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JUNE 1976
VOLUME 7 – NUMBER 3

Recording engineer producer

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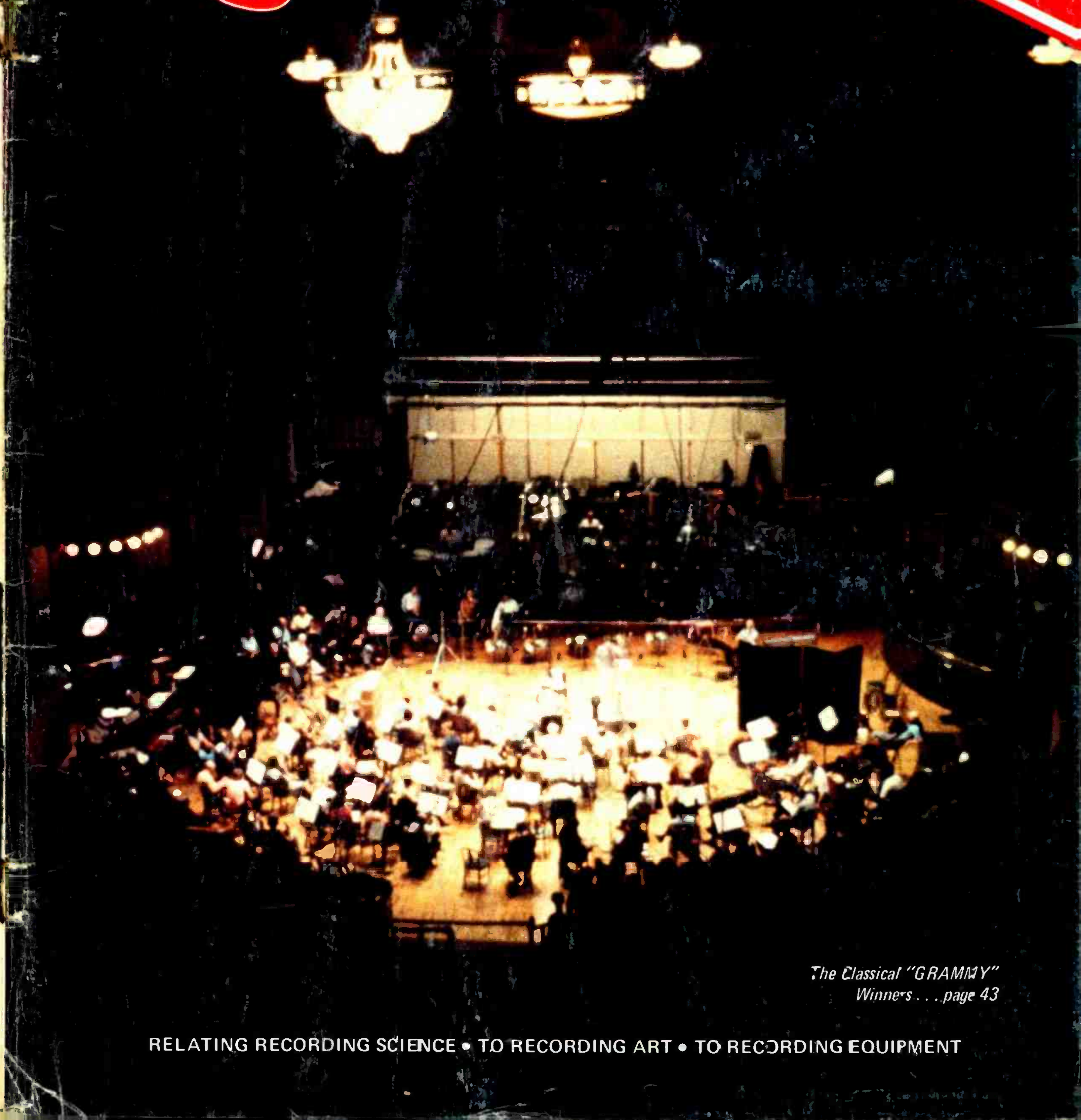

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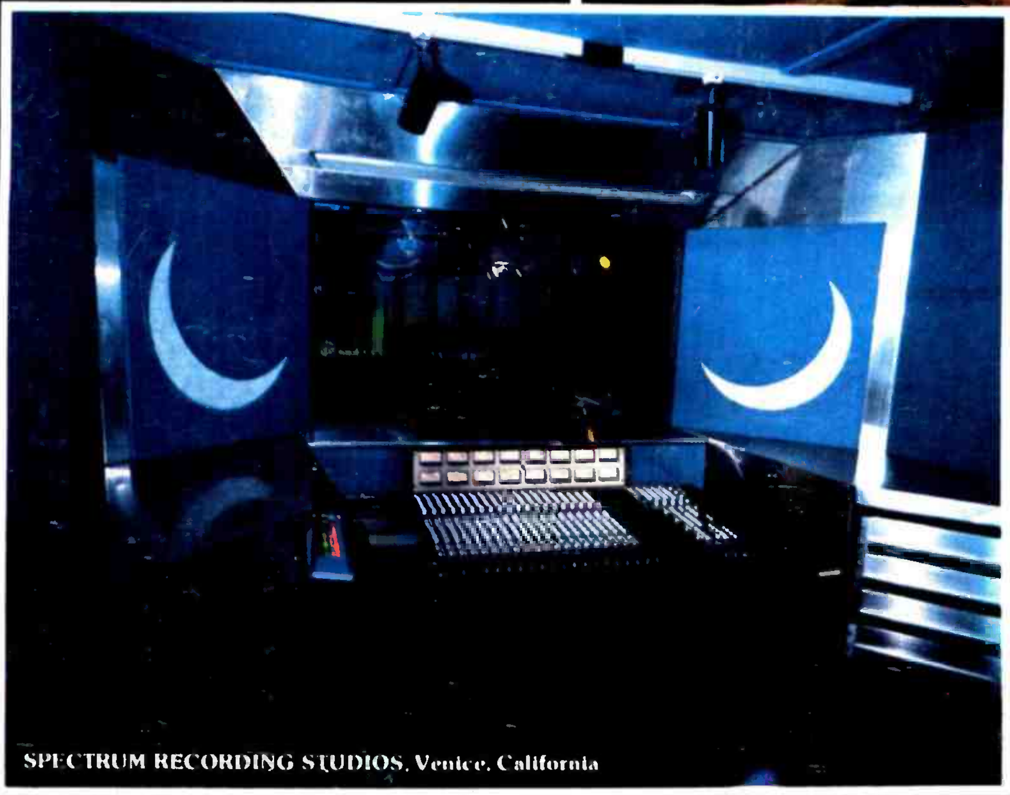

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Ampex introduces an audio recorder/reproducer that's so much better than anything else on the market, you'll find it impossible to make a comparison.

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There's never been an audio recorder like the ATR-100. It handles tape better than any previous machine, and it processes audio signals more faithfully than anybody else's best recorder.

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The ATR-100 has a triple-servo transport. Both reel motors and the capstan are servo-controlled in a tightly coupled loop. This servo arrangement does away with pinch rollers forever, and moves tape positively in both directions.

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Best Audio Recorder.

The ferrite heads are new, too, and will last a lot longer than any we've ever used before. Flux-gate head design gives unbelievable response; at 15 in/s, the ATR-100 will deliver ± 3/4 dB from 100 Hz to 15 kHz. You'll see "flatter" curves than ever before.

Precision machining of the heads means that you can change or replace them without going through a mechanical alignment routine. A limited azimuth adjustment is all that's needed. A single screw is also the key to changing heads—loosen it, unplug the head assembly and that's all there is to it. You'll take care of your ATR-100 with a minimum of simple tools.

Another tape handling improvement is provided by ceramic tape guides. These edge guides resist wear by fast-moving tape and team up with the hard ferrite heads for more hours of use with minimum maintenance.

Electronics: New Ideas Throughout

The basic ATR-100 contains only audio and transport electronics. There's an unbalanced input and output circuit, and that's all. No meters or operator adjustments.

Normal balanced input and output circuitry is contained in the optional channel input/output units. You use one, two, or four I/O channels (located in an overhead bay) with an ATR-100, giving you your choice of full track, stereo or four-track recording.

Each I/O module has its own metering, line drivers, balancing transformers, amplifiers and controls. We've provided the ability to switch meter circuits, giving you a choice of ASA ballistics with VU indication or the European EBU ballistics with Peak indication.

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Transport controls have international symbols, and no restrictions on sequencing. You can go from any transport mode to any other mode without going through stop, and without waiting for anything to slow down or speed up. Furthermore, the ATR-100 has dynamic braking that takes over in the event of a power failure. This machine always stops in a programmed manner, even without power.

You won't stretch tape with ATR-100, because it always checks to make sure the tape is fully tensioned before actuating the servos.

The power supply is heavy duty and universal. No matter where you take the machine, you'll be able to plug it in the wall.

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Every ATR-100 is capable of the four standard recording speeds—3-3/4, 7-1/2, 15 and 30 inches per second. You can select any pair, adjacent or not, and change them later if you wish.

And no matter what speeds you select, you'll get a rewind and fast forward capability of 2,400 feet in just 60 seconds. That's really moving tape.

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RECORDING engineer/producer

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

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

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JUNE 1976
VOLUME 7 – NUMBER 3

THIS ISSUE SPONSORED BY THE FOLLOWING ADVERTISERS:

Think of them as your musical instruments.

The audience can't see you. But they can sure hear you.

They don't know it, but they're depending on just one person to get the music to them. And that guy is you.

It's not something an amateur can do. It's an art. And that's why Yamaha has designed 3 superb mixing consoles with the qualities and range of controls that the professional sound reinforcement artist needs.

For instance, our exclusive 4 x 4 matrix with level controls gives you more exacting mastery over your sound than the conventional method of driving speaker amps directly from the bus outputs.

Features like that are years away except on the most expensive mixers. On the Yamahas, it's standard equipment. And so are transformer isolated inputs and outputs, dual echo send busses, an input level attenuator that takes +4 dB line level to −60 dB mike level in 11 steps, and 5-frequency equalization.

Whether you choose the PM-1000-16, the PM-1000-24 or the PM-1000-32, Yamaha gives you the flexibility you need to turn your job into an art. And because they're designed from the ground up to perform on the road, more and more professional sound men around the United States and the world are depending on Yamaha, night after night, gig after gig.

If you've never thought of your mixing console as a musical instrument, we'd like to invite you to stop by your Yamaha dealer. Once you've checked out the operation manual and tested for yourself what the PM Series can do, we think you'll come away a believer.

YAMAHA

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LETTERS and LATE NEWS

ARTS COUNCIL URGES COOPERATIVE EFFORT WITH PBS FOR IMPROVED TV SOUND

The National council on the Arts, has recommended the National Endowment for the Arts join the Public Broadcasting Service in developing a new system for delivering greatly improved network television sound. The council urged manufacturers, common carriers, broadcasters and the Federal Communications Commission to cooperate in improving TV audio both in the studio and in the home.

"Improvement of sound," the Council said, "is essential if the potential of arts programming on television is to be realized."

The statement follows a May 15 meeting during which Council members were briefed by PBS officials on PBS's new Digital Audio for Television (DATE) system. The system, in development at PBS for several years, enables television networks to distribute to TV stations up to four channels of high-fidelity audio.

Basic Description

The DATE system has two basic parts, the analog-digital-analog conversion systems and the modem. The ADA systems, as their name implies, convert the analog audio signals into digital form and at the receiver back into audio. The modem (modulator-demodulator) provides the means to convey the digitally-encoded audio along with the video. A basic block diagram is shown in Fig. 1, where it can be seen that the system can provide up to four discrete audio channels.

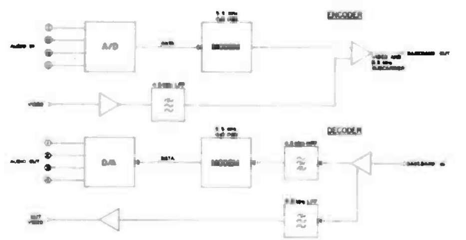

Fig. 1. Block diagram of the DATE system.

*The audio input channels are converted into data by the analog-to-digital converter. The data then modulates a 5.5-MHz carrier within the modem. The modem output is combined with video to generate the baseband signal which is fed into the long-haul transmission facilities. At the receive point the subcarrier and video are separated by filters. The subcarrier is then fed into the receiver modem where it is converted back into data. The audio data is used to recreate the original audio fed into the system at the transmit point.**

Using DATE, PBS Vice Chairman Hartford N. Gunn, Jr., told the Council, "we can take stereophonic high fidelity sound from the studio, theatre or concert hall and transmit it over most of the present microwave circuits of the telephone company along with the picture or video portion of the program. Our local stations can receive these signals with special equipment and thereby greatly improve the sound transmitted to the homes in their area."

Noting that at present, a system such as DATE can not be activated because its use is not permitted on telephone microwave circuits, Gunn said broadcasters must be "willing and able to take the lead and compel the distribution of high quality sound."

**Excerped from the introduction to the paper: DATE: A Digital Audio System for TV, by R. Evans Welmore, Journal of the SMPTE, Volume 83, March 1974.*

CUNNINGHAM ANNOUNCED AS NEW TECHNICAL DIRECTOR OF UNITED/WESTERN/COAST RECORDERS

As announced by Milton T. (Bill) Putnam, President and Board Chairman of the URC Companies, James (Jim) Cunningham has been appointed Director of Technical Facilities of the recording studios group.

Cunningham is a graduate of the University of Michigan and Northwestern University, and pioneered in the field of stereo recording and broadcasting for NBC. He had been Chief Engineer of Universal Recording in Chicago during the early 1960's. In 1967 he joined Sound Market Recording in Chicago and became Vice-President and Studio Manager.

Cunningham has published many papers in various technical journals, and has innovated a number of recording techniques in use today.

He was most recently Central Vice-President of the Audio Engineering Society, and served as a past officer of NARAS in Chicago.

URC is also the parent company of U.R.E.I., located in North Hollywood.

REORGINIZATION AT WESTLAKE, ANNOUNCE EXCLUSIVE HARRISON DISTRIBUTION AGREEMENT

It was announced at the Los Angeles AES Show that, effective May 1, 1976, Glenn R. Phoenix has been elected President of Westlake Audio, Inc. by its

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Circle No. 106

Board of Directors. Glenn was previously with the Mincom Division of the 3M Company and joined Westlake soon after its inception in 1971.

Paul Ford, prior President, has been elevated to Chairman of the Board. Westlake Audio, best known for its world-wide turn key installations, utilizes the latest and most progressive design technology including the services of its European consultant, Thomas L. Hidley. Examples are soon to be evident in both Southern California and Georgia where construction of new, multi-track and multi-media facilities are underway.

The Company also plans to expand its operation this summer in Nashville, Tennessee. Jeff Segar, Westlake's corporate Vice-President, will direct this operation.

In keeping with its tradition of providing the latest, state-of-the-art equipment to the recording industry, Westlake has also announced their appointment as the exclusive United States distributor for the Harrison Models 4032 and 3232 Master Recording Consoles. Sales and service will be provided by their Los Angeles and Nashville offices.

Glenn Phoenix, President of Westlake Audio, describes the Harrison Consoles as "the culmination of many years of creative design, incorporating the quality, functions and features our industry has needed, but has never before had the opportunity to truely experience".

Consoles have already been delivered in Los Angeles to Warner Bros. Records, Kendun Recorders and United/Western Recorders. Others have been installed in American Studios in Nashville, The Sound Room in Philadelphia, Studio in the Country in Bogalusa, with more scheduled for summer delivery.

EVERYTHING AUDIO EXPANDS, MOVES TO NEW QUARTERS

Effective on June 1, 1976 the new address of the company is **7037 Laurel Canyon Boulevard, North Hollywood, CA 91605.**

The new 6,000 square foot facility has been constructed to include a complete Everything Audio Control Room, typical of those turn-key designs already installed, as well as those in various stages of design and construction. The fully operational control room will afford clients of the company the opportunity of a hands-on interface with a wide spectrum of audio equipment offered by the company.

Along with Everything Audio, its sister company, Video Products Sales, will also be housed at this location. The corporation now boasts the ability to interface audio and video on the professional studio level.

New telephone numbers are: Everything Audio (213) 982-6200. Video Products Sales (213) 982-5600.

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When you stepped up from four tracks to eight, you knew sixteen wasn't far off. And if you've been looking for someone to put it all together for you, consider it done.

The el-tech 1616-25 recording console gives you all of this:

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- 16 Monitor/Cue Channels
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- 2 Echo Busses
- 8 Mixing Busses and 8 Direct Outputs
- 96 Point Patch Bay
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. . . and luscious good looks.

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NECAM
Computer Assisted Mixing

People tell us it was the hit of the show. The 54th AES Convention in Los Angeles. The only manufacturer confident enough to organize large scale demonstrations of its computer assisted mixing system. Nearly one thousand attendants experienced NECAM in 16 scheduled demonstrations. And, of course, it performed every time. This advanced system, utilizing a mini computer, core and floppy disc memory, and servo driven faders, is the practical and reliable approach to automation. Simple and instinctive to use, it also provides studios with an advanced time (and money) saving tape locator and cue search facility. NECAM systems are currently being built for several leading world studios. You need to know NECAM. It's in your future.

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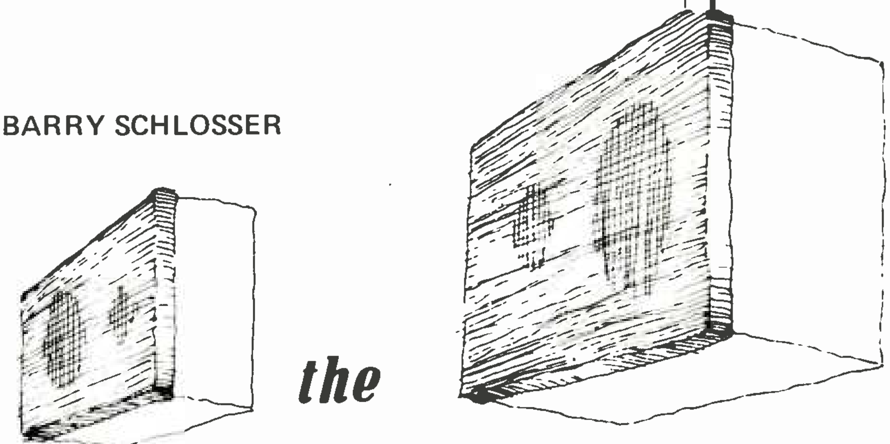
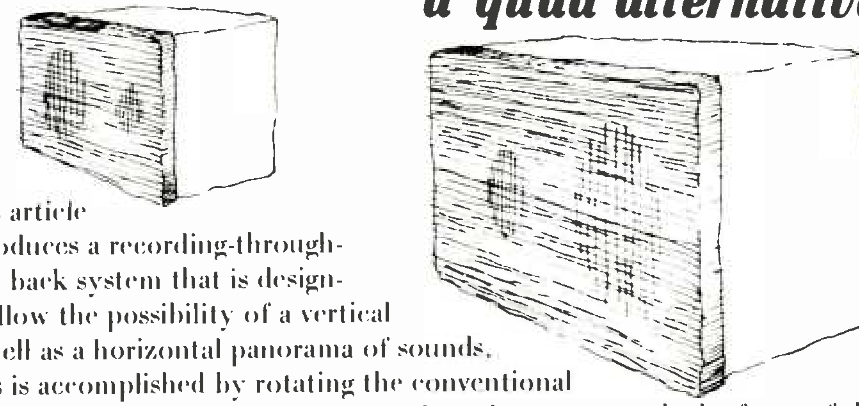
Neve

by
BARRY SCHLOSSER

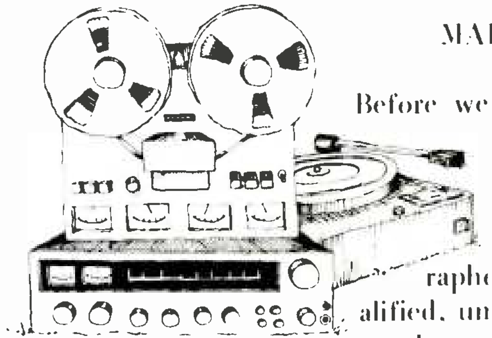


the "FyF System"

a quad alternative



This article introduces a recording-through-play back system that is design- to allow the possibility of a vertical as well as a horizontal panorama of sounds. This is accomplished by rotating the conventional "surround" quadraphonic sound field 90 degrees towards the front of the listening room. In practice, this means placing all four loudspeakers in front of the listener, instead of surrounding him with the sound field. There are two sets of loudspeakers in front of the listener, one set placed near the ceiling, and the other near the floor of the room. The "FyF System" is so-named in order to distinguish this arrangement from conventional quadraphonic loudspeaker placement geometries. The purpose of this article is to introduce the concepts behind the FyF System with enough "how-to" information to allow interested readers to experiment and, perhaps, to develop FyF System techniques of their own.



MARKETING CONSIDERATIONS

Before we delve into the technical considerations and engineering parameters of the FyF System, it might be beneficial to first discuss the reasons why previous attempts to market quadraphonic records have not met with unqualified, universal success. There have been any number of suppositions proposed as to why quad recordings haven't sold as well as expected. There are those who say that it is because the quality of music on these releases is lacking. Others blame poor sales on the lack of an international encode-decode standard. Perhaps, however, the answer lies within the perceptual capabilities of the human apparatus. It seems a well established fact that ears are designed to most accurately process frontal information. The ability to judge sounds emanating from behind the listener is much less than the ability to correctly judge sounds arriving from the front.

(continued overleaf)

One of the family

The DO54 joins a distinguished family of E-V dynamic microphones. Including the very popular 635A, the extended-range RE55, and the low-noise RE50. The new DO54 shares many characteristics with our other omnidirectional mikes, but its differences can provide a very useful alternative.

The runt

First, the DO54 is slightly shorter and slimmer than a 635A. The first mike designed with the cable connector in mind. It's not tacked on. Looks great for handheld use, and it's easier to tuck away in odd corners.

Truly full-range

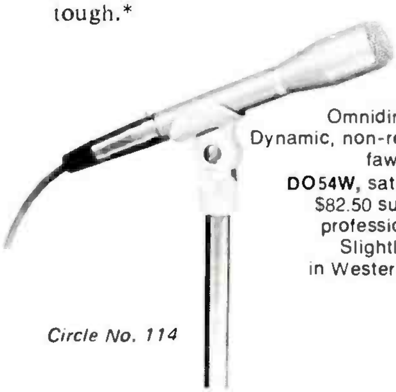
Despite the miniaturization, the DO54 boasts a response of 50 to 18,000 Hz ± 3 dB. Which rivals some very expensive competition and makes it ideal for demanding close-in instrumental pickups.

Full output

And the DO54 puts out. With level that matches the RE55, and a dynamic element that won't be over-driven by enthusiastic horns, drums, or close-up vocals.

Quietly reliable

Our internal shock protection stops handling and cable noise with the best of them. And our Acoustalloy® diaphragm is almost indestructible. The DO54 easily earns the same unconditional protection against malfunction as all our other Professional Microphones.
So its got to be tough.*



DO54
Omnidirectional
Dynamic, non-reflecting
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DO54W, satin white.
$82.50 suggested
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Slightly higher
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Add-on Flexibility

Of course the DO54 fits our 3/4" microphone options, like the security clamp (when you can't control access to the mike), a very neat stud mount adapter with switch, a most effective shock mount, and super Acoustifoam® windscreen.

In short (no pun intended) the DO54 is a versatile new omnidirectional dynamic in the best tradition of E-V. Dependably delivering great sound from a most modest package.

Get your toughest microphone problems in hand... with this new Electro-Voice DO54. Available now from your helpful E-V professional sound specialist.

*If your DO54 fails to function for any reason (cable, connectors, and finish excepted) within 2 years of purchase, send it back. We'll fix it free. And fast. And there's no time limit to replacement or repair of faults in workmanship and materials. That's our limited Professional Warranty in a nutshell.

Electro-Voice® INC
a gulton company
Dept. 561RP, 674 Cecil Street
Buchanan, Michigan 49107

The Author:

BARRY SCHLOSSER, at 25, has lived in Miami Beach and Atlanta . . . studied botany and psychology at the University of Georgia for four years . . . was granted research facilities at Fernbank Science Center in Atlanta, and appointed to the botany staff at the Russel Research Center in Athens to design and construct machinery to mass produce green algae . . . was the proprietor of a retail stereo shop for two years . . . served as the credit manager of Schookids' Records, Inc. . . . is currently vice-president of Grape City Enterprises, Ltd., a wholesale record distributing company . . . continues to play drums . . . is an associate member of AES, and one of the original members of the Society of Audio Consultants . . . has produced live concerts for the Duke Ellington Orchestra, the Earl Scruggs Revue, the Mike Greene Band, Arthur Hurley, Gottleib and others . . . most recently designed and supervised construction of FyF Studios, Inc. . . . is presently designing a recording facility based on geodesic structures . . . and most importantly, is responsible for album production at FyF Studios.

Most people find that high level sounds coming from the rear are irritating and distracting. In fact, a majority of people become startled when they hear loud sounds come from behind. Of course, it is acceptable to use the startle factor as a special effect on occasion but, to have an almost continuous stream of loud sounds arriving from the rear seemingly becomes annoying to many people. An exception to this situation would be the case of a musician playing in front of his amplification system. In this instance, the musician is not able to startle himself because he is aware beforehand of the sounds that will come forth from his amplifier. This is analagous to the inability to physically tickle yourself. Try to tickle yourself. You may laugh at the situation but you won't laugh from the sensation, simply because you know beforehand where you will tickle yourself. The tickle sensation is the result of your skin being startled in an unpredictable fashion by an outside agend; generally this is most annoying. In a similar fashion, your ears become annoyed when they are *tickled* by too many sounds from the rear. Often discussed, another major stumbling block to mass market acceptance of surround quadraphonics is the reluctance of most people to arrange their listening room to accommodate the rear loudspeakers. The majority of people it would seem, are unwilling to place their chairs and couches in the center of the room so that the proper playback perspective is achieved.

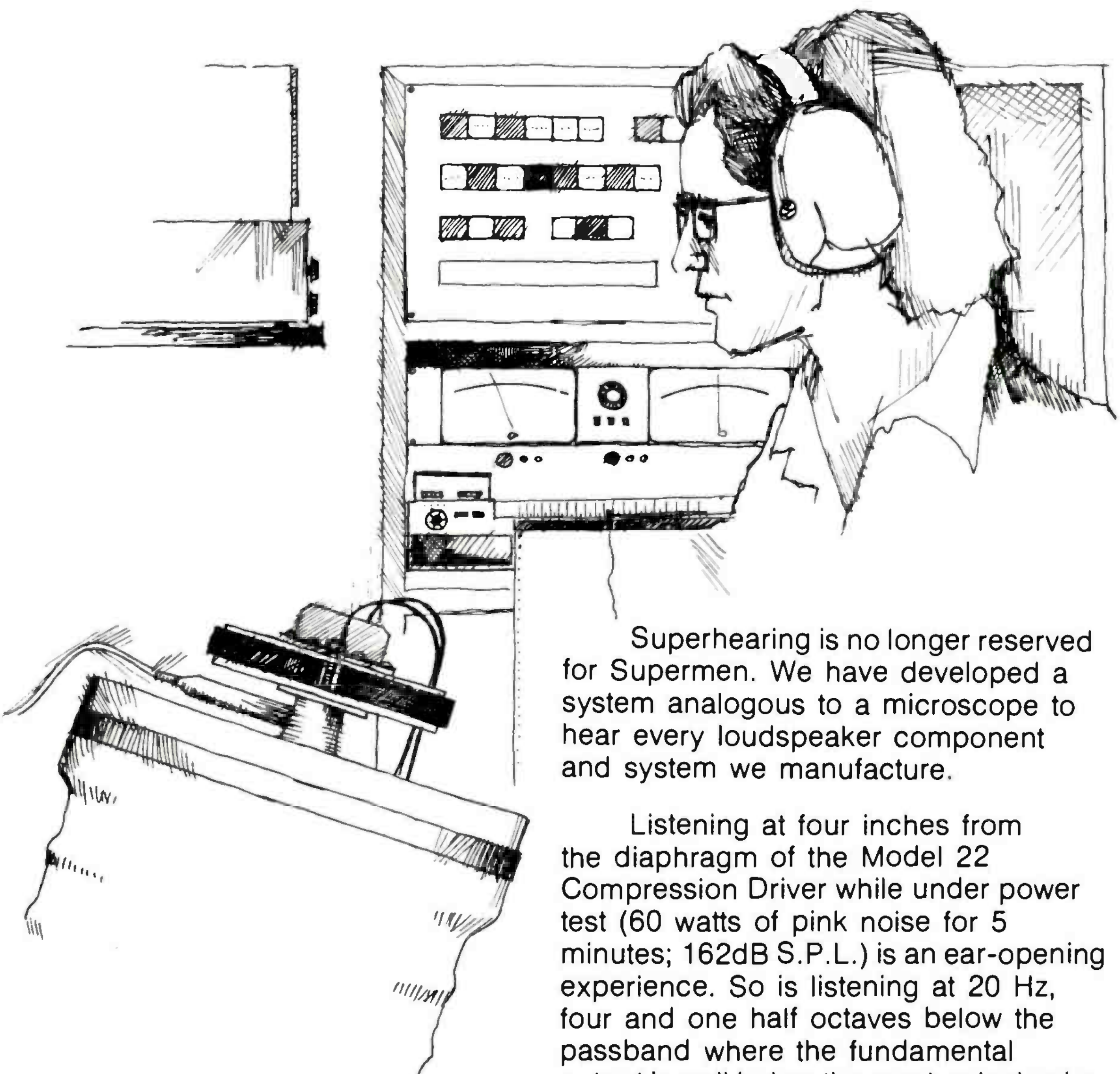
Finally, and perhaps most important, few people have been willing to spend the extra money on a quadraphonic sound system just to recover ambient invormation. There is no convincing argument as to why ambient information needs to be reproduced from the rear at all. If the music is properly recorded, it will generate its own ambient information and any high-level ambient information from rear loudspeakers may only serve to blur the image in the front.

Does the above critique of the current quadraphonic plight overmodulate your aural well-being? Don't despair. Consider what follows a genuine *choice*! No longer do you have to drive yourself crazy trying to engineer and produce a musically acceptable recording in a manner contrary to your perceptual capabilities. The FyF System can ameliorate the current quadraphonic situation in such a way that the above mentioned problems will dissipate and pale into insignificance.

THE FyF SYSTEM OF QUADRAPHONY

First, the FyF System is designed to correspond with human perceptual abilities so that the sensory receptors (your ears) receive the maximum benefit from four channels of information while minimizing listener fatigue. We propose that the best location for the four information channels is in front of the listener in the FyF configuration. To gain a clearer insight of the FyF System, we might suggest that the reader take a piece of blank typing paper and lay it horizontally on a table in front of himself. At each corner of the paper draw in a speaker system. In the center of the paper draw a small circle. The circle represents the listener. This is the conventional way of listening to a quadraphonic playback system. The listener is surrounded by the loudspeakers. Now lift the paper up so that the paper is perpendicular to the tabletop. Forget about the circle in the center of the paper and visualize that the four loudspeakers are now vertically arranged in front of you. By using a quad panner, it is of course possible to place a sound source anywhere you desire in the field in front of you. If anyone wishes to compare the FyF System to stereo (stereo being defined as two channel horizontal reproduction), a line should be drawn between the two bottom loudspeakers. This line represents the narrow panning ability of stereo. With stereo there is only the possibility of horizontal motion. The same situation applies to conventional surround quadraphonic systems. Expansion of the stereophonic field into Top Left=Top Right and Bottom Left= Bottom Right gives the ability to accurately reproduce reality. Previous systems can accurately reproduce sounds but not the sounds' placement in such a way that the sound corresponds to physical placement in any but a horizontal mode. The FyF System, we think, allows the duplication of audible-reality and physical-placement-reality within a much broader field or matrix than the narrow horizontal field of stereo and surround quadraphonics, this is because the possibility

FIGURE 1: FyF control room monitoring environment. The smaller speakers at the top and bottom comprise the FyF System monitors.



of vertical placement and vertical motion now exist.

The next important consideration is the compatibility factor. The FyF System under discussion here is fully compatible with traditional stereophonic and monaural sources. When played back through the bottom loudspeakers, traditional stereophonic and monaural sources sound exactly the same as if they were played back through a conventional stereo system. These same sources actually appear to be enhanced when played through all four loudspeakers if proper left-right balance is maintained. Results obtained when conventional surround quadraphonic records were played through the FyF System are described in the "Experiments" section of this article.

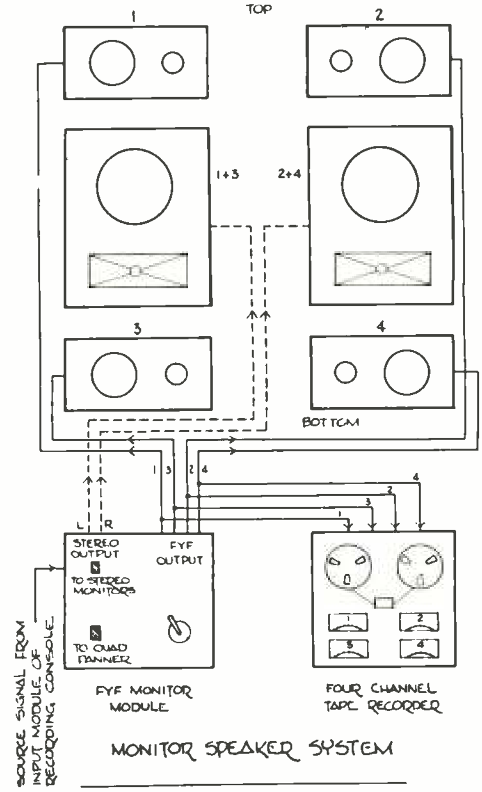
It would seem that many more people would be willing to place four loudspeakers in front of them in their listening room rather than in each corner of the room. The listeners' position in the room is not nearly as critical as it is for surround quadraphonic systems due to the fact that many more room modes and resonances are excited by surround quadraphonic fields thus making it more difficult to control the listening rooms' acoustic properties. If the playback equipment is set up for the FyF System, the furniture placement in the listening room does not impose the decorative hardships required by surround quadraphonic playback systems.

It must be kept in mind that the most important consideration is the musical experience itself. Without good music running through the equipment, the equipment is just so much junk. Accurately reproducing and enhancing the musical experience is what the FyF System, and indeed all good sound systems are all about FyF with the system proposed here we believe that (it is excitement that sells records). Entirely new and exciting production techniques are now available to interested engineers and producers. No longer do writers, musicians, producers, and engineers have to spend hopeless hours trying to figure out how to make sounds coming from the rear sound exciting and musical.

FyF SYSTEM EXPERIMENTS

Figure 1 pictures some of the equipment assembled by FyF Studios, Inc. in one of its control rooms to obtain empirical data for this article. There are three separate monitoring systems: quadraphonic, stereophonic, and monophonic. The two pairs of small loudspeakers at the top and the bottom comprise the FyF monitors. The two large loudspeakers are for monitoring stereo signals, A monaural signal can be sent to any of the six loudspeakers for evaluation. The bottom loudspeakers of the FyF monitors are also used to evaluate stereo. All of the loudspeakers are angled so that they sound best and each loudspeaker is equalized to the room for a flat frequency response through the use of several pieces of test equipment including a real-time analyzer.

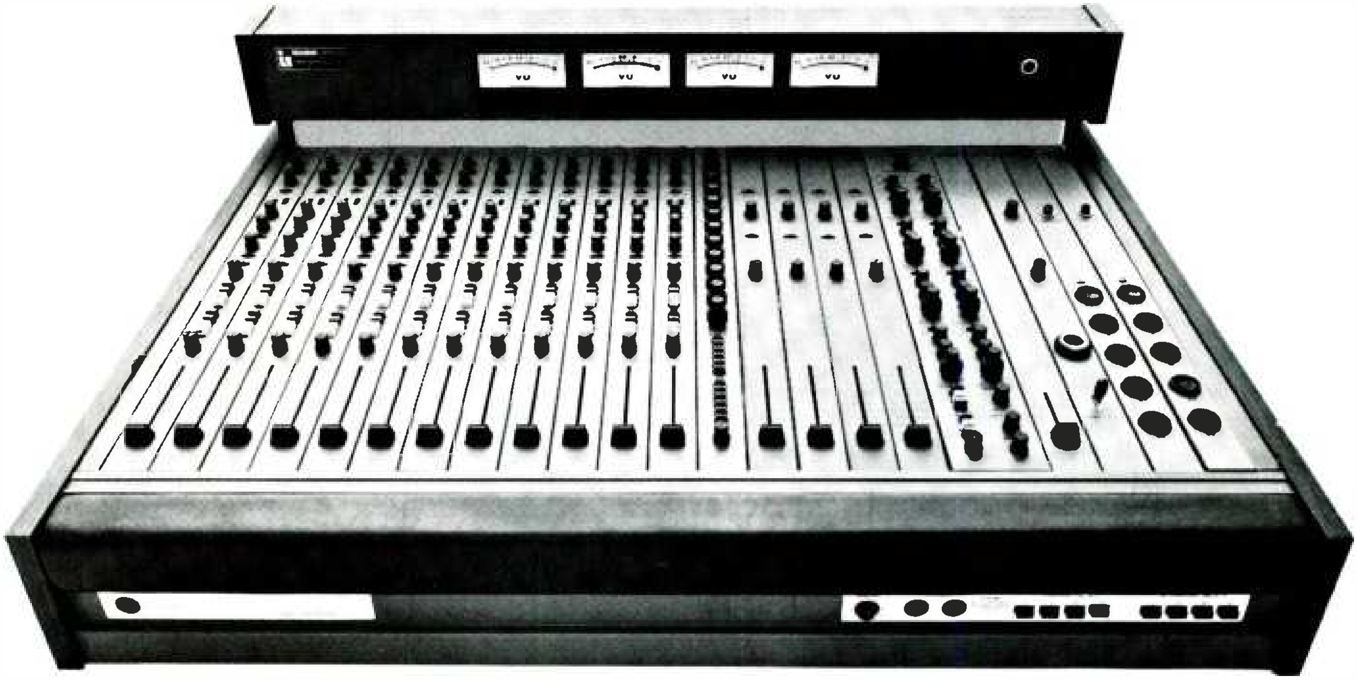
FIGURE 2:



CONTROL ROOM PROCEDURE

Incoming signals from the recording console are first evaluated on the large stereo monitors for signal integrity and musical quality. These signals can come from another multi-track tape machine and/or from live musicians. As soon as the engineer is satisfied with the accuracy of reproduction, the large stereo monitors are switched off and the FyF monitors are activated. The incoming signal is then routed to a quad-panner which enables the signal to be sent to the appropriate FyF monitor (s) in front of the listener. Each signal is processed on its own quad panner in a similar fashion until the desired balance is obtained. If necessary, signal conditioning (effects devices, eq, reverb, etc.) can be applied at this point from the input module on the console. When a suitable mix is achieved, the signals are recorded on a four-channel tape machine as illustrated in figure 2. Track designation is standardized as follows: Top Left is track 1, Top Right is track 2, Bottom Left is track 3, and Bottom Right is track 4. In order to monitor the FyF System in stereo, a switching network combines Top Left and Bottom Left (tracks 1 and 3, respectively) to stereo left and combines Top Right and Bottom Right (tracks 2

Circle No. 116



We added your inputs to ours.

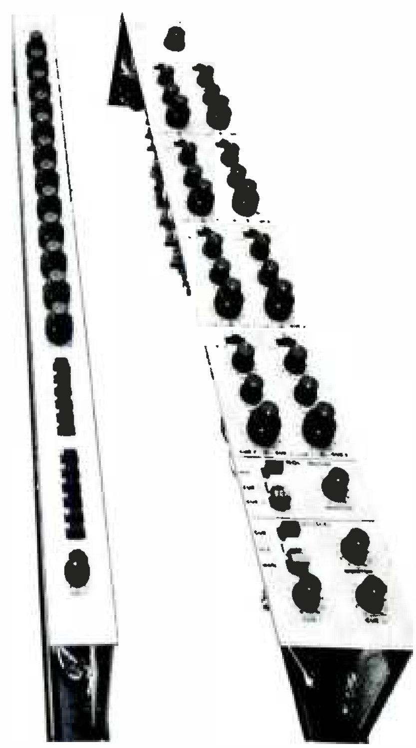
The result is the Model 10B—a good thing made better. Now there are peak reading LED indicators on each input, chassis mounted ¼" phone jacks for added stability, and the echo busses can be used with the program busses for 8-out capability.

An 8-channel Monitor Mixdown Module (Model 116) is now available optionally for direct interface. Each channel may be switched selectively to monitor buss or tape, and individual pan and gain controls are provided. Additionally, there is a split mono cue send (1 & 2), and an outboard automatic switching matrix.

When you want to expand the capabilities of the Model 116, the Model 120 Input Cue/Solo Module allows you to mult the accessory send signals from up to 12 inputs for a mix of tape cue and input cue on the split mono cue buss. And you can solo any of 12 active input channels.

The Model 10B is new. But it's built with the same design philosophy and integrity that has made the Model 10 one of the most popular mixing consoles ever. It's a creative tool that gives you the practical capabilities your imagination demands.

So if you have more talent than money, look into the Model 10B at your nearest TEAC Tascam Series dealer. Just call toll free (800) 447-4700** for the name and location of the one nearest you. **In Illinois, call (800) 322-4400.



Model 120 Model 116

Model 10B **TEAC.** TASCAM SERIES

TEAC Corporation of America, 7733 Telegraph Rd., Montebello, CA 90640 and 4, respectively) to stereo right. This switching network does not interfere with the signal path to the tape machine but is simply part of the monitoring system. The switching network can also combine left and right stereo so that a monaural signal can be sent to each loudspeaker for evaluation. Thus, in the above outlined monitoring system, there exists the capability to make sure that FyF System recordings are compatible with stereo mixes and monaural mixes.

FyF PLAYBACK PROCEDURE

After a final four-channel mix is obtained, the recording is auditioned both in the control room and in a separate playback room. Figure 3 illustrates the set-up of the monitoring system in the playback room. Two loudspeakers are mounted near the ceiling and two loudspeakers are mounted near the floor. A quadraphonic receiver is connected to the loudspeakers and the receiver is fed by either a four-channel tape deck or a turntable equipped with a cartridge suitable for reproducing SQ, QS, and CD-4 records. The receiver also has built-in facilities to decode SQ, QS, and CD-4 records.

EMPIRICAL RESULTS

Most of the tests for this article were conducted using a set of drums; however, we were able to reproduce a vocalists' voice from the top (head area), his guitar from somewhere around his waist, and his foot could be heard tapping on the floor. Getting back to the drums, we were able to reproduce the bass drum towards the floor, the snare drum two feet above the bass drum, the cymbals up in the air, and the hi-hat up in the air to the right of the snare drum. Depth of the image was found to be primarily a function of microphone distance and signal intensity. There was no difficulty at all in distinguishing top images from bottom images. There was also no noticeable hole-in-the-middle effect. When the FyF System was monitored in stereo, there was no degradation of pan assignments. The instruments fell into position exactly as if they had been recorded in stereo in the first place. The top and bottom signals merely combined to form a conventional stereo image. Similar results were obtained when the signals were played back monaurally.

In the separate playback room, results similar to those in the control room were obtained. The major difference was that the bottom loudspeakers were placed closer to the floor which served to increase the accuracy of the image. The distance between loudspeakers was a little greater than that of the control room monitors. Listeners had no difficulties in perceiving sounds emanating from each of the four loudspeakers. Vertical, horizontal, and diagonal panning were also very perceptible.

When stereo recordings were played through the FyF System, the effect was a pleasant enlargement of the normal stereo experience. In order to play stereo recordings through the FyF System, the normal left and right signals were paralleled so that whatever was in the Bottom Left loudspeaker also appeared in the Top Left loudspeaker and whatever was in the Bottom Right loudspeaker also appeared in the Top Right loudspeaker. Thus, it is possible to listen to stereo through the top loudspeakers or the bottom loudspeakers or both pairs of loudspeakers. Most listeners preferred the stereo signal played back equally through all four loudspeakers with proper left-right separation. When only the front channels of conventional quadraphonic recordings were played through the FyF System, the results were the same as the results from stereo recordings. When the rear signals of a conventionally mixed quadraphonic recording were properly decoded and played through the top loudspeakers and the front signals were played through the bottom loudspeakers, the image destabilized and shifted around unpredictably. Conventionally mixed surround quadraphonic recordings were not judged to be stable enough to be considered accurate reproductions of the source material when played back in a vertical fashion. These recordings sounded best only when the front channel information was played back through all four loudspeakers in the same way that a conventional stereo recording was played back through all four speaker systems. If the front channel information is well-preserved, as it should be if it is stereo-compatible, a surround quadraphonic recording should sound the same as a conventional stereo recording when the front channel information is played back through the FyF System. Our listening tests confirmed this to be true in most cases.

It should now be evident that the FyF System is truly compatible with all commercially available recordings since the FyF quadraphonic field can easily collapse to form the traditional narrow field of stereo or the point source of mono. Our listening tests confirm that an additional benefit of having all four loudspeakers in front is that all four loudspeakers couple together to help the system reproduce low frequency information.

TOWARDS THE FUTURE

The FyF System opens up a number of interesting areas for research. In the near future we hope to design and run several well controlled experiments to supplement the informal tests run for this article. Some questions that remain unanswered are: What miking techniques are best suited for the FyF sound field? What are the best ways to generate ambient information from four loudspeakers in front of the listener? Which encoding-decoding system (SQ, QS, or CD-4) works best with the FyF System? What kind of loudspeaker systems work best with the FyF System? How can digital processing techniques be best applied to FyF recordings? What provisions should there be on recording consoles to facilitate FyF System recordings? These questions and many more are being researched at FyF Studios, Inc. In the meantime, several album presentations are being developed and will be available to the public in the near future. It is hoped that other people will give the FyF System serious thought and possibly contribute energies to further development of the concepts presented in this article. As my good friend in the hinterlands says: "What it is is what it does and what it does is what it is; whatever a thing does do defines what that thing is."

I would like to express my appreciation to the following individuals for their encouragement, time, and concern in this project:

ERIC H. BROWN, Schoolkids' Records, Inc.

JOHN POLLOZOLLO, Hinterlands

TIM PRITCHETT, Columbia Records

BEN MURRAY, Custom Sound of Athens, Georgia

WADE MURDOCK, Dog

Additional Reading:

J. R. Ashley, ON THE PSYCHO-ACOUSTIC BASIS FOR TWO AND FOUR CHANNEL HOME MUSIC SYSTEMS – preprint No. B-6, 53rd 'AES' Convention, Zurich, Switzerland 1976.

M. T. Putnam, THERE WILL BE A SLIGHT DELAY – R-e/p, Volume 3 - Number 2 1971.

FIGURE 3: The FyF listening room (studio) monitoring system.

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

QUAD-EIGHT's "COMPUMIX III"

In 1972, when Quad-Eight introduced automated mixdown to the recording industry with the Compumix I, the reactions of working engineers and producers could only be guessed at. The system was, essentially, a trial balloon, developed to determine whether automation was indeed "an idea whose time had come". For this reason, Compumix I was designed as an add-on for existing consoles requiring no modifications or rewiring, and was made as simple and portable as possible, in the hope of gaining maximum exposure and use.

The response to this first effort led to the development of Compumix II, a permanently installed console modification which had the advantage of a bus-organized control architecture to greatly reduce the amount of wiring in the system. Operationally, however, it differed little from its predecessor. The two dominant characteristics of both were a straightforward sequential scan of the console, with the setting of every control recorded during every scan, and the use of the master audio tape as the medium for storing the automation data.

As is now well known, the first characteristic limits the number of functions that may be automated due to the necessity of repeatedly storing redundant data from the majority of controls that are seldom if ever changed during a mix. This approach results in unacceptably long scan times when many functions on large consoles are to be automated, and for this reason both Compumix I and II were intentionally limited to control of only the faders.

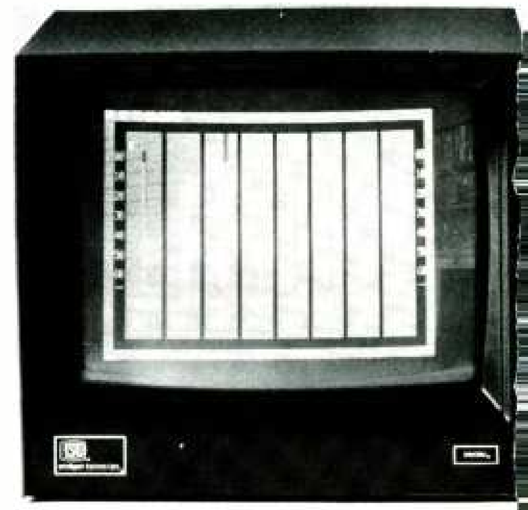
The second characteristic (storing data on the audio tape) produces the irritating phenomenon of control lag buildup, wherein successive update passes accumulate increasing control time delays due to the nonzero time required for the decode/encode cycle. Furthermore, the inevitable minor tape dropouts necessitate elaborate data encoding methods to make the system reasonably error-free.

With the increased availability and decreased cost of digital microprocessors and associated peripherals, it has recently become feasible to implement software-controlled automation systems, with none of the above drawbacks and the potential for hitherto undreamed-of flexibility and sophistication. The possibility of instantly "reconfiguring" an automation system by the simple process of changing the control program is a powerful one indeed, and will no doubt keep hardware manufacturers busy for years to come exploring its implications.

The immediate problem, however, is how to make useful improvements in the design and operation of existing automation systems, without violating certain constraints (some of which have come into focus only in the course of using the previous generation of equipment). The most significant of these constraints appear to be:

1. Familiarity. It is essential that an automation-assisted system look, feel, and operate like a conventional manual console insofar as this is possible. Ideally, the user should hardly be conscious of its presence. Distractions should be held to a minimum, and above all, the temptation to create 2001-style panels replete with banks of lights and buttons labeled

. . . the 'C3' CRT terminal shown displaying fader positions . . .



in esoteric computerese must be resisted. In short, a mixer should not have to learn the art all over again.

Conceivably, overwhelming technological imperatives could someday arise that would dictate a revised approach to console layout, but such propositions today must be considered speculative and the thorny problem of standardization remains to be dealt with.

2. Dependability. In a well-designed automation system, the weakest link will always be the data storage medium. From this standpoint, audio tape is the least desirable, since audio recorders were never intended for digital data. Auxiliary digital tape cartridge units specify an error rate of typically 1 in 10^7 bits, a usable reliability figure but some 100 times worse than the Schugart diskette used in third generation Quad-Eight systems. Further, the disk's high speed read and write capability allows most errors to be recovered by re-reading the data one or more times before it is actually needed by the system. With this technique, the bit error rate can be reduced to an insignificant 1 in 10^{12}, which for practical purposes is error-free.

3. Utility. This criterion is the least well-defined of the three, but is certainly no less important. In a negative sense, it may be defined as a freedom from "bugs" and restrictive limitations. The positive side of this coin should be the inclusion of an abundance of convenience features which are easy to learn and use, and which make mixing easier or faster or more precise. Here, it is felt, is where automation remains to make its greatest strides.

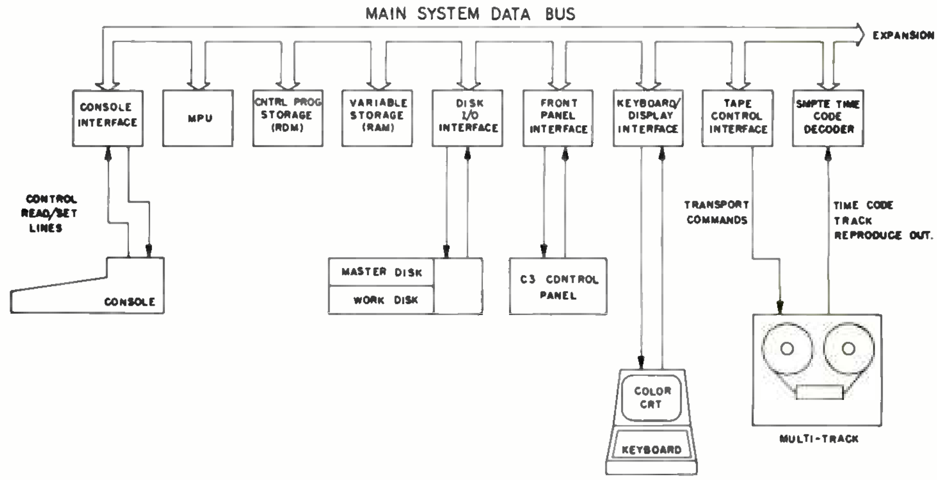


FIG 1.
COMPUMIX III HARDWARE ORGANIZATION

COMPUMIX III SYSTEM DESIGN

The Quad-Eight Compumix III (or "C3") system consists of four principal components: the console itself, the C3 processor (equipped with a small array of pushbuttons for calling up various utility functions), a dual disk drive containing the "master" and "work" data disks, and a *color* video display terminal and keyboard. The CRT terminal, a unique feature of C3, is included not only for convenience in sending written messages between the processor and the operator, but also for its extreme graphic display flexibility. The terminal contains its own microprocessor, which is programmed to control the display of fader positions, equalizer curves, audio levels, real or elapsed time, etc. It has the advantage of being completely silent in operation and the ability to present a wide variety of information in a format that can be absorbed at a glance.

Figure 1 gives a somewhat more detailed picture of the layout of Compumix III. Like most microprocessor systems, the various components in C3 are tied together by a common data bus, which is time-shared on a priority basis. All data exchanges between all components take place via the bus, which services the following subsystems in addition to the microprocessor (MPU):

– The control program storage, which consists of ROM (read-only memory) to prevent accidental erasure or modification.

– Random access memory (RAM), used for the storage of temporary or changeable data.

– The disk I/O interface, which allows the disk to read data out of or into the RAM without constant attention from the microprocessor, via a Direct Memory Access feature.

– The console interface, which converts control settings into digital data capable of being processed by the system, and vice versa.

– The front panel interface, servicing the indicators and controls on the C3 control panel.

– The display interface, which allows communication from keyboard to processor and from processor to CRT.

– The tape control interface, which enables the system to stop, start, and rewind or fast forward the master tape.

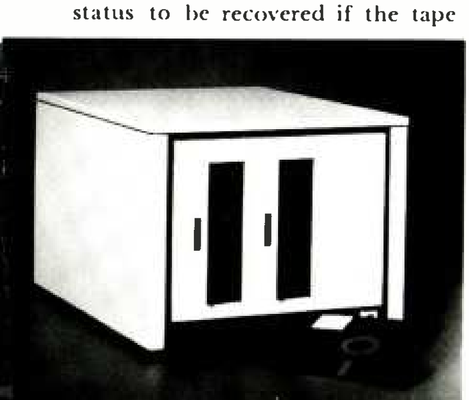
– The SMPTE time code decoder, which supplies signals to synchronize the entire system with the audio master tape.

This last feature, the SMPTE time code decoder, is the key to the operation of the disk-based C3 system, since it allows absolute synchronization between console control signals and the audio signals at all times.

SMPTE code is a television industry standard digital code formatted in minutes, seconds, and TV "frames" (which occur at the rate of 30 per second, or one every 33.3 milliseconds). Using a special Quad-Eight encoding technique designed to ensure reliability, this code is recorded on one track of the audio master tape - - either during initial music tracking, before mixdown, or during the first trial mix. Since the SMPTE code requires an audio bandwidth of less than 3 kHz, the possibility of leakage onto adjacent tracks is greatly reduced. In addition, 60 Hz subtone pulses are mixed with the time code when it is recorded. These can be read at high tape speeds and allow automatic tape positioning in the fast wind mode.

Once the time code is recorded on the master tape, all further system timing is derived directly from it.

Each "frame" in the code causes an interrupt signal to the microprocessor to be generated, triggering a complex sequence of events. The processor first updates its internal clocks, then checks the highest priority task, which is the disk service routine. After any pending disk I/O operations have been handled, the keyboard/display terminal is then similarly serviced. Next the channel service routine is activated, a program that scans the console comparing the settings of all controls with the settings recorded during the last pass of the tape. As changes are found, they are entered in memory for later saving on the disk. After either all changes have been noted, or half the available frame time has elapsed, the processor switches over and begins memorizing settings of controls that have not changed since the previous frame (these are necessary to allow full console status to be recovered if the tape is started in the middle).



. . . the 'C3' SCHUGART diskette component.

Once a full frame of data has been acquired, the processor returns to its low priority "background" duties until the next frame interrupt occurs. Using this interrupt method, a single-processor system can service approximately 60 console modules per frame. Only when there are an extremely large number of changes will any data be "held over" until the next frame. Thus virtually all control changes will be recorded within 33 milliseconds or less.

This data from the console is not immediately recorded on the disk. Instead, it is accumulated in a "buffer" area in main memory for approximately 1 second (actually 32 frames), then block-transferred to disk. Similarly, data is read from the disk in 1 second blocks; however, due to the predefined disk storage allocation format, the processor can quickly locate the data for any given frame, given only the frame's unique SMPTE time code.

Note that in the operating mode just discussed, console data is read from a specified disk file, updated, then rewritten in the *same* area of the *same* file. This need not be the case, however; by entering the appropriate commands on the keyboard the user can instruct the system to create a brand new output file on a different area of the disk. This new file will contain the information from the previous mix, plus whatever changes were entered during the last pass of the tape.

This capability allows a rough mix to be recorded on the "master" disk, then several variations of it to be stored on the "work" disk and compared with the original. At this point, C3's sophisticated editing facilities may be called into play. By noting the starting and ending SMPTE times of the best sections of each of the several trial mixes, a "cue sheet" can be developed specifying how the final mix should be pieced together. This cue sheet is then typed in on the display keyboard, and the "Audition" button pressed. C3 rewinds the master tape to the start of the selection, and plays back, switching the console settings back and forth between "takes", exactly as specified on the cue sheet.

Once the mixer is satisfied with the assembled mix, the "Copy Audition" button may be pressed, which causes the final mix to be recorded in a single continuous disk file on the master disk. The master disk may then be removed and slipped into the storage box with the master tape; additionally, a safety copy of the mixing data can be created with the "Copy Disk" button.

Both the master and work disks contain enough storage for over 20 minutes of mixing - - longer than most album sides (if necessary, a long program may be contained on two disks). Each disk may contain up to 8 different files of mixing data, each with its own unique name and a revision number which is automatically incremented by the system each time the file is rewritten (updated). These names may be examined and/or modified at will on the CRT terminal with the aid of the disk directory maintenance pushbuttons.

Mention has been made of the system's automatic tape re-

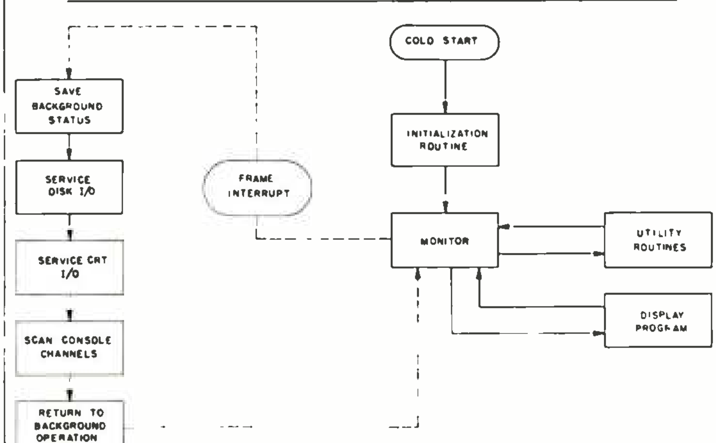


FIG 2
COMPUMIX III PROGRAM FLOW CHART

wind. In fact, storage is allocated for up to eight mark points on the tape, which may be automatically located on demand, using the "Cue forward to next mark", "Cue back to last mark", and "Cue to mark no. n" functions. C3 will fast wind to within two seconds of the desired spot on the tape, then switch to "play" to find the exact SMPTE frame requested.

OTHER FEATURES

Background Processing. As shown in Fig. 2, the C3 processor has a variety of low priority tasks to execute when not performing the interrupt service function. These include the various utility functions (called up either by a front-panel pushbutton or as needed by C3) such as tape handling, disk file "housekeeping", etc, plus the routines which acquire and format data to be displayed on the CRT.

Dedicated Console MPU's. When extremely large consoles are to be automated, it is possible to use an independent microprocessor solely to handle the console channel service portion of the interrupt task (see Fig. 2). This technique more than doubles capacity, and accommodates consoles of well over 100 modules.

Tape Storage Systems. Like its predecessors, Compumix III can function without the use of data disks, by recording the console data directly on the audio tape instead of the SMPTE time code. In this mode, the addresses and settings of the console controls are converted to a special high reliability code and written on the tape as they are picked up from the console. The data formats are quite different for tape and disk, but a C3 translating program can convert back and forth between them when required.

"Working Master" Capability. When it is desired to avoid wear on a master tape during long mixing sessions, a "working copy" of the multi-track tape can be used. For the final mixdown, the same data disk generated with the working master will duplicate the control functions exactly on the original, since both versions contain the same time code.

CONCLUSION

Quad-Eight's Compumix III represents the latest effort to implement automation-assisted mixing using the three criteria of Familiarity, dependability, and Utility stated at the outset of this discussion. One particularly attractive feature of C3 is its "open-ended" capability for future refinement, a consequence of a flexible architecture and software control. It is expected that eventually features will be added to the system that have not even been conceived yet - - an exciting prospect for audio engineers and mixers.

Understanding . . .

Magnetic Tape PERFORMANCE SPECIFICATIONS

by Peter Butt

It wasn't so very long ago that those spec sheets that came along with recording tape evaluation samples were fairly simple lists of numbers that conveyed their meanings in a very succinct way. They were nice and neat. Unexciting. Comfortable. No pictures at all, unless we count a possible photo of a reel of the product included somewhere near the beginning of the sheet. Recording tape was such a simple matter in those days. Only the inclusion of a page or two of explainatory notes set in type befitting the archetypical life insurance policy could possibly mar the serenity presented up front.

One of the sub-corollaries of Murphy's Law states that nothing is as simple as it appears to be. The practice of summarizing the properties of a magnetic tape in one or two tabulations of figures tends to minimize the fact that a number of conflicting properties and requirements must be optimized in some way to yield an acceptable tape performance compromist. Upon closer examination, one might conclude that tape, having the properties it has, within the tolerances necessary for the production of reasonable quality recordings, may be impossible to manufacture. Murphy seems to have scored another cosmic truth.

Some of the mystery surrounding the performance properties of magnetic tape began to part a couple of years ago when 3M introduced its 250, high output mastering tape. The 250 spec sheet adopted a format for data presentation resembling the approach used by European tape manufacturers for years. This general direction has been continued by 3M in its data presentations describing its other magnetic tape products. Other manufacturers of professional recording tapes have similarly taken the cause to heart, Magnetic tape data sheets exhibiting those tall, skinny graphs with undulating curves are now commonplace on both sides of the Atlantic.

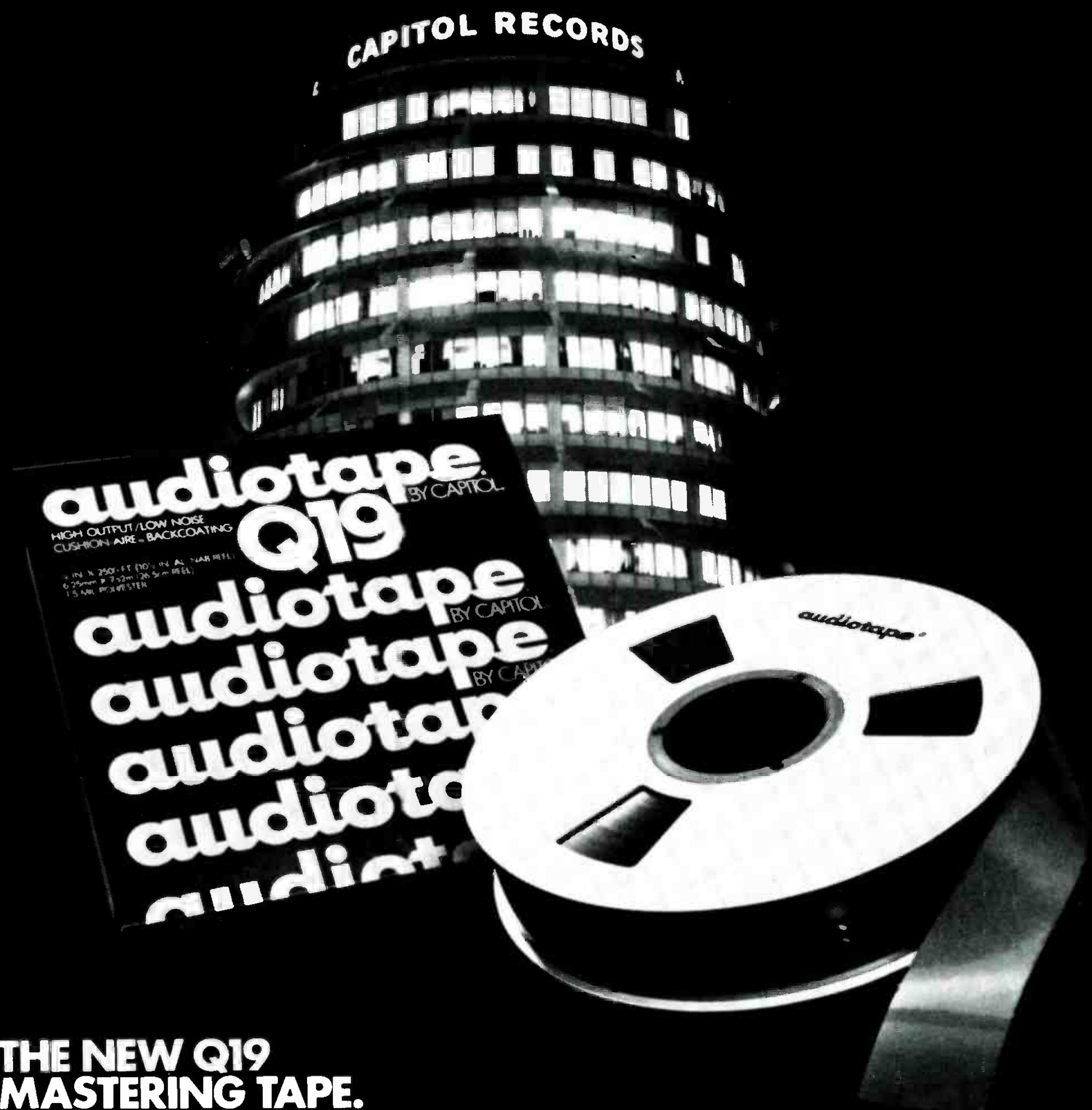
The basis for the new data presentation format is the *Deutsche Industrie Normen* (DIN). This is the German standard for measurement of the mechanical and electro-magnetic properties of recording tape that has been the standard for European specifications for years. It has the advantage of permitting fairly direct comparison of manufacturers' stated performance characteristics without the need for the prospective buyer to re-verify the measurements for himself. The use of a common method for performing the measurements as well as a common format for their statement has definite advantages whether one agrees with the tests themselves or not.

Granting that any move toward standardization of performance specifications is a step in the direction of enlightened consumerism, it may be well to look the horse closely in the mouth before formal declaration of the millenium. The age of darkness is so recently passed from us that there may yet linger a shadow of uncertainty among the body technic.

UNITS

A cursory glance over the data summaries for the four major marketers of professional mastering tapes in this country reveals that all are evidently not in as close accord as we might have led ourselves to believe. There is the basic question of uniformity of units used for statement of the properties.

Introducing a new formula for successful recordings.



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FORMULA 19 · Consistent and Reliable · Compatible · High Output · Excellent Headroom · Low Print-Through · Drop-Out Free · Cushion-aire™ Backcoating · Splice Adhesion

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Circle No. 126

This matter of units should be a sensitive point for past or present students of physics among the readership. Electromagnetic properties of materials have been expressed in terms of centimeter-gram-second electro-magnetic units (cgs-emu) and in meter-kilogram-second units (mks). The latter system of units has been standardized by international agreement since 1960 and is widely known as the Systeme Internationale, Giorgi, or SI system of units.[1] Conversion of units from the somewhat inaccurately-based cgs-emu to the SI units can be a rather sticky wicket.[2] The conversion factors are not always simple magnitudes of ten but frequently incorporate pi or its reciprocal. It's bad enough where only results are to be converted from one system to the other. The chore assumes enormous proportions if a detailed mathematical derivation is to be converted. The units in each system have different names for the same physical properties. Keeping track of Maxwells versus Webers, Gauss versus Teslas, Gilberts versus Ampere-turns, and Oersteds versus Ampere-turns per meter contributes to confusion at an elementary level that complicates understanding of information of direct utility.

Agfa has taken the trouble to state its PEM 468 performance figures in both cgs and SI units. Ampex and Capitol follow this generally, but revert to the cgs system for statement of the coercivity and retentivity of their respective 400 series and Q 19 tapes. 3M retains the cgs system completely, making comparison with its counterparts much more difficult than it need be.

Coercivity gives a rough measure of the required optimum bias level a tape requires. Retentivity tends to indicate the maximum magnetic flux that can be retained by a tape after the magnetic flux induction has been removed. How well these maximum values relate to the actual recording performance of the tape depends upon how skillful the manufacturer is in managing his manufacturing processes and optimizing conflicting parameters.

THE GRAPHS

Tape performance is a very complicated matter. The many aspects of it comprise a highly dynamic, multi-dimensional picture that changes considerably as a function of variations of the bias drive level, recorded signal wavelength, record head gap, bias signal spectral purity, and coating properties. Acquiring a manageable mental picture of all these simultaneous interactions is not at all easy. Hence the motivation for this article.

Because of the complexity of the behavior of any magnetic tape and the fact that most people are not in a position to treat electro-magnetic phenomenae and properties of materials with the same familiarity typical of discussions of football scores, it might be worthwhile to examine the matter more closely.

For a given tape machine, at a given speed, variation of the bias drive level will cause the change of tape properties very significantly. This is true to such an extent that the bias drive is the primary causitive force in tape performance. For this reason, the bias drive level is shown as the horizontal axis of the familiar skinny charts. The vertical axis shows the magnitude of remanent magnetic flux observed by the reproducing head for the range of recording bias drive levels shown horizontally. The implication of this choice of axes is that all of the properties plotted on the vertical axis are dependant upon the quantity shown on the horizontal axis. The quantity plotted horizontally is referred to as the "independant" variable while the vertically-plotted axis is called the "dependant" variable. This assignment of axes is a widely accepted convention for the display of mathematical and physical functions.

The vertical axis is most generally graduated in deciBel increments to cover the wide dynamic range that remanent tape fluxes exhibit. The zero reference level taken for this deciBel scale is some level close to what would be considered a reasonable operating level for the tape being represented. The choice of this level is not universal among manufacturers. Agfa, Capitol, and 3M have chosen the DIN reference level of 320 nanoWebers per meter while Ampex uses 260. The difference amounts to about 1.80dB. Failure to take this difference into account will tend to make Ampex residual noise, modulation noise, and printing performance look worse than that of its counterparts while showing apparently superior performance in the areas of distortion and maximum remanent magnetic flux.

The information presented on the graphical plot is affected in still more subtle ways by the measurement techniques employed to generate the data. The DIN standard 45 512 dated February, 1972 defines the methods to be used in presentation of performance data in fairly strict terms. It appears that there is no general policy among American tape manufacturers to adhere strictly to the DIN standards. A close reading of the explainations of details contained in the respective data sheet notes is necessary to retain a perspective on the applicability of comparisons of the information presented. The large print giveth and the fine print taketh away.

THE DIN STANDARD

The latest modifications of the DIN standards applicable to magnetic tape performance measurements were made in February, 1972. Translations of the

our world is about to pay you a visit.

You may have expected a glossy four color picture with this Eastlake ad—and we have them to show in a new twenty-two page brochure talking about our studio design.

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his product. These test conditions selected may differ from the conditions encountered in normal usage sufficiently to make extrapolation of test performance to application a highly intuitive matter.

Our earlier inspection of the explanatory notes sections of our data sheets have shown that for all the similarities in outward appearances, there are basic differences between measurements that make direct comparison between data summaries and curves more difficult than we would like. What is needed is a set of general rules that can guide our expectations of a given tape's performance when applied in a real-life situation.

This is not an impossible goal. Although precise prediction of tape performance test results in one situation, based upon results obtained under other conditions, probably not possible at this time, we may be able to develop a few rules of thumb that can lead our expectations in the proper direction. One of the more apparent differences in measurement conditions between manufacturers is the width of the recorded track used. Track widths vary from 6.3 mm (0.248 inches) to 2 mm (0.080 inches) to 1.8 mm (0.070 inches). As we're all aware, this variance in track width has a very important effect on the signal-to-noise ratio of the signal channel. Agfa has given most prominent recognition of this fact in the form of a curve that permits interpolation of S/N ratios for track widths other than that specifically used for the PEM 468 data. A formula that will yield fairly accurate values for the variance of the signal-to-noise ratio with track width is given here:

$$N = 10\log_{10}\frac{W_1}{W_2}$$

where: W_1 *is the reference track width*

W_2 *is the track width of interest*

N is the number of deciBels to be added to the reference S/N ratio to arrive at the S/N of interest

It is important for the reader to understand that the noise contribution of the electronics is not taken into account in the expression. The tape speeds for the two cases are, of course, presumed to be the same.

As for the matter of bias data, that gets a little more complicated. Dependent as it is upon the dimension of the recording head gap, there is no way to accurately determine biasing behavior from one test case to another by purely theoretical means. There have been attempts to predict this but none seem to have been generally successful and most attempts require rather massive computational capability to be used in any event. The general rule is that the bias current versus remanent flux sensitivity peak assumes a sharper characteristic as the record head gap dimension decreases. Also, the relative peaks at medium and short wavelengths tend to diverge as the record gap becomes narrower. The sensitivity peak also becomes sharper, for a given record head gap dimension, as the recorded wavelength decreases.

An illustration of this effect is given in the Ampex 406/407/456 data sheets, numbered T1205-5-75 and T1167-5-74, in the form of a tabulation of the amounts of overbias necessary at 10 kHz for recording heads of different gap length dimension. For the Ampex tapes, the amount of overbias necessary increases from +1 dB at 10 kHz for a 25.4 um (1.0 mil) gap to +3 dB at 10 kHz for a 6.3 um (0.25 mil) gap. These figures apply to a 381 mm/sec tape speed. Still more overbias would be necessary for the case of higher tape speeds, less for lower tape speeds.

Users of other tapes may find it useful to construct bias current versus remanent flux curves for their own application. One way this can be done is to determine the relative shapes and positions of the medium and short wavelength sensitivity curves for the machine to be used. The bias current data can be read either from the machine VU meter or from appropriate test points in the case of most professional recording equipment in wide use. The user is cautioned to perform his measurements at recording levels about 10 dB below the normal reference level used for his application. Comparison of the plotted results with the curves given on the tape specification sheet whould give an idea of how much overbias is necessary to approximate the recommended operating point given by the manufacturer.

The relative positions of the medium and short wavelength sensitivity curves should move farther apart for the case of a narrower recording gap as far as their absolute peaks are concerned. In other words, for a smaller recording gap, the shorter wavelength sensitivity peak will tend to shift to a lower bias current point relative to a longer wavelength sensitivity peak determined with the same head. For a large recording gap dimension, the medium and short wavelength sensitivity peaks will tend to coincide. For an illustration of this trend, the reader is referred to the two versions of the Ampex 406/407/456 data sheets cited earlier. The data presented in the four-page, four-color, shiny-paper folder was derived using a 6.3 um (0.25 mil) record gap while the more plain data sheet presents data derived from a 25.4 um (1.0 mil) record gap.

The prime information we are after here is the relative positions and relative steepness of the respective sensitivity peaks as observed on the tape machine of interest as compared with those measured by the tape manufacturer. Having determined this information, we can set about determining the behavior of the third harmonic distortion function at operating level with respect to the bias current and plot this on the same graph with the relative sensitivity data. We should now be able to see how the tape's performance on a familiar machine compares with that used by the tape manufacturer. We will also be able to determine the amount of overbias necessary to achieve minimization of third harmonic distortion at operating level for our specific case. Clearly, another parameter of particular interest may be chosen for this exercise.

Consideration of the time and trouble involved in the derivation of the several curves mentioned immediately above will lead to an appreciation of the time required for generation of the families of curves displayed in tape specification literature. Reproduction of the entire set of curves is a task obviously reserved for those with more time at their disposal than most people are willing to admit publicly. In fact, it takes two or three days to derive the raw data for a set of curves.

Once the basic biasing information is determined for a particular recording system, the user can fairly confidently adjust his bias drive levels to the proper amount of sensitivity reduction for his application. The empirical approach to specific application strategy is valid with respect to other performance characteristics as well.

MODULATION NOISE

Attention to the curve variously labeled "DC noise," "modulation noise," and "intermodulation noise" may be beneficial. This characteristic is supposed to give an easily measured indication of the noise that is introduced into the signal by the tape when a signal is recorded on it. DIN 45 519, sheet 2 specifies the method to be used. Briefly, the rms value of the record-head current required to yield a remanent flux of reference level on the tape sample is measured. A DC current having the same value as the measured rms current is passed through the head and recorded on the tape. The resulting DC signal is then reproduced and measured through a weighting filter. The value of this reading is then reported.

Modulation noise may be observed by feeding the reproduced output of a continuous sine wave recording into an audio spectrum analyzer having a high degree of resolution and dynamic range. Instead of the ideal narrow, smooth peak obtained from the signal source directly, a peak

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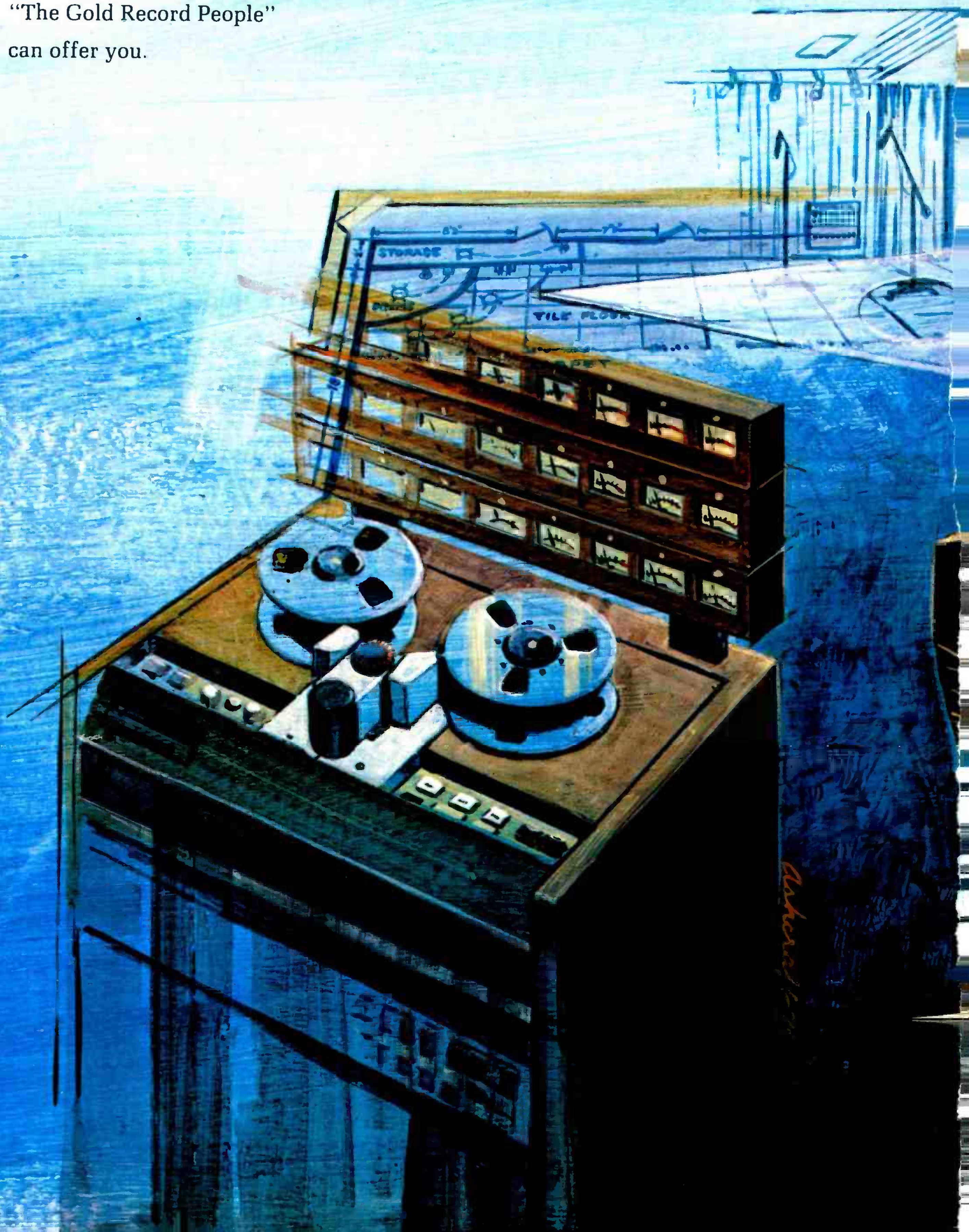


latest changes into english are very scarce. Basically, standard 45 512 part 1 applies to mechanical properties of the tape while 45 512 part 2 applies to the details of the measurement methods and the format of the summary of performance characteristics. Standard 45 513 defines the characteristics of the alignment tapes to be used in tape measurement system calibration. Standard 45 511 specifies the characteristics of professional tape recording and reproduction equipment. Standard 45 519 specifies methods for measurement of residual DC noise and print-through. There does not seem to be any agreement as to the format or other details of data presented graphically, or even whether such a presentation is deemed necessary.[3]

The basis of all the measurements described in 45 512 is the flux-frequency response characteristic of the reference tapes specified in 45 513. There is a section of the reference tapes conforming to DIN 45 513 that is used as a baseline for establishment of the relative bias levels and sensitivity data appearing in the graphic and tabular data presentations.

Strictly speaking, the 0 dB bias current point on the horizontal graph axis corresponds to the bias current value necessary to overbias the unrecorded portion of the DIN standard alignment tape by 2 dB beyond the maximum sensitivity peak, at a modulation frequency of 10 kHz and a tape speed of 381 mm/sec. Agfa and Capitol use this reference point for all of their bias data. Ampex uses a bias reference point of 1 dB beyond the 10 kHz maximum sensitivity peak at the 381 mm/sec tape speed. The tape used by Ampex,however,is standard tape MTDE 320D in Ampex publication T1246-8-75 and MTDED-315-2 in publication T1205-5-75. 3M does use the DIN reference tape unrecorded reference section for reference bias current but uses the geometric mean of the bias currents corresponding to the 0.5 dB points above and below the maximum sensitivity peak at a modulation frequency of 1 kHz and tape speed of 381 mm/sec.

As a general rule, tape sensitivity versus bias drive level tends to exhibit a sharper peaking characteristic as the wavelength of the modulation signal decreases. In addition, there is a tendency for the absolute bias current at which the maximum sensitivity peak occurs to shift downward somewhat as the recorded wavelength decreases.[4] This amounts to yet another variation in the absolute value of the bias reference point corresponding to 0 dB on the bias current axis.

Obviously, the absolute value of the bias current required to achieve a specific bias condition in a given sample of tape will vary from machine to machine. The magnetic flux field necessary to achieve that specific bias condition, for record heads of similar geometry and magnetic properties, should be very similar, however. Therefore, proportionate deviations from the bias level corresponding to that reference flux field condition should result in similar biasing changes in those similar, but different, heads. That's the whole name of the game here. Presentation of data in a way that allows a prospective tape user to obtain an idea of the ways different tapes compare with one another without the need to resort to several days of testing on his own tape machines.

Further perusal of the tape data presentations at hand reveal yet another obstacle to uniformity of magnetic tape performance data. The gaps of the record heads used for tests of the different tapes differ with respect to the gap length dimension. DIN 45 512 specifies an 18 um (0.7 mil) gap. Agfa uses a 7 um (0.276 mil) record gap. Ampex uses a 25.4 um (1.0 mil) and a 6.3 mm (0.25 mil) gap. Capitol Magnetics uses a 15 um (.6 mil) gap, 3M, a 25.4 um (1.0 mil). One might reasonably ask, "What's a micro-meter between friends and colleagues of like persuasion?" As far as individuals are concerned, probably not a whole lot. In the case of a recording head, the dimension of the gap length is instrumental in the determination of the distribution of the recording bias and modulation signal flux distribution through the tape oxide. The narrower the recording gap, generally speaking, the more difficult it will be to bias the particles of oxide that lie away from the exposed oxide surface, within the oxide layer. Thus, the depth of the recording within the tape oxide layer will tend to be more shallow for a narrow gap head than for a wide gap head. The long wavelength output level achievable for a given amount of distortion will tend to be lower for a narrower recording gap and a given tape oxide thickness. The shape of the sensitivity bias peak will tend to be sharper for a narrow gap head also. The implications for sensitivity of the tape to bias level drift should be fairly apparent.

We have, then, a case where similar properties of different tapes are being measured under conditions that may look fairly similar upon cursory inspection. The present state of the theory of the physics of the recording process does not permit precise estimation of the influence these factors have on the respective sets of data, other than in a rather general way.

With sufficient intuitive understanding of the way a given tape could be expected to perform in a given practical situation. For example, data taken with a 25.4 micro-meter recording gap would not directly indicate the exact behavior of the same tape sample used on a mastering machine having a 7 micro-meter gap. Narrower record head gaps have become the rule in studio tape machines of recent vintage, because of the demand for very flat synch playback frequency response. The sacrifice in recording performance due to the choice of a non-optimal recording gap dimension is probably a fair price to pay for the highly-prized synch response.

THE ROLE OF INTUITION

The tape manufacturer has a justifiable interest in presenting the performance characteristics of his product in the best possible light. To do this he may choose to generate his measurement data using means that take best advantage of

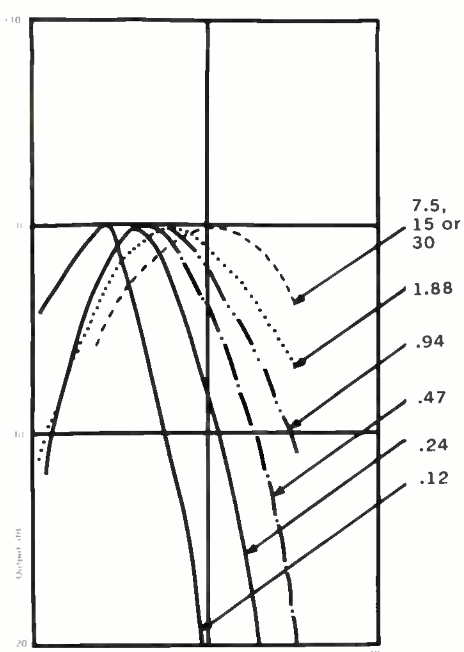


Figure A. Output versus bias current plotted with amplitudes of maximum output normalized to 0 dB output. These curves show the shift of bias current required for maximum output at shorter wavelengths. Wavelengths shown in mils. After McKnight, reference 4.

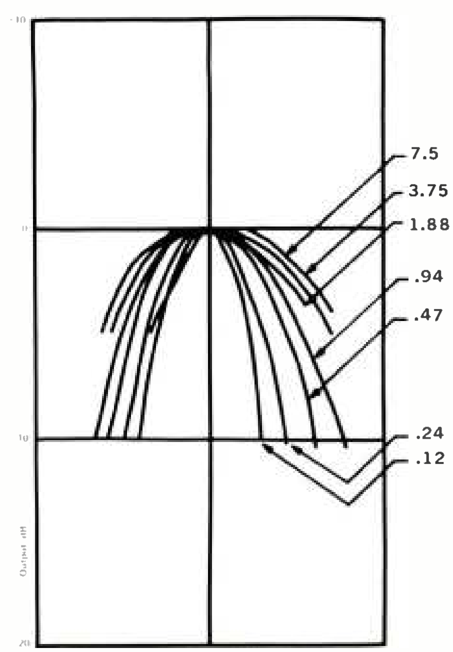
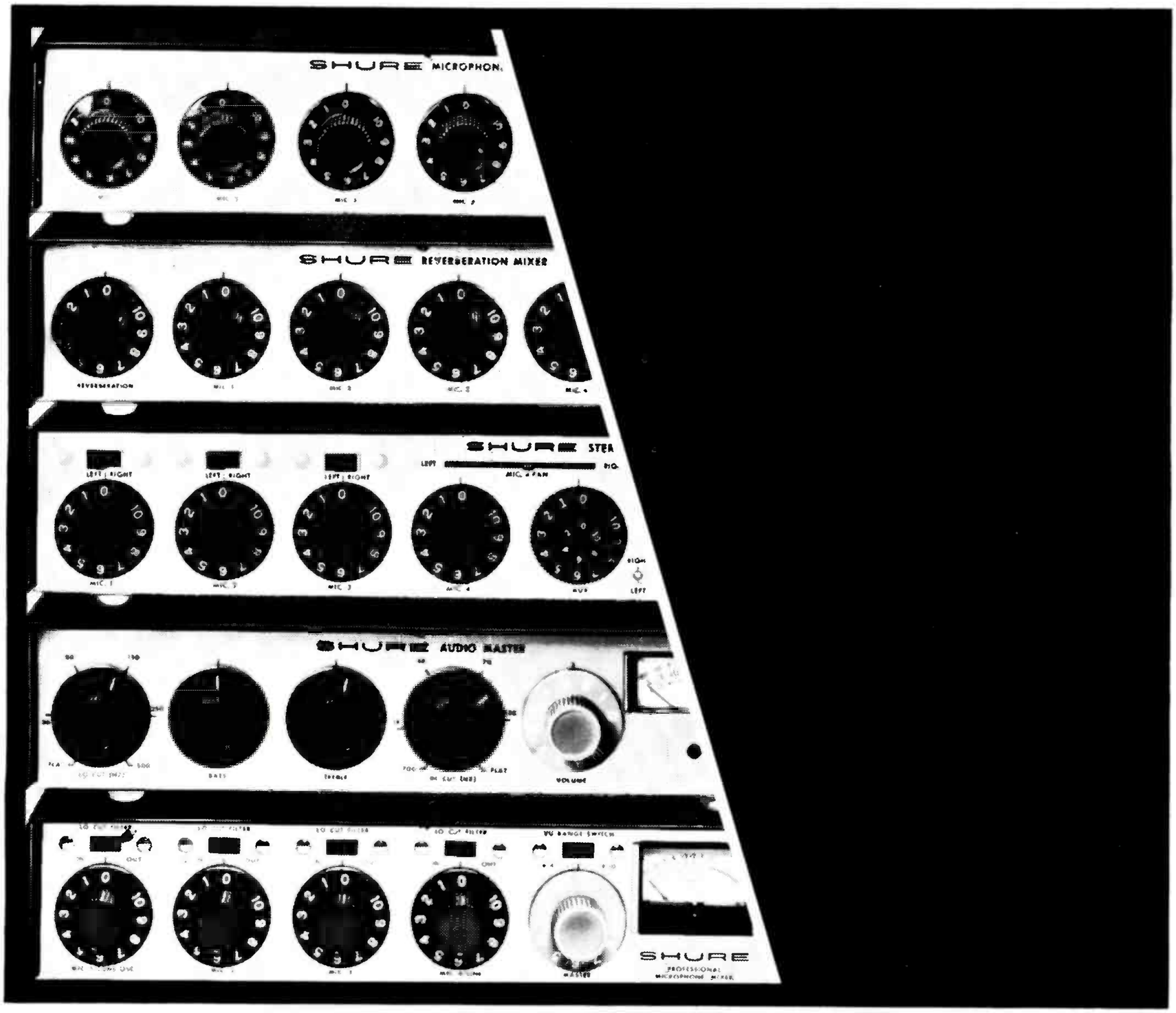


Figure B. Output versus bias current curves. Data plotted with maximum and bias currents normalized to 0 dB to show the sharpening of the sensitivity curves with decreasing wavelength. Wavelengths shown in mils. After McKnight, reference 4.



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having a wider skirt and adjacent noise peaks is observed. A method of modulation noise measurement using spectrum analysis techniques has been proposed and appears to be conceptually more closely related to the phenomenon of noise accompanying a modulating signal than the equivalent rms DC signal method.[6] The technique consists of direct measurement of the maximum height of the modulation envelope of a 1 kHz recorded signal relative to the height of the noise skirt at 800 Hz. The ratio of the two heights in deciBels would be reported at the modulation noise ratio.

The equivalent DC approach to modulation noise measurement would seem to be somewhat further removed from this troublesome phenomenon as encountered in application than the relative skirt-height method. The latter would seem to be a bit more directly representative of the way that the modulation noise effect is perceived by the ear.

There is an alternative approach to the problem of modulation noise observation that ought to give reasonably consistent readings between samples. If a recorded sinewave signal is reproduced and passed through a harmonic distortion meter, the distortion meter may be tuned in such a way that the sine wave is cancelled out and only the other signal components can be observed.[7] The problem with this method is that the harmonic distortion components of the original sine wave are included in the reading unless they are low-passed out and the resulting signal is read with an AC voltmeter.

The importance of modulation noise has increased in recent years as the use of noise reduction systems has become more common. Noise occurring only in the presence of the desired signal cannot be reduced as efficiently as the general system residual noise can. Modulation noise occurring in time and frequency coincidence with the modulating signal presents a rather interesting problem in signal-versus-noise differentiation. The contrast in background noise during the presence of modulation compared with the low noise level during periods when the modulation is absent can be extremely annoying in noise-reduction-processed programs. Recordings of solo instruments or spoken word are particularly sensitive to this problem.

RESIDUAL NOISE

The biased tape residual noise level, as opposed to the bulk-erased residual noise level, is a function of the recording head gap geometry, the bias signal wavelength, and the bias signal spectral purity, as well as of the recording bias drive level itself.[8] The DIN 45 512 specification requires a recording bias frequency of 80 kHz, ±10% for the machine used for derivation of tape performance data. Equipment considered standard professional quality generally have bias frequencies far greater than this. Ragle and Smaller report that the residual biased tape noise tends to decline, asymptotically approaching about 3 dB over the bulk-erased level, as the bias wavelength declines. This makes the value of the residual biased tape noise level dependant on the machine on which it is measured. The higher the bias frequency, the lower will be the observed biased noise level.

Further, the geometry of the recording gap is also a relevant factor. Gaps with rounded edges, and therefore impaired short wavelength recording ability, will exhibit better noise performance than record heads having gaps of better definition. Shorter recording gaps will tend to be more sensitive to the induction of tape noise due to variations of head-to-tape spacing than will gaps of longer dimensions.

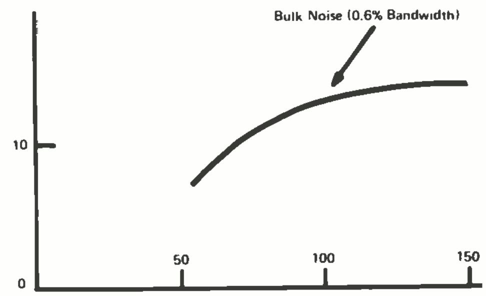

Figure C: Recorded bias signal versus bias signal wavelength, bias current held constant. After Ragle and Smaller, reference 8.

The spectral purity of the biasing signal is important in several ways. First, the presence of odd harmonics in the bias signal will introduce the familiar "bias rocks" effect due to the resulting DC component in the bias signal. The result is equivalent to the recording of a DC signal reminiscent of the DC noise measurement outlined above. This is fairly well known. Secondly, spectral contamination of the bias signal will introduce beating with the modulating information signals recorded on the tape, raising the intermodulation level of the tape channel.

There is no generally accepted figure for the spectral purity of a bias signal used for professional analog recording applications. Tremaine quotes 0.5% as a maximum distortion level.[9] Achievement of very low levels of recorded signal distortion probably require a more stringent specification than this. In any event, this discussion will suffice to sustain the point.

PRINT–THROUGH

Printing of the recorded signal from one adjacent tape layer to another is an annoying phenomenon that has been discussed earlier in a previous issue of Recording Engineer/Producer.[10] All of the manufacturers of professional master-

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ing tapes marketed in the United States use the print-through measurement method specified in DIN 45 519, with the exception of Ampex. The Ampex method differs in that the 1 kHz reference level signal is stored for 24 hours at a temperature of 70 C rather than at 20 C. This makes the Ampex test a more rigorous test than the DIN standard requires.

Again, the recording gap enters into the results of this test as does the temperature at which the tape is stored prior to measurement. The higher the temperature, the greater the printing effect. The greater the recording gap length dimension, the deeper the recorded remanent flux will penetrate into the oxide layer. The deeper the recording penetration into the tape oxide, the closer the recorded signal field will be to the adjacent tape layer and the greater will be the printing effect. Comparison of the Ampex print-through figures for the 25 um and 6.3 um data sheets will confirm this rule.

CONCLUSION

Where does all this leave us? Quite a bit better off than we were prior to the advent of the DIN format of tape data presentation, certainly. Although ignorance and bliss are reputed to be closely associated, it just may be possible that the bitter brew of the cup of knowledge may advance our cause in a more positive direction. The momentum toward a more uniform presentation of tape characteristics can only benefit those of us who use the product.

Even though there are faults and inconsistencies between the differing methods of measurement, at least steps have been taken to present the variables of magnetic tape performance in a less simplistic manner than in the past. The increasing consciousness of the compromises necessary to intelligent application of the medium to the recording of sound must necessarily yield fruit in terms of greater user sophistication and motivation of hardware manufacturers to develop hardware capable of more full utilization of recent advances in tape technology.

It is hoped that this brief discussion has contributed to a more general understanding of the limits and virtues of the performance data presentations now used by the manufacturers mentioned. Examination and comparison of the information available in current specification publications has fringe benefits in a more complete understanding of the factors that affect the quality of the recording process. More general notice of the published data will, in time, promote progress toward more comparable measurement techniques that can be directly compared with one another with confidence. The purpose here has been to point out areas where direct comparison is less than valid. Knowledge of the limits of one's tools is a worthwhile pursuit.

For those who care to pursue the matter of the DIN method of tape evaluation further, alignment tapes having unrecorded sections of the standard DIN reference tape may be obtained from:

BASF Systems, Inc.
Crosby Drive
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John McKnight, Magnetic Reference Laboratory.

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1975's classical recording "GRAMMY" winners, Columbia's

Bud Graham, Milt Cherin, Ray Moore

R-e/p: What was unique about this recording?

Ray Moore: The unique thing about this recording is that it won a Grammy.

R-e/p: Certainly there is more to it than that!

Bud Graham: No, not really. The circumstances concerning the recording sessions involved in this recording were no different than a lot of other sessions we have done at Manhattan Center. Our standard Manhattan Center set-up was used . . . except for the addition of the choir . . . about 60 voices. If you look at the diagram, you'll see we put the choir in the balcony above the ballroom floor.

R-e/p: Isn't it unusual for three engineers to get involved with a classical recording . . . is it necessary?

Bud Graham: No! It probably isn't necessary, but we have been handling our recordings that way at CBS for years.

R-e/p: Could you give us some idea of how you divided up the duties . . . on what your interaction was . . . were there others involved?

Milt Cherin: Well, Bud handled the control work . . . the studio recording on three recording sessions that were necessary for Daphnis.

Ray Moore: Milt did all the splicing of the original tapes necessary to achieve a complete and continuous performance.

Bud Graham: Ray did the Quad and Stereo mixdown from the spliced 8 track originals. Of course, Andy Kozdin, who could not be here today, he was the Album's producer, helped to keep us from *falling on our sword* along the way.

Ray Moore: . . . and of course, Hank Altman, the supervisor of the Columbia Records portable crew . . . who with his guys and Bud did the set-up; the studio on the 7th floor of Manhattan Center . . . the control room area was up on the 8th floor. Hank and Bud and those guys made sure that everything was working properly. They are the unsung herds of our group. Unfortunately Hank can't be in on this, either, because he is making arrangements for another recording project.

Milt Cherin: Ray, don't forget the fellows in disc mastering.

Bud Graham: I guess what we are trying to say is it was a team effort and many of us at Columbia get involved in the engineering production.

R-e/p: Bud, you were responsible for the mikes . . . right?

B.G.: Yes, more or less, as you can see from the diagram, we used twenty-six mikes. Fifteen were AKG c-12's. The rest were (Neumann) U-87's. The C-12's were used on the stringed instruments, the wood winds, the tympani and the celeste'. The U-87's were used for the choir, the brass and the percussion. In all cases the mikes were placed at heights varying with the instruments and the amount of ambience and/or leakage desired.

R-e/p: We are interested in the fact that you say you only use two types of microphones . . . don't you use other kinds of mikes?

B.G.: Yes! But the C-12's and the U-87's are the mainstay of our portable equipment. Mainly because they are fairly flat natural mikes. We look for the *natural* sound of the instruments in a good recording hall. We try to *create* the *sound* that way. We don't use a special mike for a specific instrument as is done in many pop or rock studio sessions. It would be impractical to use too many different mikes in the portable equipment for a classical session . . . and with our philosophy it just isn't necessary.

R-e/p: Lets discuss a couple of things that you just touched on. First using so many mikes how do you guard against the spotlight effect . . . or don't you worry about that . . . and then, a little more about the overall recording philosophy.

B.G.: Sure we worry about keeping the ensemble effect with that many mikes, twice as a matter of fact. Once at the recording session and again at the mixdown. Our recording philosophy is not to just hang two mikes and get the realism of the concert hall variety of recording. We try to get enough of individual instruments as well as the sections of the orchestra so that we have enough . . . or certainly something to work with in the final mix-down. Our realism is more to the musical score than the concert hall. So, we do purposely highlight instruments in the final mix but always within a sound or ambient framework so as to keep the highlighted instrument in a proper perspective, relative to the realism of the score . . . that's our realism.

R-e/p: You mentioned a good hall. Just how important is a good hall . . . and how do you describe a good recording hall?

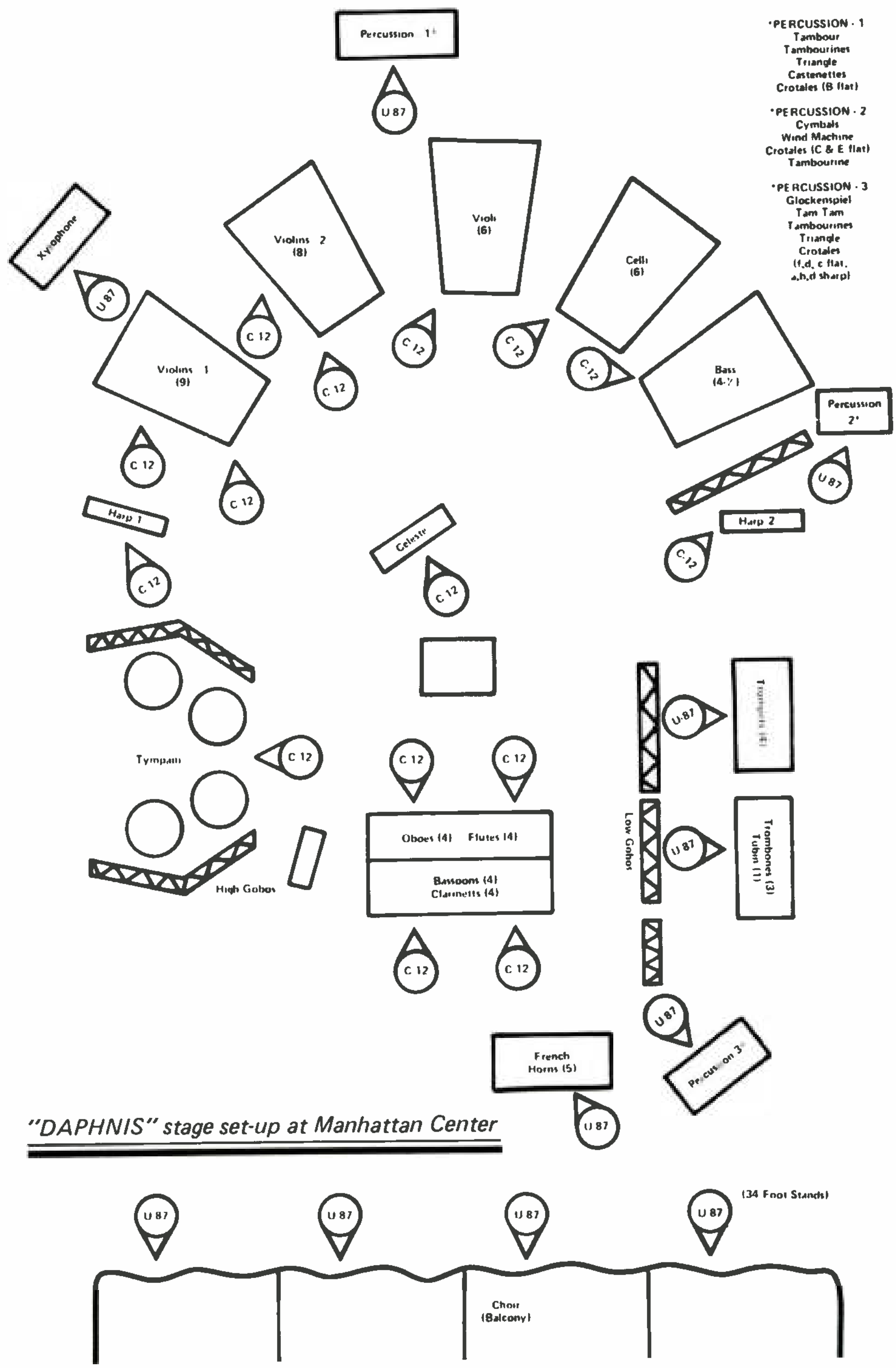
B.G.: A good hall is very important! Sure it is possible to get fair to good results in a problematical hall, but it requires a great deal of experimentation to achieve those results . . . we do it when we have to. In a good hall those results come more easily. Experimentation in these cases can be utilized on the nuances which go beyond just good sound. A good hall by our definition is a hall, a studio, a ballroom, concert-hall, or music-hall - which allows us to get close enough to the instruments and/or the sections of an orchestra without sounding as though we *are* close. Hence many of the mikes used in such a situation are subject to a fair amount of the ambience and reverberation of the hall.

R-e/p: Doesn't the use of many mikes defeat the need for a good hall by virtue of what close miking produces?

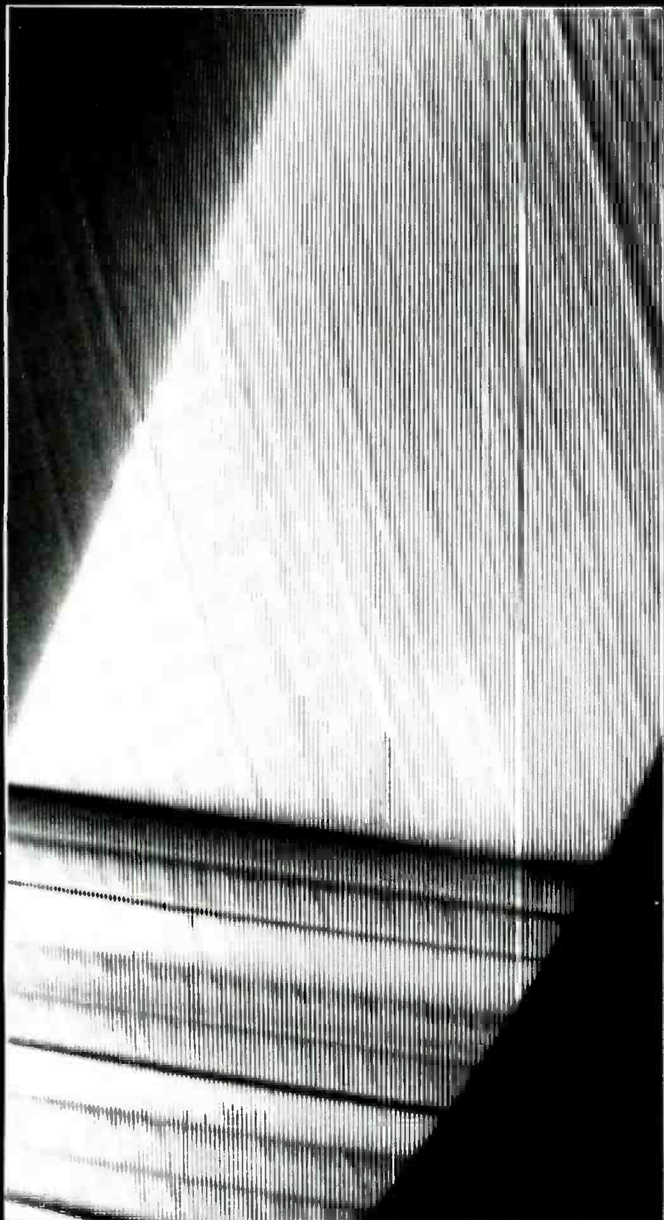
B.G.: Aren't you making an erroneous assumption about all the results of close miking. Close in a matter of degree and of definition. There is a world of difference between hanging 2 mikes 30 feet from the orchestral set-up, 15 feet high, and placing a mike a half a foot from the sounding hole of a guitar. We seldom get very close to an instrument, and if we do its because of the kind of instrument it is . . . say, a celeste' or a harp. Or, because of a problem. Then, we may usually use only a small percentage of the track in the final mix . . . only for articulation.

R-e/p: How does 26 mikes compare with the greatest number of mikes you have ever used?

B.G.: A little more than half. For example on our other Grammy nomination, Orff's CARMINA BURANA we used 50 mikes. But, of course, that piece includes a choir of 120 voices, a boys choir of 25, three vocal soloists, and an enlarged orchestra . . . additional percussion. Also, Carmina, has 24 different sets of musical forces which, during the recording sessions, required quite a bit of switching of inputs to the consoles, as we went.



"DAPHNIS" stage set-up at Manhattan Center



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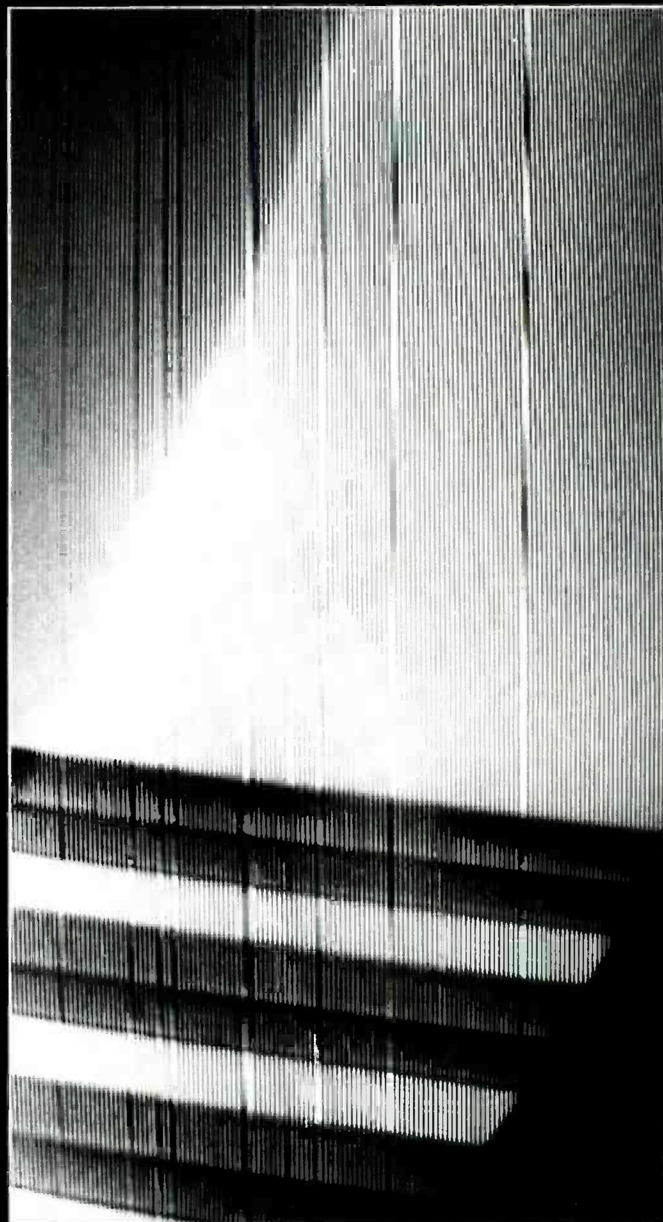

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CAPSULE BIOGRAPHIES

Milt Cherin *was born December 6, 1918 in New York City where he attended its public schools including the radio school set up to train radio officers for the Merchant Marine during World War II.*

He equalizes his life by playing guitar, domera (a Russian balalaika type instrument) making guitars and kindred instruments and by sculpting in wood.

He has been with CBS for 19 years.

Bud Graham *was born September 21, 1927 in Hackensack, New Jersey. He attended St. Peters Prep and Villanova University.*

He spends his off hours with his family, and gardening, both indoors and out.

Graham has been with CBS for 29 years.

Ray Moore *was born August 16, 1932 in Brooklyn, New York. He attended City schools and graduated from Queens College and the Polytechnic Institute of Brooklyn (now New York). He has been with CBS for 19 years and occupies his leisure hours composing music and playing piano.*

R-e/p: Yes, Carmina Burana was your other Grammy nomination in 1975 wasn't it?

R.M.: Yes! Bud, do you agree with me . . . I don't know about Milt since he was on vacation when we did the majority of work on Carmina . . . that the Carmina recording was a much greater challenge both technically and logistically.

B.G.: No doubt about it!

R.M.: Besides the 50 mikes, the musical forces of 260, Carmina was done 16 track not 8 track as was Daphnis. So even after the original session, the investment in time spent splicing and mixing Carmina was greater than Daphnis. In part this was because of the 24 different set-ups and because of the problems we encountered along the way . . . which we solved as well as those in Daphnis.

R-e/p: Speaking of splicing, Milt, what is entailed in splicing and what was entailed in splicing Daphnis?

Milt Cherin: A complete, correct and continuous performance of Daphnis was the goal of our splicing. But a complete, correct and continuous performance is our goal everytime we splice. In this case there were three sessions to draw from; one from March 1974 and two from March of 1975, done less than a week apart. One worry we had before the fact was that the session of March 1974 would not match those done in March 1975. As it turned out we were lucky and didn't have that problem, *and naturally we said we planned it that way.*

Before splicing Daphnis we had decided to go back to the 45 degree splicing angle after we had experienced some difficulties with a steeper angle - approximately 78^0 - during the splicing of Carmina Burana. It seems that Masonic Hall in Cleveland where Carmina was recorded has a reverberation that can not always be *cut* into at a steep angle. (Manhattan Center is not unlike the Masonic Hall in Cleveland as far as sound and reverberation is concerned). At times, a non-musical bump greets us at the splices. The steep angle was sort of an overkill situation since Andy (Andrew Kazdin the album's producer) had chosen to use a tape speed of 30 ips instead of 15 ips to facititate splicing. Using a tape speed of 30 ips gives us a geometric advantage for splicing which allows the tape time to be one half the real time, relative to 15 inches per second, during the duration of the splice. At 15 inches per second, one inch tape cut at a 45 degree angle exhibits a one inch long cut projected along the direction of the path of the tape in playback. The splice passes over the playback head in 1/15th of a second, real time duration. Obviously, then, the same 45 degree, 1 inch cut played back at 30 ips produces only a 1/30th of a second real time duration. The 45 degree angle prevented the problem; hence no bumps.

R-e/p: Isn't the steepness of the cutting angle more the culprit in producing splicing bumps than the ambience or reverberation at the hall?

M.C.: No, it is not the steepness of the cutting angle that is necessarily the culprit. We had used the steep angle on tapes recorded in Philharmonic Hall and never had a problem with bumps. Of course, Philharmonic Hall (Avery Fisher Hall) does not have the reverberation of a Manhattan Center or the Masonic Hall in Cleveland.

R-e/p: Was there any other problems in the splicing procedure?

M.C.: No. Once the 45 degree angle was agreed upon there were no other problems. After Andy chose the takes he wanted to use, the splicing of the complete performance took about five workdays. Splicing is very time consuming in classical work.

R-e/p: In regard to time, which of these functions, the sessions, splicing or miking takes the most time?

M.C.. That depends on the musical forces, the artist or artists, the kind of musical piece being recorded and the musical or recording philosophy of the producer. It, obviously, also depends on where the problems occur. If, however, you include the preparation, the planning, the set-up and tear down for the recording session to the length of the sessions themselves then the rule of thumb would be that the time spent in each function is approximately equal. Certainly if you include man hours it is. There are usually two or three engineers on an album project.

B.G.: On Daphnis, I believe there were three sessions totaling 10-1/2 hours. Preparing and packing the equipment for the move to Manhattan Center . . . set up and tear down . . . it took about another 12 hours.

M.C.: Splicing took about a total of 30 hours.

R.M.: Quad and Stereo mixing - 36 hours.

B.G.: Add an hour and a half for cutting the reference acetates and then another hour and a half for the masters.

R-e/p: Let's see, the total is about 92 hours! How long is the record?

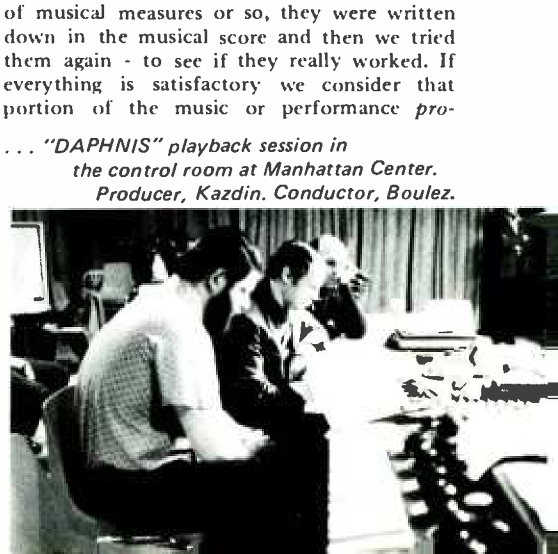
R.M.: About 55 minutes.

R-e/p: In the spirit of equal time, Ray, tell us about the mixing.

R.M.: We did the Quad-mix first! Once Andy had decided on the basic balance . . . he had chosen his track development at the original sessions and he monitored the sessions that way, so he received few if any surprises and established a basic sound frame of reference equalizing each track to achieve it. We then proceeded with what we affectionately call our *piecewise-linear-mixing-routine*. The term *piecewise-linear-mixing-routine* is a term we borrowed from our electrical engineering school days, and is the nearest thing you can get to tender loving care in engineering terms.

What it entails is deciding the changes in levels or equalization of various tracks to achieve the desired instantaneous balance or sound references as the music progresses. When decisions were made on a certain number of musical measures or so, they were written down in the musical score and then we tried them again - to see if they really worked. If everything is satisfactory we consider that portion of the music or performance *pro-*

. . . "DAPHNIS" playback session in the control room at Manhattan Center. Producer, Kazdin. Conductor, Boulez.



. . . *during "CARMINA" in the control room at Cleveland's Masonic Hall. Conductor Thomas, Ray Moore, Andy Kazdin.*

grammed. We then continue programming until we find a convenient place to stop for splicing or until we get to a complicated area where we chose a place to splice even if it isn't convenient. Since we splice frequently, the term *piecewise* certainly fits even if *linear* is obscure.

In Daphnis there were only three different musical forces; orchestra, unaccompanied chorus, and orchestra with chorus. So mixing it was fairly simple - compared to Carmina with its 24 relatively different sections and frames of reference.

R-e/p: Was there anything particularly unique about the equalization to achieve the balance on sounds?

R.M.: It's hard to say what you consider unique. Bud mentioned how we try to achieve a natural sound at the recording session. He does little or no equalization there. What is it, Bud, a little high end on the violin mike.

B.G.: Yes, a few steps at 7.5 kHz.

R.M.: We equalize when the combination of the individual tracks does not give us the balance or sound we are looking for in each of the 4 channels in Quad . . . or in each of the channels or phantom center in a stereo mix. We do it by ear. We know our monitoring system very, very well. In Quad we use four KLH6's or KLH4's. We use them in both the mix down and the recording session, in a rectangular pattern. The secret, I guess, is that we know them very very well and are able to analyze how the existing tracks are not giving us the sound or balance we want. Usually, by our standards, the equalization is seldom drastic. We get into exotic . . . lets call it exotic rather than drastic . . . when Andy hears a line he would like to bring out . . . usually a line that is important to the score but is seldom heard in recordings or in concert. Then we have to jiggle tracks, levels and equalization to achieve the effect. In that case its no holds barred . . . well almost, anyway.

In Daphnis we worked on the French Horns in that manner in several places . . . as well as a few other instruments. Also in that vein, but this time cosmetically, the trumpets at the very end of the piece were a little more than slightly prominent in their own track on the take Andy had chosen, so we got into the juggling act to remedy that situation.

After the problems of the mix were solved in the Quad mix it was relatively easy to carry over the solutions to the stereo mix. Of course our 4 track discrete Quad mix was encoded into a SQ two track tape. That process is pretty much a straight forward, rather mechanical, procedure. At this point what we have produced is a couple of two track tapes - one Quad, one Stereo ready for mastering . . . so, our job was done . . . and this time we earned a GRAMMY.

THE NEW GENERATION OF *Polyphonic Synthesizers*

BY
BOB EASTON

The famous riverboat gambler Maverick, once allowed that he could fool all of the people some of the time, and some of the people all of the time, and that those were very good odds. Apparently the marketing and use of electronic instruments fits the same space, except that everyone understands even less about it than they did about those card games. Who'd care to define exactly what a synthesizer is, and who's to say what are "good" and "bad" features for these instruments? It's clear that some instruments are truly classics, while others, though inexpensive, also aren't worth much. With little but salesmanship to "prove" which is best (by making it sell "best", of course) the field has been further enlarged by the entry of a number of *polyphonic* synthesizers.

Polyphonic synthesizers are - - as a simple definition - - those on which many notes can be played at once. Arguments abound as to which are merely organs in disguise, and which are *genuine* synthesizers - - as if the latter had been defined. Which it hasn't. Why does it matter? Well, they are all expensive, and some do jobs the others don't do. But mainly, there's a mysterious drive in every marketing manager to label something "Polyphonic Synthesizer" regardless of what it is. And that, of course, implies that all these new instruments are somehow similar, and perhaps created equal. Well, they are not, anymore than a piano and a harpsichord are - - which have different names so you'll know they're different, and *not* the same!

But since the required constituents of a single-note synthesizer have never been defined to everyone's satisfaction, let us skirt this touchy task of label-making, and see what's going on in some of the more popular, or at least most-talked-about instruments. It'll be clear that they aren't going to sound very similar, and also that most of these approaches verge on Big Bucks. It will also be clear that nobody agrees on what a synthesizer is, even yet . . .

E–μ SYSTEMS

It looks like E-Mu was first to hit the market with a genuine polyphonic synthesizer. (It's pronounced as in the first syllable of Electronic Music.) There is little question that theirs is both genuine and polyphonic, for they have the largest selection of custom synthesizer modules available from any source. And their instruments can play up to ten notes at a time - - because that's how many fingers you have, they said. E-Mu's aren't found in music stores, and are in fact assembled to the customer's order, as with other brands of modular systems. They are somewhat uncompromising in quality, and really aren't intended for the casual performer, or for that matter, the casual spender. But virtually any degree of sonic sophistication *can* be produced, depending on the skill and experience of the user, and on his budget. E-Mu's are studio instruments, and as such are great things to realize very specific results with, when doing a very controlled musical piece. Their solutions to the basic problems of polyphony are ingenious, so let's look at the problems, and then their fix for it:

Problem: Given four or more synthesizers in a polyphonic instrument, and 73 keys on the keyboard, how are the note assignments made so that non-musical events don't happen as one plays up and down the whole keyboard? The Fix: It's a multiple-solution. The keyboard contains a large amount of record-keeping logic, and knows several facts at all times. Like how many keys are already depressed, which synthesizers are already assigned to which keys, which synthesizers aren't assigned to anything, and what's more, what synthesizer was assigned to what key previously in the performance. The keyboard system then allows the user to order several conditions, such as:

1. *Always* assign the first key I press to synthesizer #1.
2. *Always* (or alternately, never) assign the next keys I press to the next available synthesizer, rather than to any arbitary one.
3. In a repetitive figure, see to it that the same keys continually go to the same synthesizers, even when they're played in a different order.
4. *Never* "steal" a voice when more keys are pressed than there are synthesizers. Instead, do nothing - - which is deemed to be better than some disruptive assault on what has been done previously.

The practical consequence of this keyboard system, is that each synthesizer can sound different, yet one still knows which sound will occur when a key is pressed. Best of all, 73 synthesizers aren't needed! The available ones are rotated among the keys pressed, on an as-needed basis.

A fringe benefit of E-Mu's digital keyboard system is digital words that represent specific notes. That amounts to an open invitation to offer a polyphonic memory system, or sequencer, that would allow the performer to load up music, as with tape, and then edit it, play it at different rates or with different timings, and completely alter the sound *after* playing

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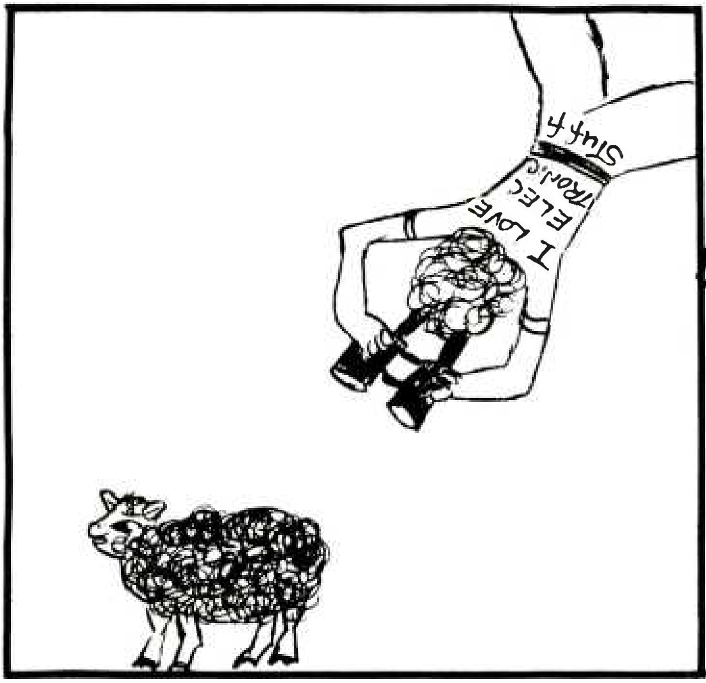
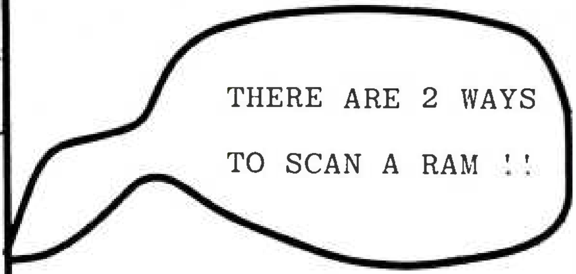




Now that I have opened the subject on rams and scans, I could go on to binary words and bits and digit my way right into automation. Ah, when I think of automation, I just naturally think of Allison Research and their new Memory Plus System, now there's a fine example of ram scanning if I ever saw one. For those of you who really want some technical information, I am ever ready to spew forth my vast knowledge on the subject of "MEMORY PLUS". So here I go!

1. It works real good.
2. It has alot of parts and stuff in it.

Now, if that isn't enough information to satisfy your technical appetite, we will send specs or you can call or write and Paul Buff would be happy to chat for days about the system and its capabilities. Well, I really must go now, it is time for me to shine the company soldering irons. (I only wish they would let me turn them of first) Take care and do find out more about "MEMORY PLUS", it really is a friendly machine.

Love,

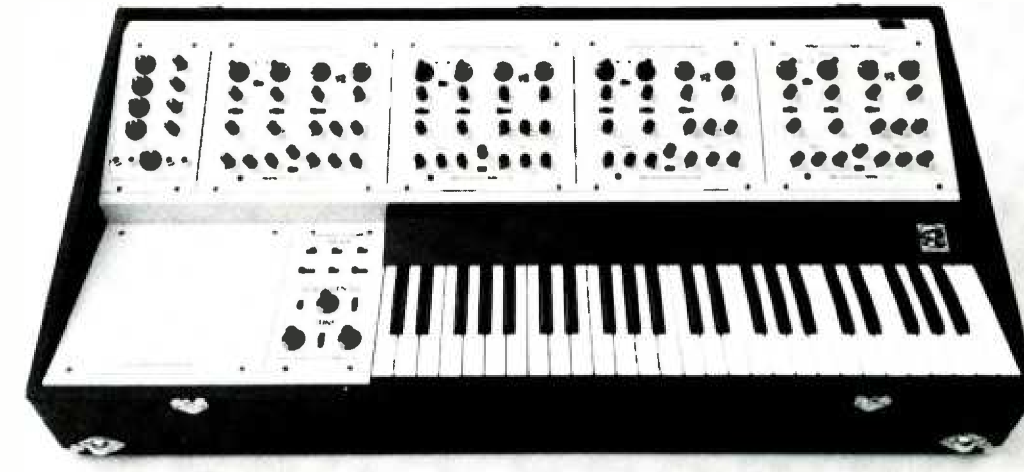
Allison

ALLISON RESEARCH, INC.
2817 ERICA PL. PO Box 40288
NASHVILLE, TENN. 37204
Dial (615) "ALLISON"
Or (615) 385-1760

Circle No. 139

it. It's very difficult, and often impossible, even for fine musicians to play multiple parts live, with the sort of exacting precision they would like to hear. The Polyphonic Digital Sequencer is a good solution, serving as something between a tape machine and an ideal editing system, and operating in good musical style. We recently heard the entire first movement of The Planets by Holst, performed by Pat Gleeson of Different Fur Music, where a polyphonic sequencer was used to assemble the majority of the piece. Super. Pat says that he really prefers the dead-accurate pitch and timing of the sequencer output, to the normal errors made by the best of musicians.

The E-Mu makes a lot of sense as a studio instrument, particularly for people who wish to end up with sounds that accurately resemble the idea they started out after. It's the classical synthesizer structure, and it's for people who know what they're doing, or intend to learn. Hear some E-Mu on Leon Russell's Will 'O The Wisp and his new Wedding Album. Frank Zappa's forthcoming album ought to see some polyphonic E-Mu, too.


. . . the four-voice polyphonic synthesizer from Oberheim Electronics, Inc.

THE OBERHEIM "FOUR-VOICE"

Tom Oberheim's "Four-Voice" synthesizer is the big seller at the moment, being affordable, *and* containing four complete synthesizers. It's built on a modular concept, but a very different one than used by E-Mu Systems, or Moog before them: Each module is an *entire* synthesizer, rather than a single function, such as an oscillator. It's available as a small two-voice synthesizer, but it can also be extended to eight voices by adding "synthesizer expander modules" to the basic system. Seems that most of them start out as four-voice machines, hence the name.

Oberheim has adopted a purists' point of view about what's needed in a polyphonic synthesizer. He feels that a lot of professional musicians want a "true" synthesizer, and specifically do not want the sound associated with "organ based" instruments.

Each individual synthesizer does have to be tuned, and then voiced to the sound that's wanted; this takes time, but pays dividends in producing exact rather than approximate musical results. To simplify matters, the Four Voice has a certain amount of "parallel control" of synthesizer parameters - - tunings, filter settings, and other adjustments can be made by master controls affecting all "voices" at the same time. Also, every single audio and control signal within each synthesizer appears on miniature connectors within the system. It's possible to interconnect these, to provide all sorts of external control of parameters affecting a sound.

Here's what is available within each synthesizer module on the Oberheim. This is repeated on each of the polyphonic voices:
- Two VCO's, sawtooth and pulse waveforms.
- Low Frequency oscillator.
- Filter: low pass, high pass, bandpass, notch.
- Three input mixer to filter.
- Two envelope generators.
- Voltage controlled amplifier.
- A lot of control attenuators.

The basic elements can be interconnected in many different ways. The envelope generators and the low frequency oscillator are used to modulate pitch, pulse width, filter cutoff frequency, and more. Each of these modules is a "classical" synthesizer - - meaning that it is structured similarly to most of the famous single-line originals. Hear it on the new Three Dog Night album American Pastime and on Stevie Wonder's next-to-be-released. Also, check Weather Report's Black Market and Herbie Hancock's Manchild.

THE ARP APPROACH

So far, all synthesizers have had voltage controlled oscillators, which in musical terms, means that notes can be produced at *any* pitch, and made to track together as a cluster, or chord. Also, a single voltage controlled low-pass filter has been used to perform timbre-shaping of the raw sound. But it's been pointed out by quite a few people that although having lots of oscillators is fine, putting them all through a single filter isn't the brightest thing to do. There are some instruments forthcoming that resolve the latter problem, and some others that dispense with the whole voltage-controlled oscillator idea, substituting a fixed-interval divider-chain concept.

ARP Instruments and others have introduced "String Ensemble" synthesizers over the past two years, as an economical approach to a job that can be impossibly expensive, whether done by multi-tracking of single-line synthesizers, or by your local Musician's Union. Most string synthesizers contain a high-frequency oscillator, which drives a rate-multiplier to produce the twelve-tone scale. Each of these "top octave" notes in turn drives a

Circle No. 140

divider chain, to produce intervals for all lower octaves. It can't get out of tune, and vibrato can still be produced, as with real VCO's, by frequency modulating the master oscillator. Why does it sound like strings? It's "specific synthesis" - - circuitry optimized to produce a wave-shape reminiscent of violins, violas, and maybe even a cello. The instrument always has some "extras" to help the illusion: vibrato, an adjustable crescendo, and delays mixed back into the sound.

ARP's string synthesizer may have been the first instrument to employ bucket-brigades - - which relates to why they're called "ensembles". Usually three of these analog delay lines, each having 185 stages and operating with frequency modulated clocks, are used to provide delay, somewhat complex pitch errors, and when summed together - - a pretty nice "ensemble" voice for background tracks.

But the filtering in the "String Ensemble" is fixed, while in classical synthesizers, a voltage controlled filtering scheme shapes each note's timbre. It occured to the people at ARP Instruments that their machine might profit from being connected to the filter of their other synthesizers, with key triggers causing each note to be individually shaped. That's what's being done, and it is now available as ARP's entry into the polyphonic synthesizer field. It has the advantage of being inexpensive if you already have one of the two instruments, and it very definitely produces a whole new group of sounds from the basic String Ensemble.

The POLYMOOG starts out this way too, but introduces some ingenious new twists to the *not-voltage-controlled* oscillator game. And it's one of the largest product development efforts yet seen on a new instrument:

THE POLYMOOG

Moog Music's forthcoming entry to the polyphonic club strikes a compromise between positions taken by other instruments so far. It's called the PolyMoog, and is a 72 key fully polyphonic instrument, having both preset and variable settings. And to a degree, the preset sounds can be altered by instructing the instrument to give the player manual control of certain preset characteristics. If you wished to embellish the sort of low pass filtering decided on by the Moog people, you could manually over-ride that aspect of the sound, leaving the rest of it stock. Or large portions of a preset can be altered, to the extent of using presets only as sonic points of departure.

What is very different about the Polymoog, is an approach to instrument architecture that takes advantage of this being 1976, with current economies in fabrication of custom intergrated circuits. It uses 72 custom I.C.s, one per key, to solve a problem that has kept polyphonic instruments of many sorts off the market so far: Each key now has its own VCA (voltage controlled amplifier) and VCF (voltage controlled filter). Any number of keys can be played at once, and all can have identical characteristics, commonly adjustable by a single set of controls.

The Polymoog starts off somewhat like an organ, to the extent that it uses two master oscillators and two parallel divider chains to produce the intervals. But each note can consist of two "ranks" - - a sawtooth and a pulse wave. (They can be detuned, too.) This mixture is filtered, and has a loudness contour (envelope) placed on it by envelope generators, one a one-per-key basis. This "minimum" amount of synthesizer activity is brought to an output appropriately called "direct". For those interested in something more cultured, a set of resonant filters having adjustable frequency and emphasis (Q) can be placed in the signal path, providing another final output. Also, a standard "Moog type" voltage controlled low-pass filter (24 dB/octave, low distortion) is available, and can be controlled by a footpedal as well as preset or variable envelopes. Another output for that, as well as different other mixtures of effects!

The Polymoog has eight preset voices, and a ninth you set up yourself. There are violins, piano, clavinet, harpsichord,vibes, brass, more strings at octaves, and a funky synthesizer sound. More importantly, it's one of the few electronic instruments around so far with a touch sensitive keyboard - - it senses key velocity, and produces a loudness response that resembles the behavior of a piano quite well.

We spoke to Mike Boddicker, a heavily called keyboard man in Los Angeles, who also acts as a product evaluator for Moog Music. He's been using the Polymoog on Stevie Wonder's new album, (to drop a name,) and gets a lot of requests to trot out the new Moog. That's probably a good sign, when he has the only one around and it's not in the stores yet. But Mike says they plan to start delivering this July, and it'll be in the middle-four thousand dollar range.

GUITAR SYNTHESIZERS: 360 SYSTEMS

While most manufacturers have been busy making keyboard-controlled synthesizers, 360 Systems, a small Los Angeles based company, has been producing devices that allow traditional "live performance" instruments to be used as synthesizer-controllers. An interface called a "Pitch Follower" is connected between a "real" instrument and the synthesizer; it translates the original signal into pitch, envelope, and trigger signals compatible with the synthesizer. Though originally manufactured for use with single-line instruments, it has evolved into a six-channel Polyphonic Controller, for use with electric guitar.

The electric guitar is an ideal synthesizer controller, since it is polyphonic, common and easy to play, and in terms of pitch, it allows the musician to "play in the cracks" between notes. Best of all, a guitar permits the performer to control note dynamics. It's also easy to extract separate signals from each string, for running a synthesizer. The result doesn't sound much like keyboard synthesizers either, partly since people approach music differently on guitar, and also because the guitar is expressive in different ways.

The workings of a guitar synthesizer can be understood by viewing one channel of it as a slice of the entire system. Three separate signals need to be created per string, in order to accurately represent what a guitar player is doing. These are pitch and envelope control voltages, and a trigger, which represents when a

. . . Michael Boddicker and the Polymoog Synthesizer from Moog Music, Inc.

. . . 360 System's polyphonic guitar synthesizer

string is picked. Pitch-to-voltage conversion is accomplished by isolating the fundamental of a note, and measuring the period of the waveform. The log of a voltage representing *period* will be increasing with *frequency*, in this case at a 1-volt/octave rate. This signal controls the pitch of all oscillators.

String dynamics are represented by an envelope follower circuit. It's an average-rectifier followed by a low-pass filter. Being non-linear, it produces a result that is better suited to operating a synthesizer's VCA than a real guitar envelope is. In fact, a guitar envelope isn't very good at all for shaping the amplitude of most synthetic sounds!

The last circuit produces triggers to operate the synthesizer's envelope generator. It responds to flat picking, so that when a pick leaves a string, a new trigger results. This way, a filter sweep can be made to shape each new note played. Legato performance doesn't retrigger the filter, so one's right-hand technique on guitar determines what the synthesizer will sound like.

360 System's guitar synthesizer uses six "real" synthesizers. Each of these contains two oscillators, a low-pass filter, envelope generators, a mixer and voltage-controlled amplifier. It can produce up to eighteen notes on a full chord, and can play any transposition of a performance. In fact, any string can be individually transposed *while* playing, to produce chord clusters that couldn't be played on a normal guitar. It's a fair amount of effort to set up voicings on six synthesizers, but then again, it sounds like six instruments playing at once. There is also a small advantage to playing a guitar synthesizer, that isn't true of its keyboard counterpart: You know what sound is on which string, while on keyboard, you only know this under some circumstances.

The guitar synthesizer works out best in the hands of people who can orchestrate. It's possible to have say, brass instruments on the top two strings, a violin and viola on the next two, and some bass figure on the A and E strings. Guitar players seem to have a limitless appetite for good new sounds, and this synthesizer may keep them occupied for a number of years. It gets the musician into playing new kinds of material too, much of which would never be played on guitar - - unless the "right" sound was available.

FOR MORE INFORMATION

Write to the manufacturers mentioned here for all the facts. Some of them will have instruments not covered in this article, to be released soon.

OBERHEIM ELECTRONICS, INC., 1549 9th Street, Santa Monica, Ca. 90401. 360 SYSTEMS, 2825 Hyans St., Los Angeles, Ca. 90026. E-Mu SYSTEMS, 3046 Scott Blvd., San Jose, Ca. 95050. MOOG MUSIC, INC., Academy Street, Williamsville, NY 14221. ARP INSTRUMENTS, 320 Needham Street, Newton, Ma. 02164.

New Products

dbx ANNOUNCES PLUG-IN REPLACEMENT FOR DOLBY

dbx announces introduction of its K9-22 noise reduction system card, a direct plug-in replacement for the CAT-22 noise reduction card which forms the basis of the Dolby "A" system.

The dbx card option allows any 361, M16 or M24 Dolby "A" equipped studio to convert to dbx noise reduction and be in operation in minutes with no wiring changes or other procedures required. The dbx K9-22 cards are supplied in a convenient Halliburton instrument case for ease of carrying from one location to another. The case also stores the unused Dolby CAT-22 cards while they are not in use.

The dbx K9-22 cards may be purchased in any quantity at a cost of $250 per card, permitting additional channels to be purchased as and where needed. The cost of converting a Dolby M16 to dbx is approximately $4,200, or less than half the cost of a complete 16 channel dbx or Dolby noise reduction system.

"We hope by this means to make dbx noise reduction abailable to any studio or performing group that needs it, even if they are unable to purchase the free-standing dbx system," says Larry Blakely, Director of Marketing for dbx. "With the K9-22 card option available, a producer or engineer who prefers dbx noise reduction can in essence take his own dbx with him wherever he goes, provided there's a Dolby "A" rack to plug it into when he gets there. The card option will also be of interest to mastering houses and record labels who need the dbx desperately when they need it, but who have difficulty justifying the duplication in capital equipment," Blakely said.

dbx, INCORPORATED, 296 NEWTON STREET, WALTHAM, MA. 02154

Circle No. 146

AMPEX INTRODUCES ATR–100 RECORDER

The ATR-100 features an exclusive tape handling system first developed by Ampex for computer tape transports that totally eliminates the use of pinch rollers, and advanced signal electronics with performance specs that are as much as 10 dB or an order of magnitude better than any audio recorder on the market. "The ATR-100 is a milestone in practical sound recording that spans nearly 30 years of pioneering leadership in audio engineering and incorporates modern computer technology."

A totally new transport design includes a fully servoed tape drive system permitting substantial improvements in tape handling and timing accuracy.

The ATR-100 has a signal-to-noise ratio of better than 80 dB (full track at 30 ips) and an overall record and reproduce response of ±3/4 dB, 100 Hz -15 kHz at 15 ips (almost visually undetectable on an oscilloscope), compared to a typical performance spec of ±2 dB quoted by other recorder manufacturers.

Sound Alternative: 2200

In the pursuit of good sound, a new alternative has arrived-the SAE 2200. Our design goal; to offer the sound quality and reliability of our 2500 in a more modest power and price package. The realization of this goal is an amplifier whose specifications become the standard by which others are judged:

—100 Watts RMS/Channel (both channels driven) from 20Hz to 20kHz into 8 Ohms.

—0.05% Total Harmonic Distortion from 250mW to rated power, from 20Hz to 20kHz.

IM Distortion at 8 Ohms	0.05% Max.
Input Sensitivity	1.5 Volts, RMS
Frequency Response at rated power	+0.25dB, 20Hz to 20kHz
Noise	Greater than -95dB below rated power
Transient Response of any Square Wave	2.5usec, rise and fall time
Slew Rate	40 Volts/usec.

(Specifications comply with FTC requirements for power amps.)

With these credentials the possible uses of the 2200 become almost limitless. There's tri-amping, or driving efficient speakers, or studio monitor systems, or small listening rooms, or large listening rooms with small budgets or ...

Besides performance, the 2200 offers SAE quality contruction, and features. For example:

LED POWER DISPLAY-Utilizing 15 Light Emitting Diodes for each channel, the 2200's power output is instantly displayed in watts from 20mW to full power.

FULLY COMPLIMENTARY CIRCUITRY-This assures low distortion, plus offering high slew rate, wide bandwidth, and accurate reproduction of any waveform.

STEADY-STATE USE-Everything from the monocoque construction to the plug-in board design and massive heat-sinks are SAE designed for continuous performance under the most demanding conditions.

SIZE PLUS-Measuring only 8 in. deep and 5.25 in. tall this 19 in. rack mount beauty is truly an efficient power package for any place or application. PLUS the 2200 offers relay speaker protection and a FREE 5 YEAR Service Contract.

PRICE: $450.00 (suggested list)

SCIENTIFIC AUDIO ELECTRONICS, Inc.
P.C. Box 60271 Terminal Annex,
Los Angeles, California 90060
Please send more information on the 2200 power amplifier

NAME ______________________

ADDRESS ______________________

CITY ____________ STATE ______ ZIP ______

SAE

Scientific Audio Electronics, Inc.
P.O. Box 60271 Terminal Annex, L.A., CA. 90060

Circle No. 147

circuitry and occupies a counter space only 42 1/2" long by 21" high by 17" deep.

GARNER INDUSTRIES, 4200 NORTH 48th STREET, LINCOLN, NB. 68504 (402) 464-5911.

Circle No. 158

CAPITOL MAGNETICS ADDS FORMULA Q19, COMPATIBLE MASTERING TAPE FOR HEAVY DUTY USE

Designed for heavy duty studio use, Formula Q19 will give excellent reproduction with good headroom for original recording, dubbing and other recording uses. Optimally balanced for low bias noise and minimal print through, the tape also has a new backcoating formulation guaranteeing high scratch and wear resistance, smooth pack with high speed winding and low abrasion of capstans and guides.

Available in 1/4 in., 1/2 in., 1 in. and 2 in. widths and all popular lengths, Formula Q19 will be marketed under the company's Audiotape by Capitol label. Width selections include either 1 or 1 1/2 mil thickness.

Formula Q19 is fully compatible. No change of record setting will be necessary for most studio machines.

CAPITOL MAGNETIC PRODUCTS'1750 N. VINE, LOS ANGELES, CA. 90028, (213) 462-6252.

Circle No. 159

LEXICON ANNOUNCES MODEL 102-S STEREO DELAY UNIT FOR RECORDING STUDIOS

In addition to providing the same flexibility and unequaled audio performance established by Delta-T 102, the new STEREO 102-S system incorporates two independent delay units in one chassis. The unit also features a new VCO module which for the first time gives recording engineers comprehensive signal processing capabilities in the time domain. New effects possible are more natural double tracking, flanging, vibrato, time delay panning, extreme pitch modulation, and signal transformation for special effects.

The two delay lines can be used independently with totally different signals (for example, when the need for simultaneous delayed echo and doubling arises). The VCO may be set to control only one channel while the other remains clocked by the crystal. System specifications include 90 dB dynamic range, 0.2% total noise and distortion, 192 ms delay per channel with 3 ms taps, dual five-position LED headroom indicators and computer quality components and construction for high reliability.

The new stereo system is totally modular and allows the user to start with a more economically configured mono system which is field expandable at a later time through plug-in modules. For studios that would normally use two delays per installation, this feature allows a savings in excess of $1,000 when adding a second delay.

LEXICON, INC., 60 TURNER STREET, WALTHAM, MA. 02154, (617) 891-6790

Circle No. 160

9x3 "MIC–SPLITTER" INTRODUCED BY SESCOM

"Mic-Splitter," is designed to split up to nine microphones, three different ways, and is now available from Sescom, Inc.

Principal design feature is that the unit has three outputs for every input. As a result, it can handle multiple feed requirements such as during a live performance when three simultaneous outputs are necessary.

An example of this application would be when one output is used for the PA system, a second for monitor or foldback sound, and a third to TV or recording.

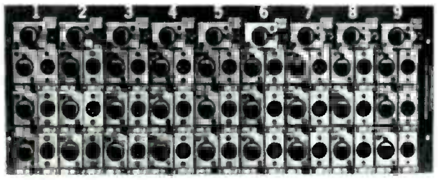
Other applications include remixing in

recording studios and for TV overdubbing and remixing.

The 9x3 was originally developed on a "customized basis" for major recording studios and TV networks for "microphone split bridging". Now is the first time, according to Franklin J. Miller, Sescom president, that the unit has been manufactured and priced for general use.

Design features include male and female "XLR" type connectors, Sescom transformers with isolation resistors, phase reversal and ground lifter switches.

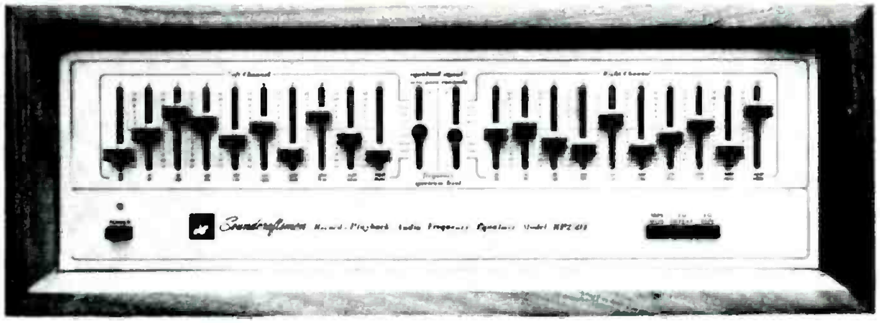
Important performance specifications are its rated primary and secondary impedance of 150-250 ohms, but it is actually designed to bridge a microphone at over 1200 ohms impedance. The input level is -10 dbm @ 30 Hz with less than .2% THD.



Priced at $1045.00 the 9x3 measures 7" x 19" for standard rack mounting, weighs 8.5 lbs. and comes in a blue leatherette carrying case.

SESCOM, BOX 590, GARDENA, CA. 90247, (213) 770-3510.

Circle No. 161



SOUNDCRAFTSMEN RP 2204 TAPE–PLAYBACK EQUALIZER

Soundcraftsmen's new Model RP2204 offers outstanding improvement in stereo reproduction through Octave-by-Octave graphic equalization. The RP2204 can be used for equalization of tape recordings and an ENVIROMENTAL TEST RECORD is included for listening environment equalization.

The equalizer is designed for connection to the tape monitor circuit of any stereo receiver or preamplifier. It includes its own tape monitor inputs and outputs with front panel pushbutton selection, so that the tape monitor feature is still available to the user through the equalizer's selector switches. Front panel pushbutton selection also provides tape record equalization switching and automatic "equalizer-on" for environmental equalization.

Other features include two completely separate ten-octave equalization panels, with plus or minus 12 dB boost and cut provided individually for each octave. Separate equalized-signal zero-gain controls are provided for each channel, enabling exact balancing of input to output levels within an 18 dB range.

Specifications include: S/N 96 dB, THD-less than .1% at 2v; typ .05% at 1v, Toroidal and Ferrite core inductors, and, an eight section ferrous chassis.

Price: $329.50.

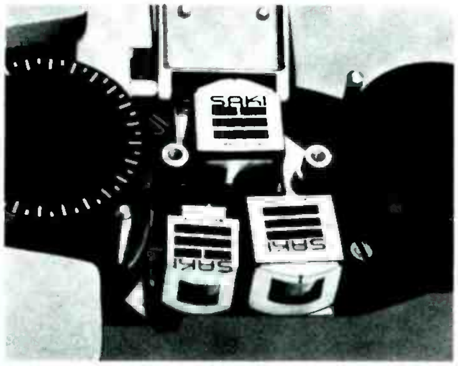
SOUNDCRAFTSMEN, 1721 NEWPORT CIRCLE, SANTA ANA, CA. 92705, (714) 556-6191.

Circle No. 162

Circle No. 163

NEW FERRITE REPLACEMENT HEAD FOR MINCOM RECORDERS DEVELOPED BY SAKI MAGNETICS

Saki Magnetics, introduces the first hot-pressed, glass-bonded ferrite head for Mincom studio recorders, available in all track formats. The new Saki head is manufactured in hot-pressed ferrite with glass-bonded gaps.



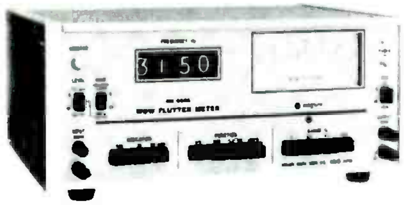
The head is plug-to-plug compatible with the original metal head used in the Mincom recorder. A reasonable life expectancy for this Saki ferrite head is 10 times that of the original metal head.
SAKI MAGNETICS INC., 1649 12th ST., SANTA MONICA, CA 90404, (213) 452-8611.

Circle No. 164

SABOR INTRODUCES NEW WOW AND FLUTTER METER

A new Wow and Flutter Meter that includes a built-in digital frequency meter has been announced by the Sabor Corporation.

The Meguro Model MK-668C features selectable calibration to permit reading peak, average or effective values of wow and flutter to conform to ANSI/CCIR/DIN/IEC, NAB or JIS respectively. Separate readings may be made for unweigh-



ted values.

Tape speed error is indicated on the digital frequency meter. This contrasts with a conventional instrument which reads in terms of percent on an analog meter.

Accuracy of the frequency counter is ±1 Hz ± the stability of the internal crystal controlled oscillator. The 4-digit frequency counter is separately useable in the 10 Hz to 9999 Hz range.

Two input voltage ranges are available: 0.5mV to 30 mVrms and 5 mV to 30 Vrms. Wow flutter range is 0.003% to 10% at inputs above 30 mVrms and 0.01% to 10% with inputs from 0.5mV to 30 mVrms. Measurements can be made at 3.15 kHz for DIN, IEC or ANSI and at 3.0 kHz for JIS, NAB or CCIR.

Price is $975 and the MK-668C is available from stock.
SABOR CORPORATION, 12597 CRENSHAW BLVD., HAWTHORNE, CA90250 (213) 644-8689

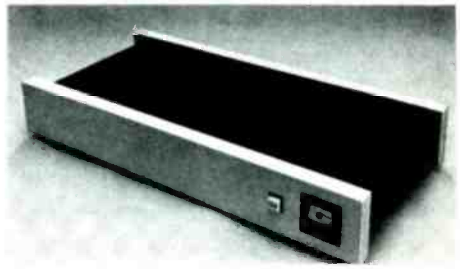
Circle No. 165

NEW GARNER CONTINUOUS BELT TAPE ERASER HANDLES TAPES UP TO 10 1/2" WIDE

Garner Industries has expanded its line of high speed, continuous belt erasers to include a new model that erases audio/video tapes up to 10 1/2" wide. This new eraser, the Model 105, along with Garner's Model 70 for 7 1/2" tapes, is described in a brochure now available from Garner.

The brochure describes how both Garner models can easily erase most tapes in one 4-second "hands-off" operation. Tapes are placed on a continuous belt and passed over high flux coils to produce an erasure which meets or exceeds professional recording standards.

Along with test reports from radio and television studios, sound studios, and education institutions, the Garner bro-



chure also contains specifications on both the Model 105 and the Model 70.
GARNER INDUSTRIES, 4200 NORTH 48th STREET, LINCOLN, NB. 68504 (402) 464-5911.

Circle No. 167

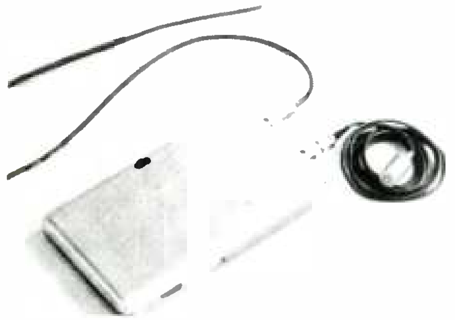
950 MHz DIVERSITY WIRELESS MICROPHONE SYSTEM FROM THOMSON-CSF LABS

The design techniques of this system assure that the amplified sound will be true high fidelity without any signs of fading, interference, or channel crosstalk. Fade-free reception is obtained by the automatic diversity switching of two antenna systems for optimum signal selection. Interference-free reception is provided by operation in the relatively wide and unused 947-952 MHz frequency band. Channel crosstalk is virtually eliminated by a combination of

Circle No. 166

high transmitter frequency stability and receiver selectivity.

50 milliwatts of transmitter output power permit the ultra-sensitive receiver to pick up a usable signal within a minimum 500 foot range. Highly accurate transmission and reception allow the simultaneous use of up to fifteen channels. The wide dynamic range of the transmitter combines with its unique soft level limiting circuit to retain the fidelity of any loud or soft sounds.



A flat audio response from 50 Hz to 15 kHz and a signal-to-noise ratio greater than 60 dB are the indices of the ability of this wireless microphone system to provide true high fidelity.

Two versions of this system are being made available. The RM-100 includes a five-channel receiver with five transmitters. The RM-102 is a single channel version suitable for applications requiring either only one channel or separate receiver locations.

THOMSON-CSF LABORATORIES,INC., 37 BROWNHOUSE RD., STAMFORD, CT. 06902, (203) 327-7700.

Circle No. 168

SESCOM'S "SPLIT MATCHER" IMPROVES SOUND OF ELECTRIC GUITAR THROUGH PA SYSTEMS OR RECORDING CONSOLES

Featuring a balanced low impedance output with a ground lift switch that plugs into any PA system, mixer or recording console, the Model SM-1A eliminates the practice of placing a microphone in front of a speaker system. As a result, a mixer gets more of the original sound from the pickup without additional distortion.

Important design features include placing an amp and pickup switch and a filter switch for bass guitar, along the side of the unit. Thus by merely flipping a switch

into the operating position, the output of an amplifier can be plugged into the box; or the high frequency signal in the filter position can be cut.

Important specifications include: primary impedance at 60K ohms; secondary impedance at 150-250 ohms; and the input level at -10 dbm @ 30 Hz .2% THD.

Priced at $36.30.

SESCOM INC., P.O. BOX 590, GARDENA, CA. 90247, (213) 770-3510.

Circle No. 170

SOUND WORKSHOP COMPACT EIGHT TRACK RECORDING CONSOLE

Known as the Model 1280 the newly introduced compact recording console has 12 inputs, 8 outputs, and 8x2 stereo control room monitor mixer as well as an 8x1 musicians cue mixer. An independant 2 track mixdown buss is provided as well. Each of the 12 inputs features 3 band equalization, 35dB trim control, push equalization, 35dB trim control, push-button track assign matrix, full panning, echo send, mute, solo, 26dB mic pad, and straight line slide faders, as well as pre and post fader patch-points.

All ten output busses feature Sound Workshop's unique tri-lite LED readouts which allow level monitoring of all outputs at a glance, even when recording 8 tracks simultaneously. Superior transient response, and low distortion is assured by transformerless low Z mic pre-amps. Balremote operators...

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anced inputs for phantom power are available as an option, using gapless lamination transformers. Dual echo returns permit full panning and EQ on the chamber return.

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BOOK REVIEW

"MODERN ELECTRONICS MATH"

by

Jerrold and Martin Clifford

686 pages — *424 illustrations*
$12.95 hardbound. . . . *$9.95 paperback*

Written as a shortcut course for techniques and hobbyists, and as a refresher "boneup" for engineers, the 686 page volume. "Modern Electronics Math"

continued overleaf . . .

assumes only that one knows how to read and can understand the numbers 0 through 9. It's designed to prevent a lack of math savvy from holding one back in electronics. The authors lead the reader step by step from simple arithmetic functions to progressively more sophisticated problem solving.

The text is extremely easy to read and understand. Each new item is clearly explained so that it relates to preceding items - as well as to items to come. The learning experience is reinforced throughout by pertinent sample problems - electronics problems with everyday applications. Each paragraph is a rigorous self-teaching tool, a detailed source of mathematical know-how, so that the reader learns without a tutor at the pace he selects for himself.

In addition to sample problems, each Chapter ends with a series of review questions - questions that help the user think about the easiest way to solve difficult problems. This book puts within the reader's grasp everything about electronics math he thought was out of reach. Math comes alive as concept after concept is applied to actual electronics problems, following the authors' premise that electronics and math are so closely allied as to be practically indivisible, and that a knowledge of math is more than just a tool for the solution of problems in electronics . . . that a true understanding of the nature of electronics can be obtained only through mathematics.

"Modern Electronics Math" leads one gradually through an understanding of basic arithmetic, the theory of numbers, AC and DC circuits, Ohm's law, algebra, logarithms, and binary numbers. The reader is systematically prepared for even more advanced work: Boolean algebra, for example, and the construction and reading of graphs, scales, and nomograms, etc. Finally, two Chapters are devoted to simplified descriptions of differential and integral calculus. Math tables and formulas are collected in the last two Chapters for easy reference. Thus, within one handy volume is a complete electronics math course geared to today's needs.

CONTENTS: Basic Arithmetic for Electronics – The Decimal Number System – Math for DC and AC Circuits – Algebra for Electronics – Trigonometry for Electronics – Logarithms for Electronics – How to Work with Binary Numbers – Octal and Hexadecimal Numbers – Miscellaneous Number Systems – Boolean Algebra – Graphs, Scales, and Nomograms – Differential Calculus for Electronics – Integral Calculus – Math Tables – Formulas for Electronics – Answers to Problem Review Questions – Index.

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"MODERN ELECTRONICS MATH" is available from R-e/p Books. Send Check or money order. $12.95 hardbound, $7.95 paperback to R-e/p Books, Box 2449, Hollywood, CA 90028

CORRECTION

The article "Stereo Miking Techniques Using Computer Pattern Analysis", by Steven B. Fuller, appearing in the April, 1976 issue of R-e/p contained several errors. Radical signs over the numeral "2" appearing in the second paragraph of the article on page 20 were omitted. The microphone patterns shown at the center of figures 6, 6A, 8, and 9 were shown incorrectly relative to their respective response patterns. In each of these cases, the silhouetted microphones should be rotated 45 degrees counter-clockwise for proper relation with the pickup patterns. Figure 12 was incorrectly labeled and shown as cardiods at 90 degrees rather than cardiods at 120 degrees, as was apparent from the text. R-e/p apologizes to Mr. Fuller and to all readers who may have been mislead by these errors.

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