjusting the azimuth for minimum phase shift, there should not be a reduction in output on any channel in excess of 2 dB. Having switched the electronics to sync mode, repeat the procedure for the record head. When the azimuth has been set at 15 KHz it is recommended that the test tape be monitored on the oscilloscope as the other frequencies are reproduced. The phase shift *must* improve as the test frequencies become lower. If this does not happen, it is possible that azimuth has been set to the wrong peak and the procedure should be repeated. If difficulties are experienced in finding the correct peak, the azimuth may be set on the oscilloscope using a lower frequency.

On older machines with limited high frequency response in sync it will be necessary to use this same technique to set the record head azimuth. The alignment tape can now be removed and the machine threaded with blank tape. Place all channels into record.

Adjust the bias level on all tracks in an identical manner. Perform a complete record system alignment, setting all tracks to identical signal levels and high frequency response specifications. If all record head tracks are not biased very similarly, there will be apparent differences in the observed geometry of the record gaps. The reader should remember that the point at which the recording process occurs is not at the actual gap of the recording head pole-piece but at some point after the tape passes the trailing gap edge. The higher the level of the biasing field, the farther away from the gap edge the point of actual recording will be. Failure to keep this fact in mind can cause a bit of confusion in the mind of the technician. This is why the setting of record head azimuth in sync mode may not yield the same results as setting it in record mode.

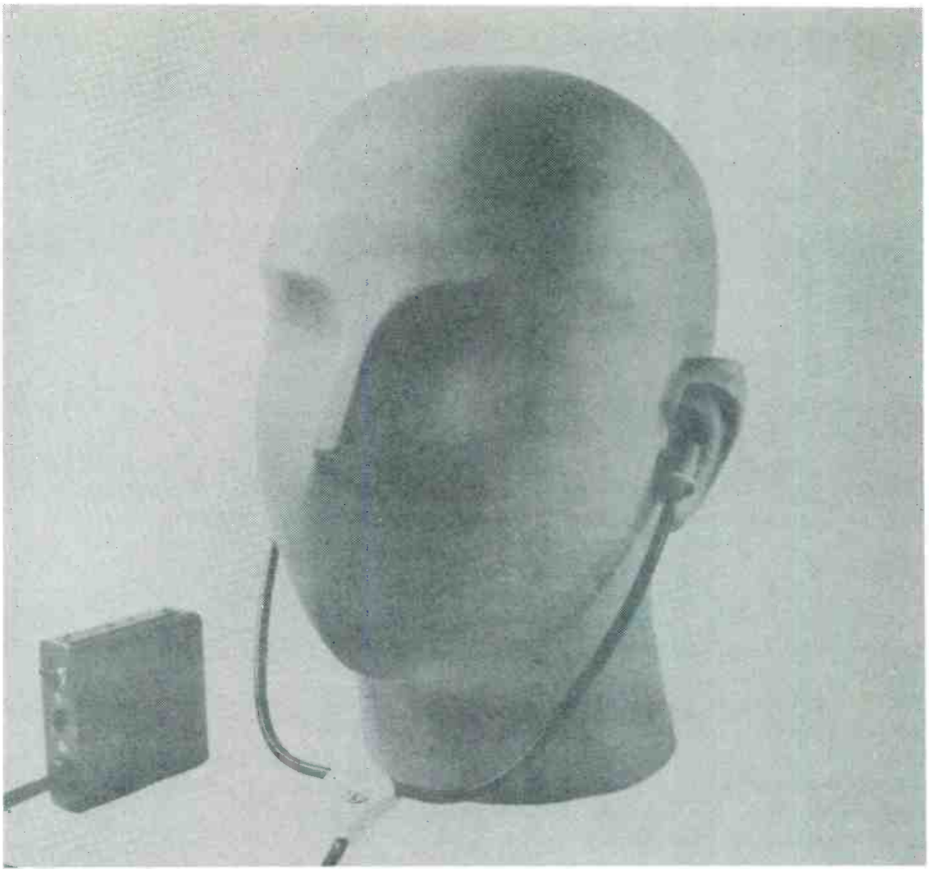
With a 15 KHz signal present at the inputs, check for phase shift on the oscilloscope.At this point, if the shift is tolerable, no further adjustments should be made. Of course, an improvement could be made by adjusting record head azimuth, but this would tend to degrade phase shift characteristics when a tape is played back on another correctly calibrated machine.

It should be noted that it is not absolutely essential that a dual trace scope be used. A Lissajous figure could be used instead, but this makes the adjustment much more difficult to accomplish and necessitates more complicated calculations to obtain a numerical value of the phase-shift.

The Lissajous pattern does not easily yield an indication of the polarity or sign of an indicated phase-shift. A lead of thirty degrees looks the same as a lag of thirty degrees.

To calculate a value of phase-shift, proceed as follows: Adjust the sweep so as to make the 15 KHz signal occupy ten divisions on the scope. This makes each division equal to 36 degrees. It is now possible to make direct phase-shift readings from the scope.

When service personnel observe the procedures I have outlined, and when recording engineers recognize the limitations of the multi-track tape recorder, it may be that in the future, "Phasing" as a problem, will slowly fade away into the ambient noise level and be heard of no more.



DUMMY HEAD (Binaural) RECORDING . . . Does it offer alternatives or advantages?

A review of the psychoacoustic phenomenae involved, as well as a report on recent developments . . . . by Howard Cummings

BACKGROUND

The governing laws by which the human ear can perceive distance and direction are, even in this scientifically advanced age, not yet entirely clear to researchers. Different theories and differences of opinion still exist, some dating from the 1870's, but there remain a few premises which are obvious: 1) That when a sound originates from a certain off-center direction one of a listener's ears will hear that sound as being louder than will the other ear. There is an *apparent* difference in the sound's intensity as the sound reaches each ear. 2) Unless a sound emanates from either directly in front or directly in back of a listener, a time differential will exist between the time it takes the sound to reach each ear.

Efforts to better understand these effects have led to experiments in what may be broadly classified as Binaural Recording, also known as "Dummy Head Stereophonie". Binaural recording ("two ears") differs from stereo recording in the essential dimensional perspective each is capable of effecting. The difference is that binaural sound attempts to exploit *the psychoacoustic* principles of intensity and directionality in all planes, while stereo finds it difficult to produce an aural perspective any broader than the two concentrated sound sources originating in the left-right plane.

True, by incorporating the correct phase and time delay relationships between stereo signals the human brain can, indeed, be subliminally deceived into hearing movement and direction in several planes. These sound impressions which the brain constantly receives continually update the memory-association process to help identify a particular aural experience with a certain distance or perspective. So, what we are relying on in appreciating the art of today's stereo mixer is, to a good degree, an acquired aural response to sounds we have been hearing throughout our experience: the relationship between the sounds perceived by the left and right ears resulting in the so-called *phantom center*.

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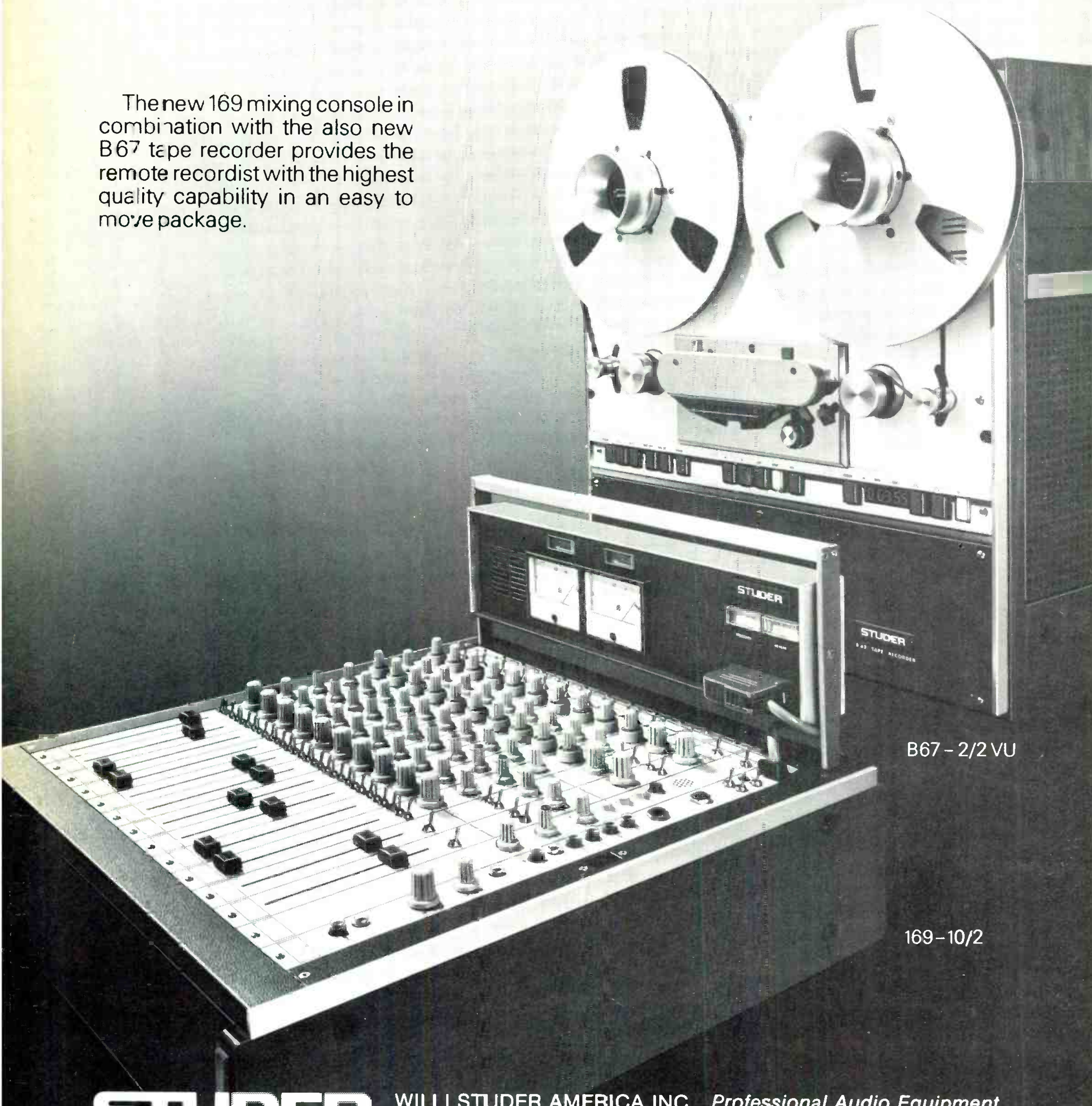

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one pair of microphones (left-right effect) which will also yield 360°, "all around" sound requires more than just placing the mircophones a short distance from one another, as if to simulate ears.

Prior to describing the research done by Sennheiser, the West German maker of highest quality microphones and earphones, in developing their Kunstkopf (artifical head), it may be interesting to review some supportive research in the field of psychoacoustics as it applies.

Tests conducted by Roffler and Butler and published under the title, "Factors That Influence the Localization of Sound in the Vertical Plane", in 1968, conclude that the ability of listeners to locate sound in the vertical plane was governed by three factors: a) The stimulus must be complex. b) It must include frequencies above 7 kHz. c) The pinna (ear flaps) must be present.

Complex stimuli, the report concluded, are more readily localized than simple stimuli, because localization is made easier by changes in the waveform of the stimulus due to the differential delays which vary in source and pitch. I. C. Whitfield makes this point in his book, "The Auditory Pathway". To paraphrase: the old fashioned fire bell is easier to locate over the sirens used by emergency vehicles.

Earlier, in 1962, Nordland used an artificial human head with microphones at eardrum position, when he succeeded in measuring the interval time differences for clicks and inter-aural phase differences for tones as a function of azimuth (horizontal).

Also in 1962 Batteau used a model of the human pinna (ear flaps) to confirm the necessity of these physical devices to establish sound directionality. He conducted two experiments whereby a series of speakers were set-up in a circle around a pair of omni-directional microphones spaced approximately seven inches apart. Speaker output was channeled to a person in another room wearing headphones. The person was then asked to identify from which direction the speaker output occured. In this instance the listener found it difficult to localize the sound source, and was often confused. In part two, the microphones were fitted with replicas of human pinnae. The listener then found it much easier to discern not only source, but front and rear localization.

Additionally, it was found that the ridges of the pinna also provide reflections and coloration to sound entering the ear. Sound directly entering the ear is later followed (albeit very quickly) by delays from the ridges. Interestingly, Batteau (and others) found that the delayed portion is less from above than from the sides or below.

THE KUNSTKOPH (Dummy Head)

When Sennheiser began their experimentation in the development of their dummy head recording system they did not actually know how the human sensory apparatus translates impressions into spatial perceptions. However, it was felt that such an investigation should start by reverting to nature's way of hearing sound by placing a "head" of some sort between the microphones. The optimum design for the actual head was accomplished by arriving at an *average* head on which was mounted an *average* set of ears. Testing was conducted using a number of external refinements; wigs, different skin flexibilities and densities, etc. It was found that none of these added or subtracted significantly to the response.

Regaring microphone placement, at one time the Sennheiser engineers tried putting the microphones inside the head, at the ends of simulated ear canals. This idea was discarded when it was found that the sound was abnormally influenced by the ear canal twice: the first time as the sound traveled in through the canal, and the second on playback when the sound had to travel the same path the other way, therefore overemphasizing the part the canal played in the process.

After exhaustive testing Sennheiser found that the best point for microphone placement was 15 mm (5/8") outside the ear canal, where there was minimal variation in response between test persons under recording conditions combined with the best placement for reproducing the sound using "Open-Aire" headphones.

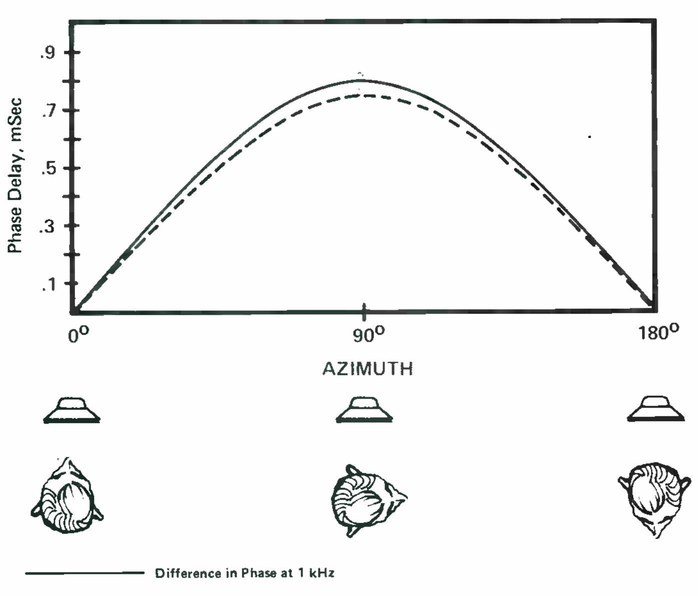
By placing the microphones at the outer ear at the same point where one hears sound the actual hearing experience is very closely approached. By recording and playing back at the outer ear the original sound experience is recreated.

As Sennheiser has developed it, the head is of a special plastic material which conforms to the size and shape of an average head, and wieghs approximately one and a half pounds. The dimensions are approximately 11" high from the base of the neck to the top of the head, 7¾" from the nose to the back of the head, and 7¼" from ear to ear. Consistent with their research, the ears are detailed down to the ear flaps and ridges, the portion of the ear that forms the audio collection area for the two exernal condensor microphones. The system's carrying case is used to mount the head on during operation, and simulates the mass of the upper torso. The upper torso, neck, shoulders and upper chest, have also been found to influence sound perception.

The optimum reproduction of binaurally recorded material is through "Open-Aire" headphones, thus returning the sound in concentrated form to the area from which it originated, in the sense that the sound originated (was heard) for the recording at the microphones (ears).

Reproduction of binaurally recorded

Differences in phase and time as a function of horizontal Azimuth measured with a dummy head (Nordlund, 1962)



Author's note: "Open-Aire" headphones are a product of the Sennheiser Company. Approximation of their effectiveness in listening to binaurally recorded material can be acheived by listening on the usual tightly sealed earphones held slightly away from the ears, although it is recognized that this is both cumbersome and inconvenient.

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material through "Open-Aire" headphones was found to add to the listening dimension by giving one a "free feeling" instead of being sealed off from the outside world. This imitates the natural hearing experience.

Reproduction of binaurally recorded material via speakers will result in a fairly good stereo recording, but the sound will tend to be too directional, and will be lacking in various planes – notablely that of front-to-back and top-to-bottom. This is because the head relies on its position in the recording environment to place the listener in that same spot during playback, therefore producing a *three dimensional* or spatial reproduction with the use of headphones which allow for some ambience, as mentioned above.

The effect is said to be even more pronounced and even more realistic if a person uses his own head with the microphones, instead of the plastic head. Perhaps, this is so because the dummy head can only be a compromise between the various head shapes and masses of various people.

In combination with other sensory inputs, vision, and the like, it should be noted that while listening a person makes unconsious small movements to *zero-in* on the sound being reproduced. With this in mind, tests were conducted by Sennheiser whereby a conversation was recorded on the dummy head and later recorded on a human head. The results: The recording from the human head with the incorporation of the slight head movements in various planes sounded more realistic.

The two microphones of the MKE2002 system are condenser in design, and are mounted in a stethoscope-like frame. If not mounted on the dummy head, they are omni-directional in pickup pattern, and only become directional when mounted on the dummy head or a human head. If a polar pattern were plotted in a mounted position, a mild cardioid pattern would be observed.

Windscreens are also available for recording under adverse conditions. These consist of a foam material and are placed over the microphones to minimize low-frequency rumble or wind effects when outdoors. There is a slight dampening affect at frequencies above 16kHz, but the directional affect is not impaired.

APPLICATIONS

The whole idea of using the dummy head tecnique is somewhat different from standard microphone technique. It has usually been the aim of the recording engineer to bring the studio or concert hall atmosphere into the living room of the listener. The dummy head technique actually does the opposite – it brings the listener into the studio or concert hall.

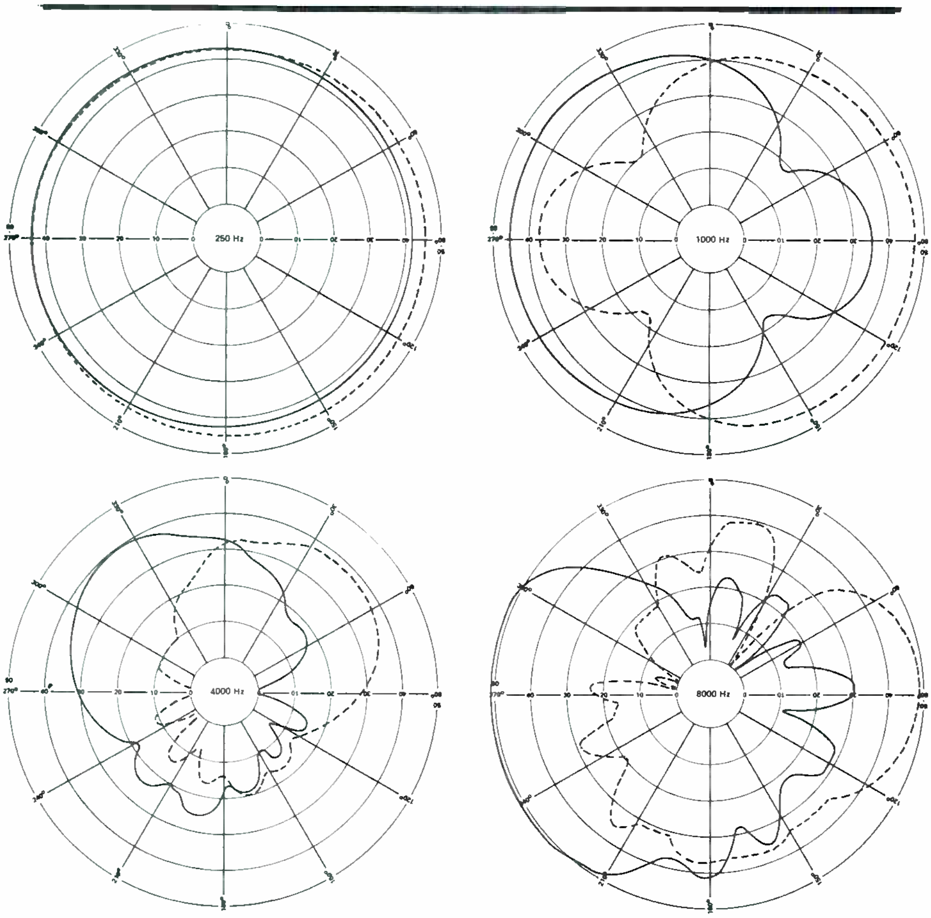
For placement in a recording environment, a person can count on accurate reproduction wherever the dummy head may have been placed. A person in an audience may not always have the opportunity to sit in the center of the presentation – he usually sits on the periphery where he will undoubtedly hear room reflections in combination with direct sound, and this is the experience that will be duplicated on payback. This placement may not be ideal for all listeners especially for use in the recording studio environment where orchestral and rhythm section placement may vary from the concert form of presentation. If one should choose this form, the conductor's area may be recruited to capture optimum balance and an *acoustic snapshot* of the recording.

Experimentation may be desired for those who would like to try dummy head placement on *widespread* instruments to enhance drums and piano. The results of the binaural recording could then be mixed with the rest of the multi-mike set-up to see what type of interesting effects may be obtained. For example, Producer/Engineer Alan Parsons used this technique to record thunderstorm effects on his LP, TALES OF MYSTERY AND IMAGINATION.

The German record company, DELTA ACUSTICS has used the Sennheiser equipment to record a number of dummy head discs which showcase studio music for playback at stands in Airports, Railway Stations, and the like. *

SOUND REINFORCEMENT MONITORING

For the sound reinforcement mixer who sits backstage, or in an isolated sound booth, or out in the alley in a sound truck during a concert and seldom has the chance to hear what the actual presentation sounds like to the audience, dummy head technique may present a solution. By monitoring the head in the center of the recording environment, the engineer now has the chance of becoming a member of the audience and can thus mix accordingly. The opera house in the Kennedy Center, Washington, D.C., is an instance where this has been done.



ARTIFICIAL HEAD POLAR RESPONSE PATTERNS

The four polar plots show the amplitude response patterns for the pick-up microphones at the right and left sides of the Sennheiser artificial head. The ability to differentiate direction solely from relative amplitude responses of the right and left detectors can be seen to increase remarkably with the frequency of the sound source. These curves are only part of the total picture, however. They do not show the relative delay and phase-response phenomenae that are very important in sound source localization.

**Author's Note: While listening to this demo disc on headphones, I found myself smiling and looking out of the corners of my eyes to see if someone was actually standing or walking near me while listening to certain parts of the record. An uncanny realism is evident, which must be heard to be believed.*

ARCHITECTURAL ACOUSTICS

A sound contractor may be responsible for improving the intelligibility of an airport P.A. system. By recording announcements in a *before* condition, making necessary acoustical changes, and then recording and measuring announcements as they are heard in various parts of the acoustical environment in an *after* condition, optimal results may be achieved. All reverberation, extraneous noise, sound sources, and frequency response may be evaluated vis-a-vis human hearing through this technique.

DRAMA

The Canadian Broadcasting Corporation (CBC) has staged a number of experimental radio dramas incorporating dummy head technique into production. In a standard radio play, actors would be situated around the studio reading their scripts. In the dummy head production, the drama would actually have to be acted out — the people would actually have to move about the studio while acting out the script in order to convey the motion and direction of all sound planes. Picture, if you will, the murderer sneaking up on his intended victim from behind, footsteps coming closer, closer, until he makes his move — directly from behind you! In essence, you are there! You become the victim! This effect, along with the other musical and sound effects of this scene, leads to a heightened sense of realism, drama, and audience involvement.

AUDIO/VISUAL

As was perhaps hinted at in the example cited earlier, of the German record company, dummy head recording seems perfect for those audio/visual or just audio presentations frequently seen at conventions, shows and in public demonstration areas, where those interested are invited to pick up a set of earphones to listen to some sort of program. In these cases the full potential of binaural recording can be controlled from beginning to end — the playback being controlled through the use of the optimum "Open-Aire" headphones.

HEARING AIDS

On the West Coast, there have been reports of experimentation using two tie-clip omni-directional microphones attached to a stethoscope-like headphone and a small stereo amplifier. This device has aided the hard of hearing, not only by amplifying the sound and increasing intelligibility, but also by reinstating sense of direction. This effect also serves as an early warning device for impending peril for the hard of hearing. If the person were walking down a street and suddenly hears a car coming from the left side, he can respond and take action accordingly. Normally the person would not receive that directional information as easily from monaural hearing aids.

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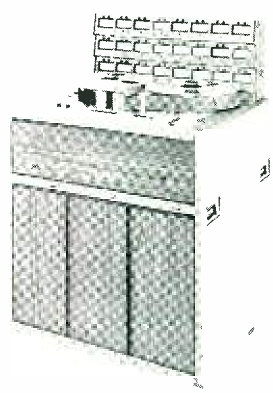

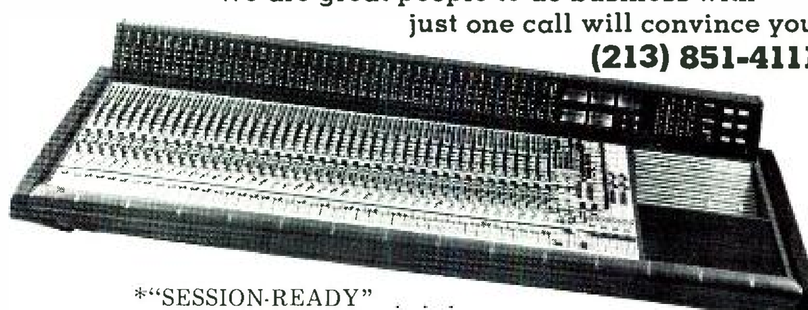

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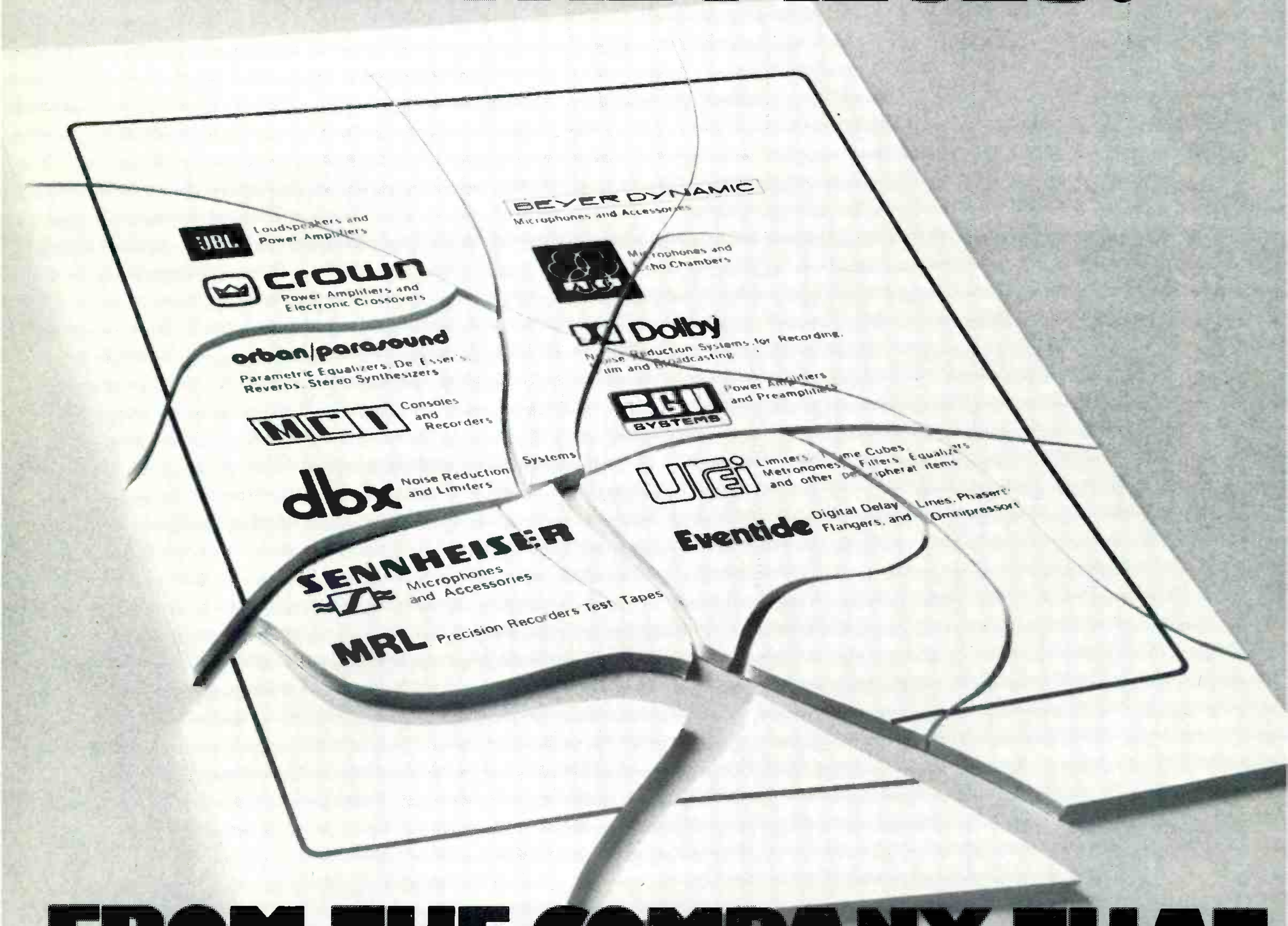

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. . . continuing the series of reports on automated mixing philosophy and hardware . . .

MCI's *'helping hands'* Automated Mix-Down System

MCI is announcing the release of its JH-50 Automation system this month. The system was designed primarily to fit into their JH-500 Series Mixing Desks. This series of console has been advertised as "Automation-ready", having all the control switches built into the panels, and Voltage Controlled Amplifiers already installed for Channel and Echo Return Faders. The Mute system is also designed to be controlled easily by automation. The following article presents the basic concepts of design and operation of the automation system.

Design Philosophies

Three major hardware decisions must be made before any console automation system can be designed. These three decisions are interdependent. To a very large degree, they depend on (1) how many functions are to be automated in the maximum system, (2) how much complexity is acceptable in the controls and operation of the system, and (3) the total, installed price of the system.

A survey of current studio requirements has shown that about 24 to 40 automated functions are needed to implement modern mix-down techniques. It was found that most studios can not economically justify a larger system since initial cost, operating costs, and complexity of controls increase rapidly with a larger number of automated functions. It has also been found that control of the system must be as simple as possible, since it is imperative that the system be operated by Producers and Mixers who are not necessarily technically oriented and may never have used the equipment before.

The three hardware decisions are:

1. THE STORAGE MEDIUM

Two mediums are in current usage, (a) the Audio Master tape, or (b) separate digital storage system locked together with the Master tape by a SMPTE code device (or equivalent).

Choice of the storage medium depends principally on the number of functions to be automated and the relative costs of the two systems. A separate digital storage device with synchronization equipment usually adds from $3,000 to $5,000 to the total cost of the system.

MCI has found that the Audio Master Tape (multi-track master) is a satisfactory medium for storing data in a system designed for automating level functions, and that it has the advantage of automatic synchronization, no additional storage problems, and considerable savings in both initial cost and operating costs.

Thus, in the MCI system two tracks of the multitrack Master tape are used for storing automation data, successive updates of the mix being "ping-ponged" from one of these tracks to the other. The data rate is 9.6 baud (about a 10 kHz bandwidth). A recording level of about -10dB furnishes a satisfactory signal to the automation system, and will not bleed through to adjacent audio tracks.

2. THE RECORDING METHOD

Two methods are in current usage: (a) Signal digitalization and "data packing", or (b) Sequential scanning. Obviously, in a system containing a very large number of automated functions, "data packing" would be necessary. However, "data packing" has several inherent drawbacks, not the least of which is the loss of a great deal of reliability and usually has a limiting factor in the number of "updates" it can recognize within a given time frame. A tape drop-out can cause a major error during a mix, since a significant piece of data not repeated would be lost.

If sequential scanning can be made at a rate close to the limits of human perception, a tape drop-out will not be noticed, since the loss of a single data word can be ignored until the next scan. A drop-out would cause an error of less than one-tenth of a second in a progressive level adjustment. It has been found that sequential scanning and recording of all data can be easily accomplished in a moderate sized system at a rate which takes advantage of this gain in reliability.

Automation systems which use data packing (and usually have insufficient storage) often overload when it is necessary to make a large number of changes at the same instant. The system MCI has chosen – sequential scanning and a recording of data from all functions on every scan – can never overload, since it is always working at the worst case data rate.

3. THE UPDATE METHOD

A critical factor in designing an automation system which has few restrictions, and can be adapted to almost any use, is the "Bounce Delay". This is the time required to read a VCA value off the tape, process it, and re-record it on an Update track. The methods used divide roughly into systems using Analog Updating and those using Digital Updating.

In the Analog Updating method, two full scans are required to complete an Update. During the first scan, the VCA value is read off the tape, converted to an analog signal and sent to the console. The analog circuits in the console process the update by adding the READ value and the FADER value, thus producing a new analog voltage. On the second scan, the analog value is picked up, coverted back to a digital signal and recorded. The

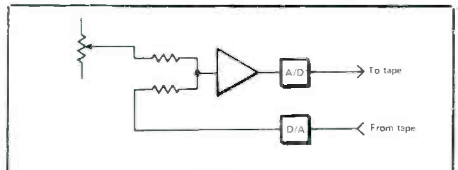


FIGURE 1

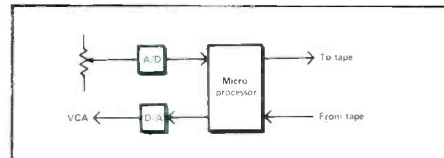


FIGURE 2

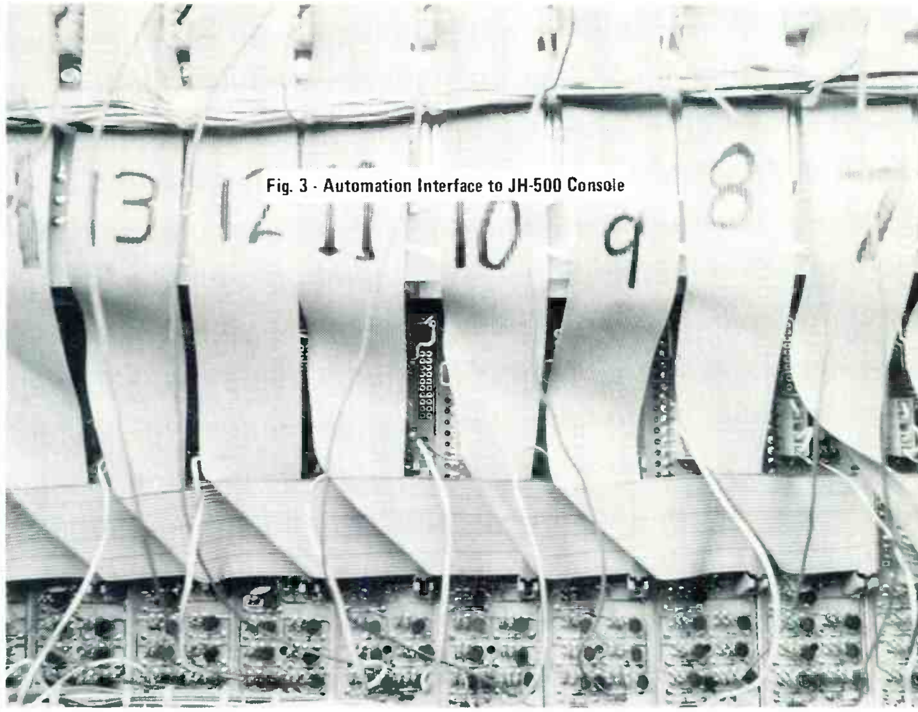


Fig. 3 - Automation Interface to JH-500 Console

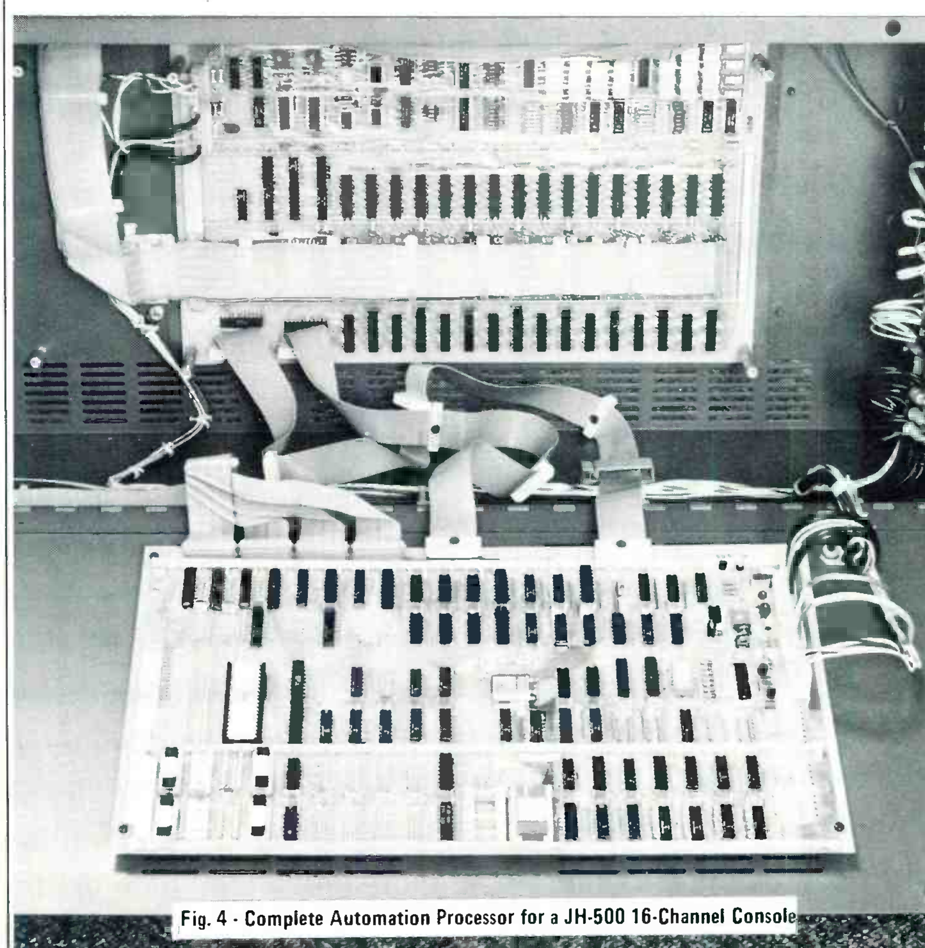


Fig. 4 - Complete Automation Processor for a JH-500 16-Channel Console

Bounce Delay is variable, depending on the number of VCA's being accessed during each scan. Also, the accuracy of the system is severly limited since the digital data is being processed by analog circuitry. New inaccuracies are added to the signal each time an update is recorded. (See Figure 1.)

The Analog Update method has delay times of several milliseconds which add to the total delay each time an update is made. This results in a significant delay between the Master Audio tape and the Automation when several updates have been made. Because of the inherent inaccuracies of the system, automation systems using the Analog Update system frequently do not attempt to achieve an eight-bit (256 step) resolution.

By comparison, the Digital Updating method is completed in a single scan. The VCA value is read off the tape, digitally processed, and is re-recorded during the same scan. The total "Bounce Time" is the process time required to read the data, apply a mathematical formula, and write the resultant. (See Figure 2.)

The "Bounce Time" in the Digital Update system used by MCI is about 1.2 milliseconds (.0012 seconds). Thus, if it is necessary to use as many as 100 updates at some point on a mix, the total delay will be only about 1/8 second. Since the value read off the tape is not processed by analog circuitry, a full 256 step, eight-bit resolution is easily achieved.

Figure 3 shows the plug-in interface boards which convert a standard JH-500 Series Console for full Automation. Figure 4 shows the complete hardware package for 16 channel Automation.

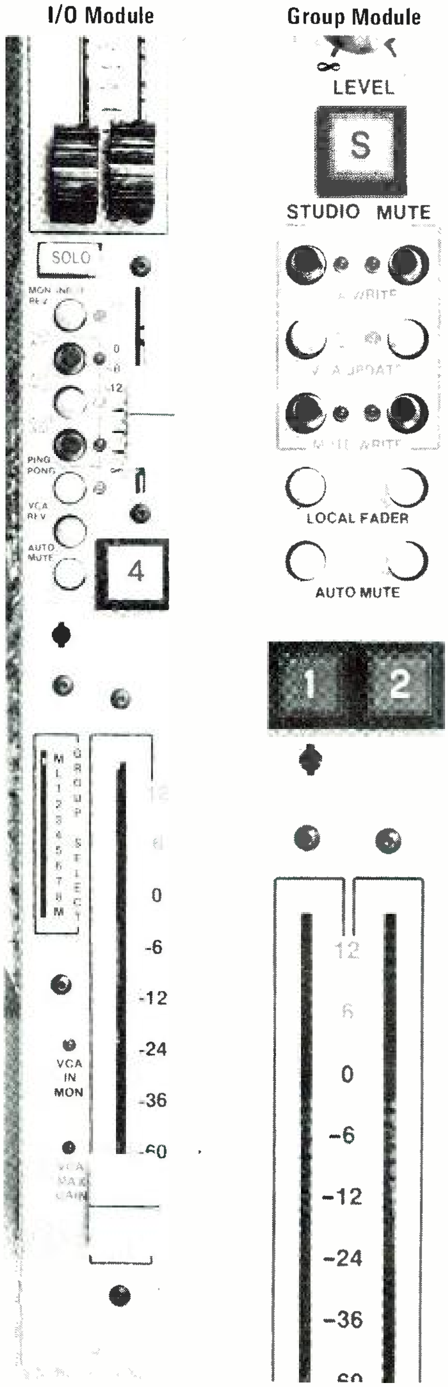
Operational Criteria

Reliability, flexibility, and accuracy of reproduction are all very important attributes of the final automation system, but they are worth very little if the man/machine interface was not well designed. When an automated mix-down system is being used, it is absolutely necessary that all complications of circuitry or sequencing be completely transparent to the user. The operator, engineer, or producer, must be able to control the system without taking his mind away from his primary function – *creating the best mix that he is capable of producing.* If he must focus his attention on HOW to get the effect he wants, he will often lose sight of WHAT he wants to accomplish.

MCI has made a significant reduction in the mechanics of operating automation controls by introducing *Automatic Nulling.* This feature allows the operator to change from one mode of operattion to another without having to null his faders. When the operator presses the button which signals his intention to change the mode of operation, a temporary mode is introduced, which automatically maintains the previous mode until the new mode can be initiated without a step change in levels.

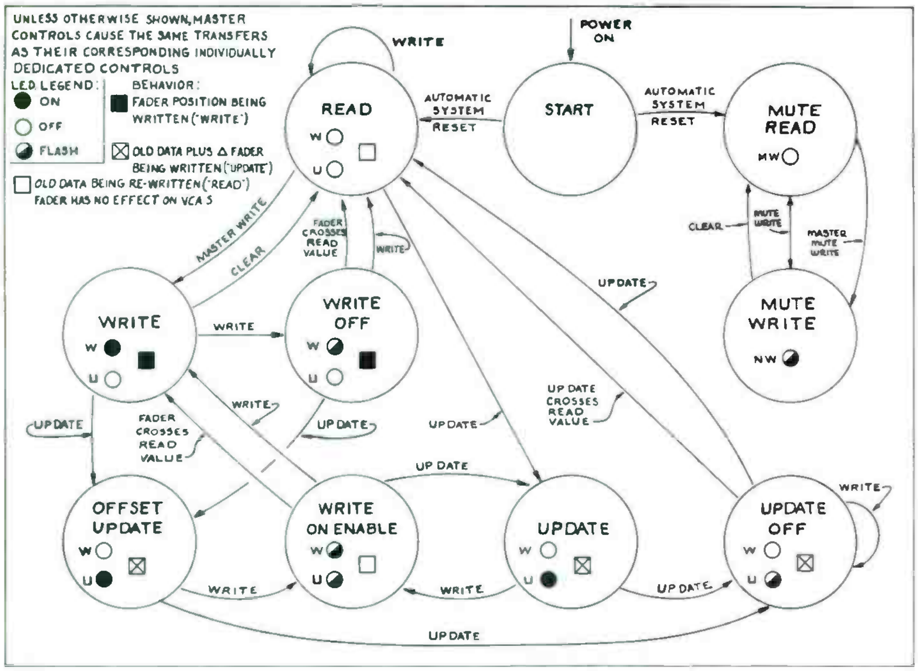
Figure 5 shows the simplicity of the controls. There are just three buttons and three LED's on each I/O module, on each Echo Return, and on each Group control. A similar set of buttons are used for the Master Controls. Each set of buttons controls the same set of functions for its module or group.

Figure 5



There are four stable modes of operation which are controlled by pressing the proper button. The LED's indicate the mode which has been initiated. Intermediate (or temporary) modes are indicated by blinking LED's and are provided to allow the Automatic Nulling circuits to work. (See the State Mode Chart, Fig. 4.)

FIGURE – 6



1. **WRITE MODE**

When Automation is initiated, READ mode (a stable state) is in operation. This is shown by all LED's being OFF. To record an initial automation track, press the MASTER VCA WRITE button. All the VCA WRITE LED's will come ON, indicating that WRITE mode has been initiated. While the recording is playing, adjust the controls to produce as good a mix as possible When the writing of the initial automation track is complete, press the CLEAR button and the complete system will return to the READ mode.

2. **UPDATE MODE**

This mode is used to modify the original mix. The original mix is played back and the sum of the original mix plus new FADER movements are recorded on a new track. When the point in the playback which must be updated is reached, press the UPDATE button and move the appropriate FADER to produce the effect desired. The appropriate UPDATE LED will light to show that the module (or Group, or complete console) is in UPDATE mode.

When the playback reaches a point which no longer needs updating, press the UPDATE button again and the UPDATE LED will start to flash, showing that the temporary state of UPDATE OFF has been initiated. When the FADER crosses the READ value on the original automation track, the LED will go OUT, showing that the system is back into READ MODE. (The fader continues to control the level until the LED goes out, so the operator can produce a pleasing blend from the update back to the READ value. At that point, the FADER loses control.

3. **MUTE WRITE MODE**

This state may be initiated by pressing the MUTE WRITE button. The MUTE WRITE LED flashes to show this state. You may write the MUTE functions into the tape either simultaneously with the original WRITE mode or at any later UPDATE of the FADER functions. This is a stable state, completely independent of the WRITE and UPDATE functions. MUTE WRITE can be cancelled by pressing either the MASTER CLEAR button, or by pressing the MUTE WRITE button a SECOND time.

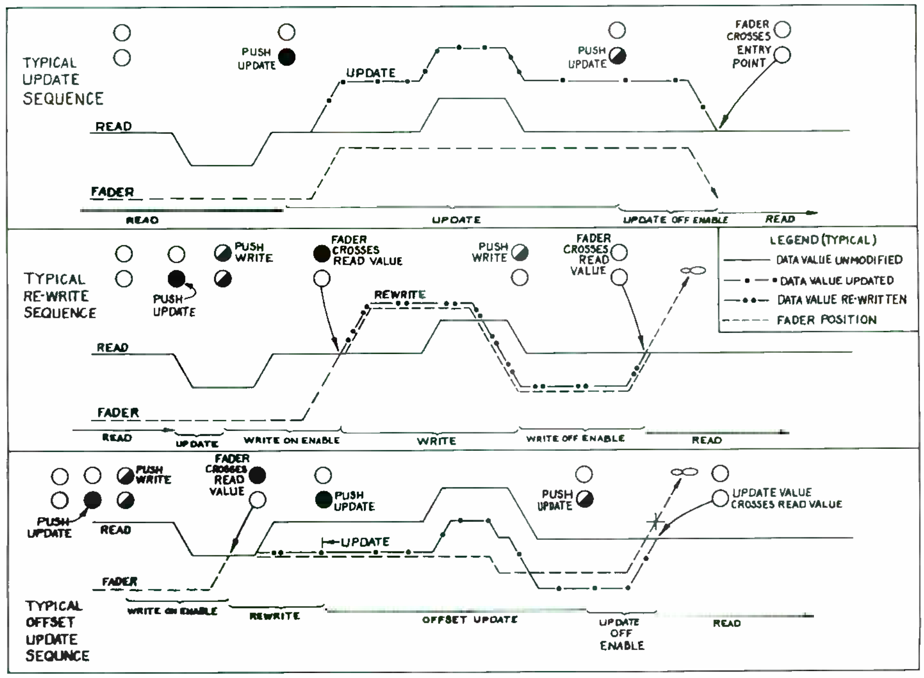
4. REWRITE MODE

This mode is used when the initial track is so badly written that a complete rewrite is less trouble than an UPDATE. This mode is reached through UPDATE mode. From the READ mode, the appropriate UPDATE button must be pressed, followed by pressing the WRITE button. As soon as the write button is pressed, both the UPDATE and the WRITE LED's will start to flash indicating that the temporary state of WRITE ON ENABLE has been initiated. When the Fader crosses the current READ value, the UPDATE LED goes OFF, and the WRITE LED goes ON to indicate that WRITE MODE has started. The appropriate fader now has complete control of what is being written on the new data track. The original READ value is NOT summed into the new track – as it was in the UPDATE MODE.

When leaving REWRITE, press the WRITE button again, and the system will shift into the temporary WRITE OFF mode. When the Fader crosses the original read value, the system will shift into READ mode. This insures a smooth transition to and from WRITE mode.

5. OTHER MODES

FIGURE – 7



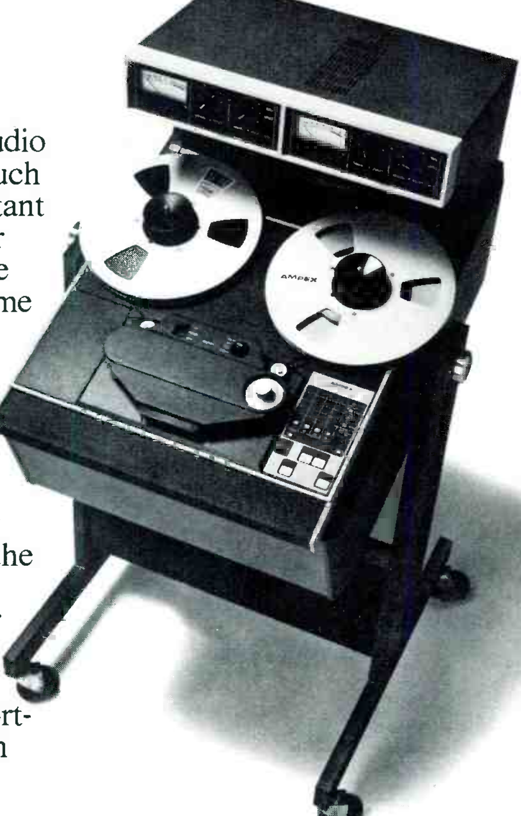
Other states, using combinations of the WRITE and the UPDATE buttons are possible, as shown on the STATE MODE CHART. Also there additional ways of moving from one stable state to another shown on this chart.

Figure 7 shows a chart of a typical UPDATE sequence and a typical REWRITE sequence. These charts show the required action to initiate each step of the sequence and the condition of the LED's across the top of the chart. Solid lines show the action of the appropriate FADER, and dot-dash lines show the resultant UPDATE (or REWRITE) value recorded on the new track of automation data.

Systems will be shown at the AES Show in Paris, and delivery is scheduled to begin in early March.

You're selling time, but they're buying sound.

ATR-100 is the sound buy.



You'll probably buy your ATR-100 because no other audio machine in the world offers such amazing fidelity. Every important performance specification for the ATR-100 is better than the competition provides, and some parameters are a full *order of magnitude* better.

But after you get used to your ATR-100, you'll discover a mechanical feature or two that you've never seen before. Like dynamic braking that stops tape safely even if the power is off. And a "smart" transport that waits for proper tension before moving the tape. And a remote control that fits in your hand like a portable calculator, complete with LED status indicators.

Finding edit points on a new Ampex ATR-100 is a two-finger pleasure. Twirl the knob on the capstan, and servo motors move both tape reels. You can rock back and forth over a note, syllable or sneeze as easily as pointing your finger.

It's been a long time since you've seen this sort of claim, but here it is, in writing: ATR-100 is the world's best audio recorder. It was designed for studios that can't take chances.

AMPEX

Complete technical and performance specifications are available in a free brochure. Write us at 401 Broadway, Redwood City, California 94063, or call (415) 367-2011.

For additional information circle No. 50

New Products

LEXICON ANNOUNCES NEW LOW COST AUDIO DELAY FOR SMALL INSTALLATIONS

Lexicon, Inc., manufacturer of Delta-T audio delay systems, has announced a new, low cost digital time delay system specifically configured for sound reinforcement applications in small school auditoriums, churches, theaters, clubs, etc. Dynamic range is better than 90 dB; noise and distortion are less than 0.1%. The new Delta-T Model 92 has a complete system price well under $2,000 yet offers the advanced electronics and major features of higher priced, large system Delta-T units. According to Lexicon the Model 92 answers an industry need for a high quality, high reliability audio delay system for the smaller, lower budget sound reinforcement installation. Due to its excellent audio qualities and ultra low distortion, the Model 92 also is expected to find wide use in studios and live performance applications as well.

Model 92 provides two adjustable audio signal delays of up to 120 ms, each controlled by a single front panel knob. As standard features, the system also includes audio input and output transformers, an automatic, fail-safe, audio bypass feature, silent power up/power down circuitry, and rear mounted XLR-3 type audio connectors. A five-position LED headroom indicator calibrated in 10 dB increments below limiting simplifies level adjustment and verification. All units have universal power compatibility, 115/230, 50/60 Hz, and have international connectors and detachable power cords. Input and output levels are adjustable through the front panel.

The Delta 92 has been designed for rack mounting and requires only 3½ inches of panel space. It employs Lexicon's proprietary floating point digital encoding techniques to achieve unmatched audio quality and dynamic range. The unit has plug-in modules for its memory and audio subsystems to simplify field maintenance. Its construction is all solid state employing MOS-RAM memory and low power Schottky IC logic for low power, high reliability operation. The compact Delta-T 92, weighing only eight and one-half pounds, can be installed permanently or easily transported from one location to another.

Specifications

Dynamic range: 95 dB typical.

THD and noise: Less than 0.1% at 1 kHz reference limit level.

Frequency response: 20 Hz to 12 kHz, +1, -2 dB measured 14 dB below limit level.

Delay capacity: 120 ms per output.

Delay outputs: Two outputs are standard, each independently adjustbale.

Delay selection: 7.5 ms steps via front panel selector knob.

Delay accuracy: ±0.1% of setting plus 0.17 ms long term.

Input characteristics: Balanced floating transformer coupled, 10 K ohm input impedance min.; adjustable from +8 to +18 dBm for reference limit level.

Output characteristics: Balanced floating transformer coupled, source impedance 100 ohms max.; adjustable from +8 to +18 dBm for reference limit level.

LEXICON, INC., 60 TURNER STREET
WALTHAM, MASSACHUSETTS 02154
PHONE: (617) 891-6790

For additional information circle No. 51

INTERFACE TO INTRODUCE TWO NEW MIXERS

The Model 316 mixer is intended for high performance low cost 16 track recording. This mixer has features similar to the 104/108 series except that it is 16 track and uses a six inch conductive plastic attenuator instead of four inch, and modules contain a mute button. Main-

frames for 16, 24, or 32 inputs are available, with many options.

The 104S mixer is intended primarily for low cost high performance broadcast applications and has modules with two stereo line level inputs and one mike input, mix bus switching, equalizing, pan pot, echo send, and conductive plastic slider attenuator with cue/preview switch at the bottom of the slider travel; program switcher modules switch the main and monitor outputs to any of the four internal stereo mix busses. Outputs are transformer coupled.

INTERFACE ELECTRONICS, 3810 WESTHEIMER, HOUSTON, TX. 77027
PHONE: (713) 626-1190

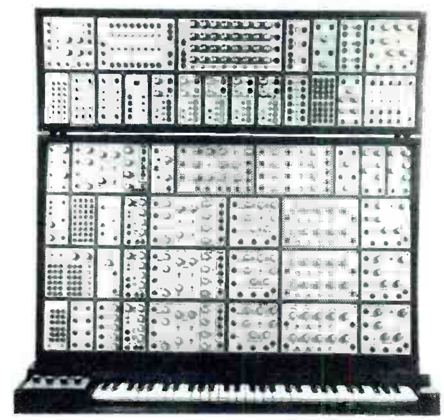
For additional information circle No. 53

MODULAR MUSIC SYNTHESIZER DISSECTED IN E-MU MANUAL

A new publication from E-mu Systems, Santa Clara, California, dissects the intricacies and sound synthesizing potential of modular electronic music synthesizers.

The publication was written as a catalog/manual describing the various modules and possible uses of the E-mu Modular Music Synthesizer. However, E-mu offers such a wide range of module types (37) as well as an almost infinite number of possible configurations, that the publication amounts to a thorough textbook on the subject of electronic music synthesis.

Among the many features discussed in the 90-page publication are the following: user-designated "pre-patch", a method of inter-connecting modules while keeping the control panel uncluttered; polyphonic keyboards, capable of controlling several synthesizer channels or "voices"; polyphonic sequencers, capable of producing sophisticated counterpoint; and dual transient generators, which offer precise voltage control of the attack and decay of individual notes or sounds. Fully modular keyboard systems are available for as little as $3,000.



E-mu feels this 90-page manual will be of interest to educators, students of electronic music, performing musicians, recording and broadcast engineers, and advanced hobbyists.

For further information, or to order the manual ($5.00), contact the manufacturer directly:

E-MU SYSTEMS, 3046 SCOTT BLVD., SANTA CLARA, CA. 95050
PHONE: (408) 241-0788

For additional information circle No. 55

WHITE MODEL 150 OCTAVE BAND ANALYZER

The Model 150 Octave Band Analyzer is a new precision, battery operated, hand-held instrument, that incorporates triple-tuned filters to meet the ANSI 1.11, Class II specifications. Level in each of ten ISO octave bands centered from 31.5 Hz to 16 kHz is displayed on a 10 x 14 LED matrix. Display ranges are 14 dB and 28 dB for a resolution of 1 dB or 2 dB. Acoustic sensitivity ranges are calibrated from 34 dBspl to 110 dBspl. Either flat or A-weighted measurements may be made. A precision microphone which may be detached for remote measurements is supplied with each instrument.

The Model 150 is shipped with a Model 151 Pink Noise Source, a battery charger and a carrying case. The Model 150 will

For additional information circle No. 54

the performance-proved HARMONIZER –
Now . . . HUNDREDS in use at MAJOR STUDIOS
and with TOP CONCERT PERFORMERS
THROUGHOUT the WORLD!

THE INDISPENSABLE HARMONIZER

The Eventide Harmonizer is the most versatile special effects instrument ever packaged in a single unit. Not only is it a full-fledged digital delay line, a pitch changer with a two octave range, and an anti-feedback unit which allows boosting of sound levels, but it can also be used to speed up or slow down tapes. Although it has only become available recently, it has already produced some of the wildest effects on record.

The possibilities are endless. Its digital circuitry and Random Access Memories (RAM's) transpose input signals anywhere up to an octave above or below the original, while preserving harmonic ratios (and thus musical values). As a low-cost, versatile delay line, the Harmonizer is used for "doubling" (or tripling) vocals, for delay equalization in sound reinforcement, and for many types of reverb/echo special effects. An optional phase-locked Keyboard allows the performing musician to construct harmonies in real time. In addition, through the combination of feedback, delay, and pitch change, the Harmonizer permits previously unavailable special effects. Only personal experience and experimentation will show you the full range of its capabilities.

Harmonizer H910:	. . . . $1500.00
Readout option 02:	. . . . $125.00
2nd output option 04	. . . . $240.00
Keyboard (mono)	. . . . $500.00
Keyboard (polyphonic)	. . . . . . $600.00

Write for spec sheet and name of nearest distributor.

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For additional information circle No. 56

operate about five hours between charges, while the Model 151 will operate at either line or microphone level for about 30 hours from nine volt transistor batteries.

Uses for the Model 150 Octave Band Analyzer include sound system set-up, speaker placement, speaker check-out, horn aiming, octave equalization of hi-fi systems and noise surveys.

The Model 150 will be available in April of 1977. The suggested list price is approximately $1,400.00.

WHITE INSTRUMENTS, INC.
P.O. BOX 698, AUSTIN, TX. 78767
PHONE: (512) 892-0752

For additional information circle No. 58

NEW ELECTRONIC MUSIC LABS ADAPTER MAKES A SYNTHESIZER POLYPHONIC

A new polyphonic synthesizer adapter has been introduced by Electronic Music Laboratories, Inc., of Vernon, Connecticut.

The EML POLY-BOX is a versatile one octave keyboard which gives a monophonic synthesizer a unique polyphonic capability. Digital memory allows the programming of any chord or pitch combination into the POLY-BOX which can then be played through a synthesizer. Pitches follow the synthesizer oscillator through portamento, vibrato and keyboard transpositions and can be produced from one octave above to three octaves below the synthesizer oscillator. Manual tuning, variable phasing and a low pass filter help make it possible to play bass lines or harmony on the POLY-BOX keyboard while playing melody on the synthesizer.

The POLY-BOX is a durable, lightweight instrument using digital and analog technology. It weighs 15 pounds and is 19" wide by 12" deep by 5" tall. Output is adjustable from 0 to 5 volts nominal and the frequency range is 5.5 Hz to 8342 Hz. There are up to 26 pitches at semitone intervals simultaneously available.

The EML POLY-BOX can be added directly to any patchable synthesizer or to smaller instruments with minor modifications and is available directly from the factory and from selected dealers.

The POLY-BOX sells for $475.00.

ELECTRONIC MUSIC LABS, INC.
P. O. BOX H, VERNON, CONN. 06066
PHONE: (203) 875-0751

For additional information circle No. 59

NEW FULL TRACK MX–5050 RECORDER FROM OTARI

A new full- track version of Otari's MX5050 series professional recorder has been announced which provides full-track single-channel record and reproduce capability, plus half-track two-channel reproduce capability as well.

The new machine incorporates all the professional features of the other MX-5050 models, including: front panel edit and cue controls, motion sensing, precision mounted splicing block on flip-up head cover, professional XLR connectors for line-in and 600 ohm balanced line outputs (at +4 dBm or -10 dBm), front adjustable bias and equalization, built-in test and cue oscillator.

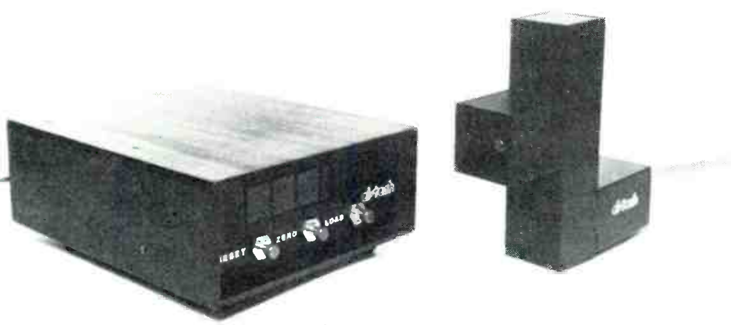
Price is $1,450. Options include rack mount kit, portable case, and remote control unit.

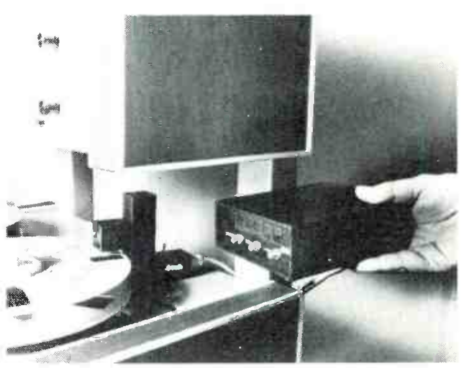
OTARI CORPORATION, 981 INDUSTRIAL ROAD, SAN CARLOS, CA 94070
(415) 593-1648

For additional information circle No. 60

THE EL-TECH "TAKE FINDER"

This newly announced product is a two piece sensor and read out device which numerically locates program material on tape machines. By counting each revolutio of either the supply or take-up reel, a number is indicated to the operator from a LED read-out for all locations on the tape. The read-out can count up to 99,999 counts, and is housed in a small (2"x5"x6") box which can be located up to 25 feet from the tape machine. To pick up tape reel rotation, an opto-electrical sensor is placed near either the tape machine take-up or supply reel. A small





For additional information circle No. 61

light source illuminates the tape reel and photo transistors sense the light being reflected. Small pieces of black tape are affixed to the tape reel to interrupt the reflected light reaching the sensor. By sensing the direction of movement of the black tape, the sensor determines the direction of the rotation and signals the display box to count either up or down. The counter has a capacity of 99,999 counts which allows many pieces of tape to be counted per revolution and consequently, for greater location accuracy.

A built in memory allows one location number to be stored so the operator can return to the desired location. When the displayed number equals the memory number, a contact output is available to be used to stop the tape machine. The sensor can be adjusted for different tape widths so the Take Finder can be easily moved from machine to machine. Since the sensor operates on reflected light, no electrical or mechanical connections are necessary on the machine.

The front panel switches reset the counter to zero, load zero into the memory, and load the displayed number into memory. A rear connector allows these functions to be remotely located from the counter. In a recording studio application, this would allow an engineer to place these controls near his tape remote and locate the counter in the console meter panel.

Since the sensor will function on most any machine which has reels, applications on video, audio and film machines are possible. The present sensor is designed primarily for horizontally mounted studio machines, but a small sensor for smaller machines will be available within 2 months.

Price: $349.95

EL-TECH, P.O. Box 23108, NASHVILLE, TN 37202 (615) 256-1650

For additional information circle No. 62

YAMAHA INTRODUCES THE F-1030 FREQUENCY DIVIDING NETWORK

The newly announced product is a fully professional electronic crossover, switchable for two-way or three-way operation. By splitting the main audio feed from the mixer, the F-1030 will drive the power amplifiers for a biamplified or a triamplified speaker system. The F-1030 is said to be ideal for use in portable concert reinforcement systems, studio monitor systems, discotheques, commercial quality, performance and versatility.

Crossover frequencies are selectable (250 Hz, 500 Hz, 800 Hz, 1 kHz, 1.2 kHz, 1.5 kHz, 2 kHz, 2.5 kHz, 5 kHz, 6 kHz, 7 kHz, and 8 kHz). To further complement the characteristics of many different speakers, the crossover slope rates can be switched to 12 dB/octave or 18 dB/octave. A switchable 40 Hz high pass filter protects low frequency speakers from sub-audio transients.

The user can select from a pair of balanced XLR or unbalanced phone jack inputs, with each pair wired in parallel for "chaining" to additional crossovers or other devices. The outputs also come to XLR connectors and phone jacks, and the XLR's have "polarity reversing" switches to facilitate acoustic phase matching. The paralleled input connections and polarity-reversable outputs simplify wiring by eliminating the need for special adapter cables. The input attenuator and three output attenuators are all stepped and dB-calibrated for easy and precise level adjustments.

The input is high impedance, and will accept signal levels up to +30 dB (24.5 volts). The output will produce up to +24 dBm (12.3 volts) signal level in a 600-ohm load. Three LED's indicate when any of the output levels reach +14 dB. There is 10 dB of headroom left above the LED turn-on point (to +24 dB). Maximum gain is +6 dB. The frequency response of the combined outputs is flat (+0.5 dB, -1.5 dB) from 20 Hz to 20 kHz. T.H.D. is less than 0.05% at +24 dB (12.3 volts) output, from 30 Hz to 20 kHz. Hum and noise are -76 dB below maximum output.

The F-1030 comes with a comprehensive operator's manual detailing its specifications, operation and technical applications. The manual is available separately for a retail price of $4.00.

**YAMAHA INTERNATIONAL CORP.
MUSICAL INSTRUMENT DIVISION
6600 ORANGETHORPE AVENUE
BEUNA PARK, CA 90620**

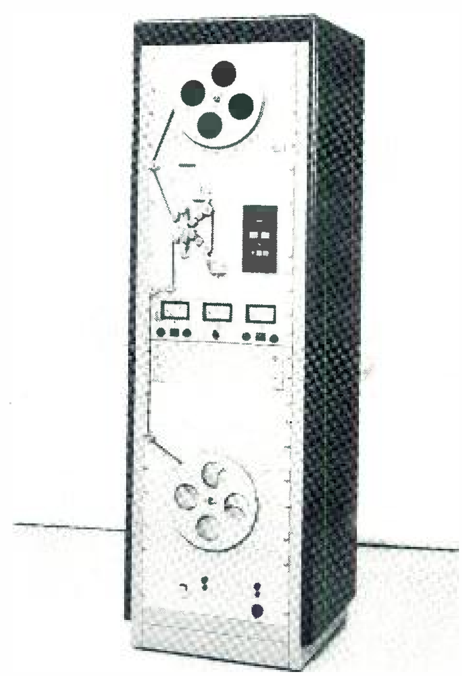
For additional information circle No. 63

LOW COST PROFESSIONAL SUPER 8 RECORDER

Wide Range Electronics Corporation has just introduced a new low cost super 8 recorder for the professional user that fulfills the requirement for magnetic sound recording where high volume production is not necessary but high quality performance is a must.

WRE Model 840 Super 8 Recorder meets all SMPTE standards for S-8 magnetic sound recording and is available in models to accommodate all fullcoat and pre-striped S-8 film formats. This new system is based on the field proven quality performance of the famous WRE Model 800 High Speed Super 8 Transfer Console and the design simplicity and flexibility of the WRE 880 Super 8 Playback Inspection System.

Like the WRE High Speed Transfer Console, the Model 840 utilizes the "on the drum" technique as the most reliable means of recording S-8 film. Head to film intimacy is maintained under virtually all film conditions. The dual drum, dual sprocket film transport design provides a simple and easy film threading path.



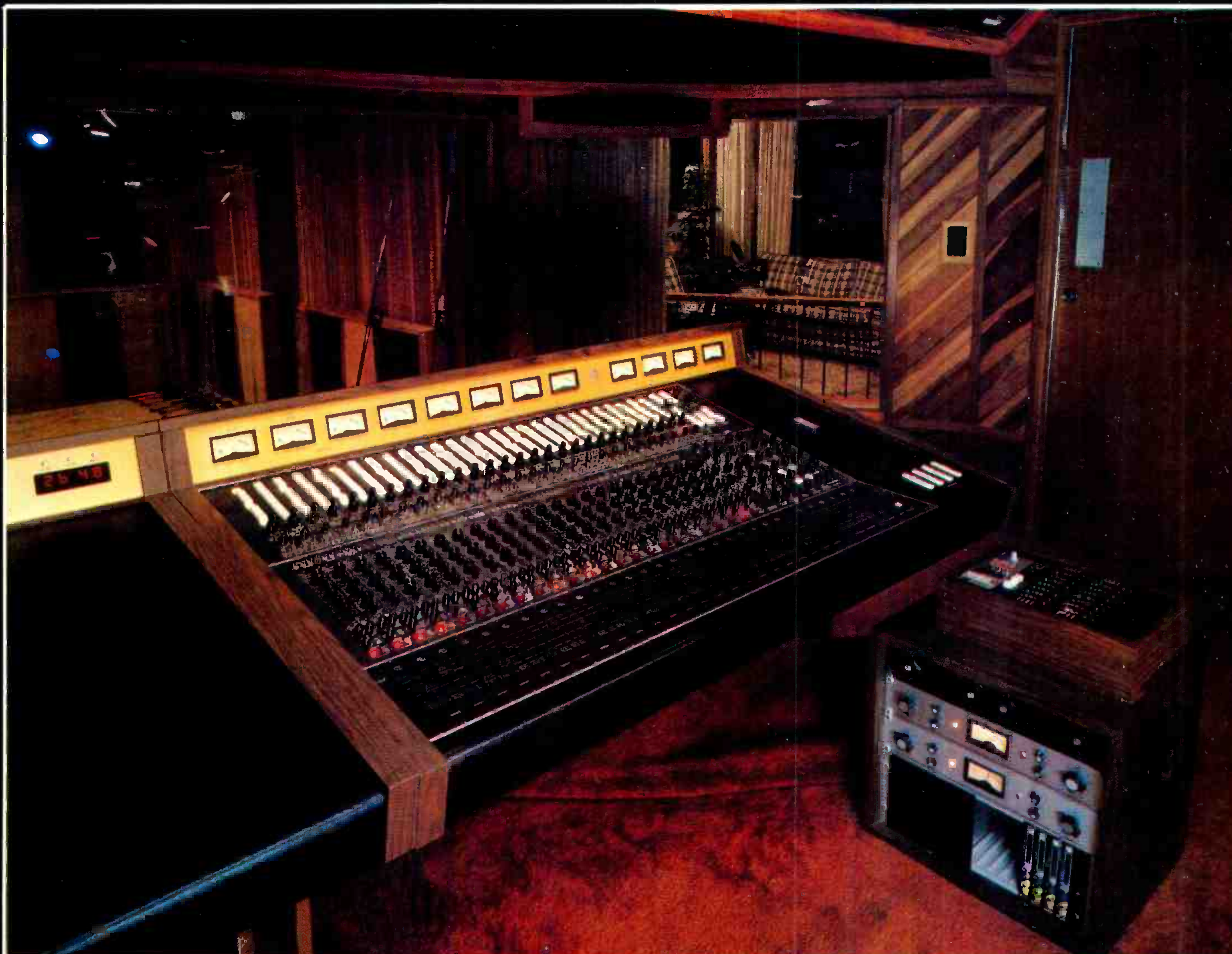
Frequency response is typically 50 to 8 kHz (±2 dB) for record and playback. Maximum wow and flutter is 0.10% rms and total harmonic distortion is less than 1.5%. Film capacity is 3,000 feet. Record and playback operation is at 24 frames per second (20 feet per minute).

Standard Model 840, Type B, Super 8 Recorder will accommodate 16mm (1–3) and single strand S-8 film and is compatible with existing tape playback devices and dubbers in or out of interlock. The Type B system includes two channels each of record and playback electronics. VU meters for record/playback and bias current level monitoring, footage counter and attractive rear lighted push button control panel to accommodate all functions. Models for other film formats are also available. The price for the Model 840, Type B, is $6,750.00 and delivery is 30 to 60 days.

Optional accessories and features available are: interlock and sync/inter-

For additional information circle No. 64

Another incomparable SPECTRA SONICS Model 1024-24 Audio Control Console at Chateau Recorders. North Hollywood, California.



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

Capability: **SPECTRA SONICS** audio control consoles provide an immediate initial capability that may be increased to 32 inputs and 32 outputs, at minimum cost. The flexibility of the system will provide line/microphone selection, attenuation, equalization and, through assignment controls, various other combinations for the most sophisticated signal processing now required in today's studio.

Reliability: **SPECTRA SONICS** audio control consoles have an established reputation of superior reliability. Through creative design, the circuitry is developed to function well below operating limits to enhance an extended life for the components. Through empirical data on **SPECTRA SONICS** audio amplifiers, a reliability rate of 99.9% has been derived. These amplifiers are used in **SPECTRA SONICS** audio control consoles and materially contribute to system reliability.

Performance: **SPECTRA SONICS** audio control consoles are guaranteed to outperform any other console in the world in noise, frequency response, distortion, and peak overload. All consoles are provided with documented data acquired in tests of the complete system. Guaranteed performance specifications are: Frequency Response, ± ¼dB 20HZ-20kHZ; Signal/Noise Ratio (microphone input), not less than 82.5dB below +4dBM, output for a -50 input (50 ohms source); Signal/Noise Ratio (line input), not less than 87dB below +4dBM output for +4dBM input; Harmonic Distortion, less than .01% at +18dBM (1kHz); Intermodulation Distortion, less than .02% at +4dBM; Crosstalk, not less than 60dB at 20kHZ (typically 80dB).

770 Wall Avenue
Ogden, Utah 84404
(801) 392-7531

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Hollywood, California 90028
(213) 461-4321

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For additional information circle No. 67

OPAMP MODEL 2008-RS RECORDING STUDIO CONSOLE

The newly introduced unit is described as a 20 input, 8 output, 4 echo buss, 16 track mixdown-monitor system.

Input channels consist of mix slide pot, film type, with 90 dB of attenuation. Input select: 0 dB, -10 dB, -20 dB, -30 and Mic. Level. Line 1 (transformer input), Line 2 (single ended input), and Line 3 (oscillator).

Echo includes 20 echo send controls, 4 echo return gain controls, 8 echo return assign push buttons (lighted), for 4 echo busses.

Low frequency EQ (±12 dB): 40 cycles (shelf), 100 cycles (shelf), and 300 cycles (Peaking).

High frequency EQ (±12 dB): 1.5 Kc (Peaking), 3 Kc (Peaking), 5 Kc (Peaking), and 10 Kc (Shelf).

Output assign includes 8 output assign lighted alternate action push button switches.

Metering includes 8 - 4½" Simpson lighted VU Meters for output assign channels; 4 - 2½" Simpson unlighted VU Meters for echo return; and, 2 - 4½" Simpson lighted VU Meters for stereo mixdown.

Additionally, there are: 8 master pots, 16 stereo earphone pots, 16 mixdown concentric pan-gain potentiometers, talkback and slate push buttons, mike gain controls, built-in 5 frequency oscillator, stereo record and monitor amplifiers, and external rack-mounted power supplies.

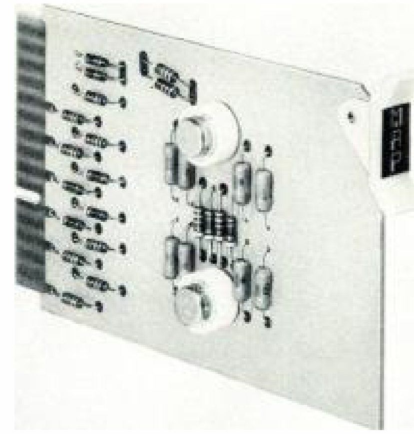
The console in kit form is priced at $9,515.30. Wired the unit sells for $15,225.30.

OPAMP LABS, INC., 1033 NORTH SYCAMORE AVE., LOS ANGELES, CA. 90038. PHONE: (213) 934-3566

For additional information circle No. 68

MODULAR AUDIO PRODUCTS INTRODUCES SOLID-STATE REMOTE CONTROLLED FET AUDIO SWITCHING MODULE

Modular Audio Products New Model 4011 FET Audio Switching Module pro-



vides four fully indpendent, remote-controllable audio switch circuits on a compact, plug-in PC card, with gold plated edge contacts.

Ideal for use in high quality professional audio applications, Model 4011 may be utilized individually, to perform various switching functions in console mixing channels; or in multiples, to form large or small Matrix type switching/routing systems.

Designed for optimum performance in 600 ohm balanced or un-balanced circuits, Model 4011 features: High speed — 2 microseconds from ON to full OFF state (-100 dB @ 1 kHz, +4 dBm input Nom.). Low Distortion — 0.07% @ +4 dBm output, 1 kHz; and High Input Level capability, +22 dBv Maximum. Frequency response of the unit is extremely flat from D.C. to 20 kHz (-0.1 dB Max.). Power requirements: ±15 VDC @ 10 mA Maximum per card. Dimensions: 2-¾" high x 4½" long x 13/16" wide.

This latest addition to the well known MAP IMPAC Series of "Integrated Modular Professional Audio Components" is part of a Total Modular System Concept, employing various plug-in modules, standard 19" w. x 3½" h. Rack Mounting Card Frames and Modular power supplies. MAP IMPAC offers a high degree of flexibility in the design of Modular audio systems, now featuring the option of self-contained or remote controlled operation.

MODULAR AUDIO PRODUCTS
50 ORVILLE DRIVE, AIRPORT INTERNATIONAL PLAZA, BOHEMIA, N.Y. 11716, PHONE: (516) 567-9620

For additional information circle No. 69

AGFA-GEVAERT HAS UNLIMITED SUPPLY OF PEM-468 MASTER TAPE

Agfa-Gevaert, Inc., has announced the PEM 468 Mastering Tape, from Europe, is available in almost unlimited supply in this country. Previously it had been in limited supply.

In announcing the availability of this mastering tape, Robert A.M. Coppenrath, president, noted that PEM 468 is not only a high output, low noise, but also low print mastering tape.

"The characteristics which make PEM 468 superior, and hence more expensive than American made tapes, are its reliability, consistency, and uniformity. The tolerances are plus or minus ¼ dB within a roll, and plus or minus ½ dB from roll-to-roll. As a high output, low noise tape it also offers low print . . . up to 10 dB less print than other popular mastering tapes", Maria Curry, national sales manager, said.

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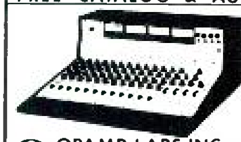

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1 - Urei 964 Digital Metronone	$275.00
1 - AKG C/60 mike w/supply	$100.00
2 - JBL Model 99 speaker systems, walnut	$140.00 ea.
2 - Altec 612 speaker cabinets	$80.00 ea.
4 - dbx Model 187 (16 channel total)	$900.00 ea.
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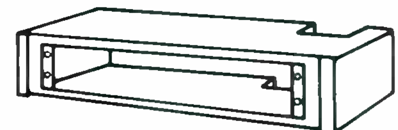

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FOR SALE: Neumann Computer Controlled Disc Mastering System including many accessories. Currently operating in Ruston, Louisiana. Replacement cost in excess of $120,000. ASCO selling price – $50,000. Write or call for details: RON NEWDOLL - ACCURATE SOUND CORP., 114 - 5th Ave., Redwood City, CA 94063 (415) 365-2843

EMPLOYMENT

VIDEO ENGINEER: Western Pennsylvania college seeks qualified individual to maintain and supervise newly-installed ETV one-inch color studio. Responsibilities include technical production of ETV programming for campus and associated cable channel, all equipment repair and maintenance, facility scheduling and liaison with department media producers. B.S. and prior color experience required. First Class License and knowledge of digital electronics desirable. Available immediately.
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Model 532 is a single channel version of UREI's 530 Dual Graphic Equalizer, offering real economy for recording, sound reinforcement, radio and TV, and monaural music systems. The nine equalizers are centered at each octave from 50 Hz to 12.5 kHz. Input may be operated balanced or unbalanced and the transformer-coupled output amplifier is capable of delivering +20 dBm into a 600 ohm load. Signal-to-noise ratio at maximum output is 110 dB, distortion is below 0.5%. Half rack size. Available from your UREI dealer.

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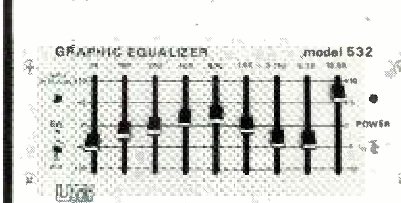

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For additional information circle No. 75

LATE NEWS . . .
(continued from page 14)

KLEFFMAN NAMED AUDIO/VIDEO V.P. BY AMPEX

As announced by Arthur H. Hausman, Ampex president and chief executive officer, Donald V. Kleffman has been promoted to vice president and general manager of the audio-video systems division of Ampex Corporation.

Kleffman had been named general maager of the audio-video systems division in May of 1976 after four years as marketing manager. He joined Ampex in1959 as an applications engineer. Kleffmen has held both product and marketing manager positions since 1964.

In 1975, Ampex introduced 16 new audio-video products for the professional audio-video industry.

FREEMAN NEW BGW NATIONAL SALES MANAGER

In an announcement by BGW Systems president Brian Wachner, Wayne Freeman has been appointed to the position of National Sales Manager of the precision electronic power equipment manufacturer.

Freeman had formerly been sales manager of Telesco International Corporation, and will handle the sales management for BGW's professional and consumer lines.

STAN FORCE NEW HARRISON V.P., ENGINEERING

Stan L. Force has been appointed Vice President of Engineering according to an announcement by company president Dave Harrison of the Nashville based pro-audio equipment developer and manufacturer.

Force received a BSEE from the University of Kentucky in 1973.

"Stan is a professional musician, and has worked as a recording studio mixing engineer, as well as a cutting engineer in several Nashville studios, and is very well versed in all the requirements of the recording industry from both a hands-on and a technical point of view, which we feel is an absolutely necessary experience for a person with the responsibility of developing and manufacturing the most advanced forms of studio equipment." said Harrison in making the appointment.

Force joined Harrison at it's inception in 1974, and has been actively involved in the design and development of all Harrison products.

MARCH 19 & 20 DATES FOR FIRST PRO AUDIO SWAP MEET

The Express Sound Company, Inc., of Costa Mesa, California, and International Automated Media, of Irvine, both Orange County based companies, have announced their co-sponsorship of the first annual Southern California Professional Audio Equipment Mini Exposition and Swamp Meet. The event will take place at I.A.M.'s new state-of-the-art 24-track facility on March 19th and 20th.

Spokesman from both companies expressed hopes that this kind of program will offer pro recording enthusiasts an opportunity to exchange information and ideas with others in the field as well as the chance to offer their old or un-needed equipment for sale or swap.

Additional information may be obtained by contacting Express Sound Co., 1822 Newport Boulevard, Costa Mesa, CA 92627. Phone (714) 645-9143.

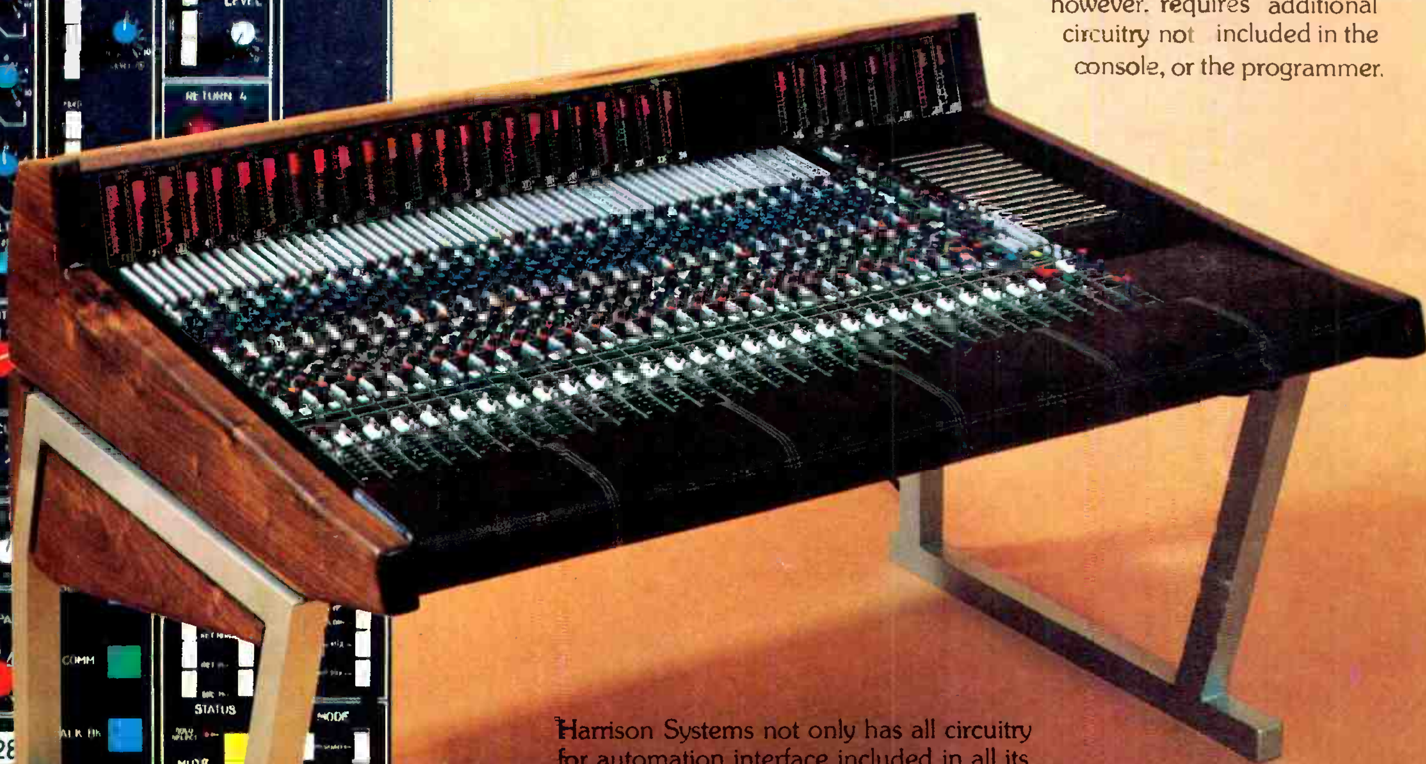
STUDER-REVOX OPENS NEW REGENSDORF, SWITZERLAND WORLD HEADQUARTERS

Called Plant III, the newly constructed ten story (2 underground levels) building combines under one roof all headquarters functions, centralized research and development departments, data processing, as well as the addition of some of the company's most modern mechanical and electrical production facilities.

It is expected that 500 of the company's 1,450 employees will be housed in the new facility when the near Zurich building is fully occupied.

According to company spokesmen, the expansion is in line with the Studer-Revox dedictation to a philosophy of farsighted development of the finest in Audio Products.

NOW
Affordable Automation From HARRISON 2824



Automation ready means different things to different people. At Harrison Systems, it means that our consoles can be connected to an existing and available automation programmer with a simple multi-pin connector. Every Harrison console ever produced has this capability. Some are now being used for automated mixing.

Some manufacturers of recording consoles *claim* to be automation ready. Interface with existing programmers, however, requires additional circuitry not included in the console, or the programmer.

Harrison Systems not only has all circuitry for automation interface included in all its consoles, but beginning January 1977, all consoles shipped will have been interfaced and tested with an automation programmer during final test. This insures plug-in compatability should you automate in the future.

True Automation Readiness...
part of the NO COMPROMISE philosophy at

Harrison

Harrison Systems
P.O. Box 22964
Nashville, Tenn. 37202
Tel. (615) 834-1184 Telex 555133

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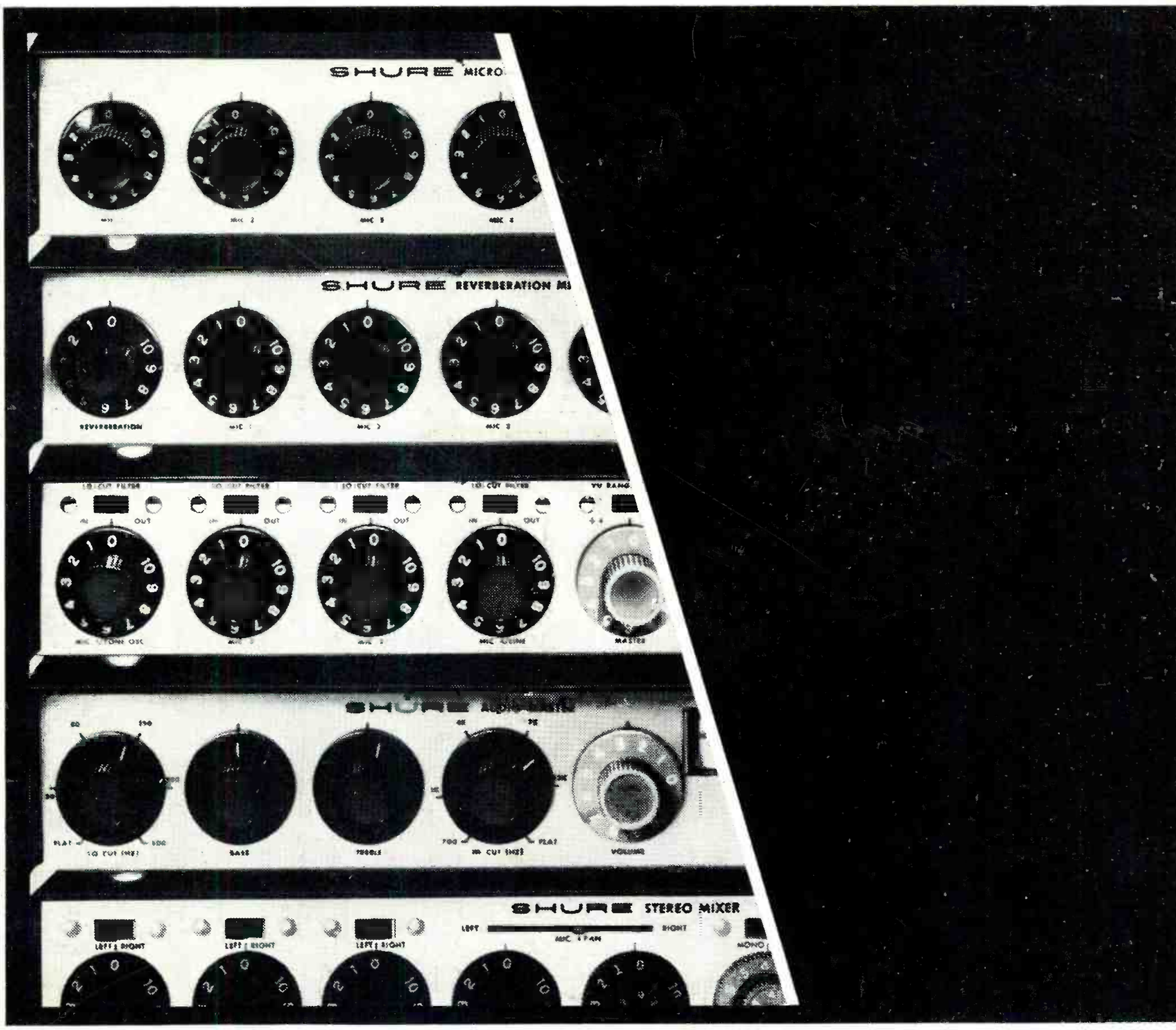
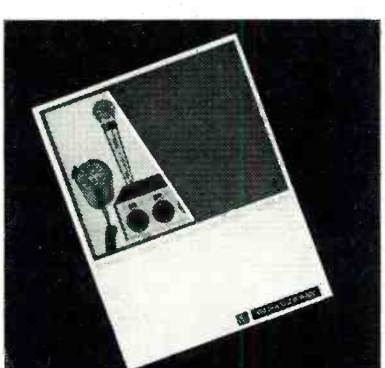


Panel full of miracles.



Shure mixers, audio level controllers, and feedback controllers are all designed to deliver more audio control, more features, and more performance dollar for dollar than any other components with similar audio features. Their compact size and modular "stackable" design mean they can be easily combined in various configurations in even the smallest spaces. And they're versatile—their input-output flexibility equips them for an extremely wide range of audio applications, giving you control you never thought possible without bulky, expensive installations. You can easily put together a system that's exactly right for your precise needs without putting extra dollars into built-in features you really don't need. For the details on our entire line of miracle workers, write:

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