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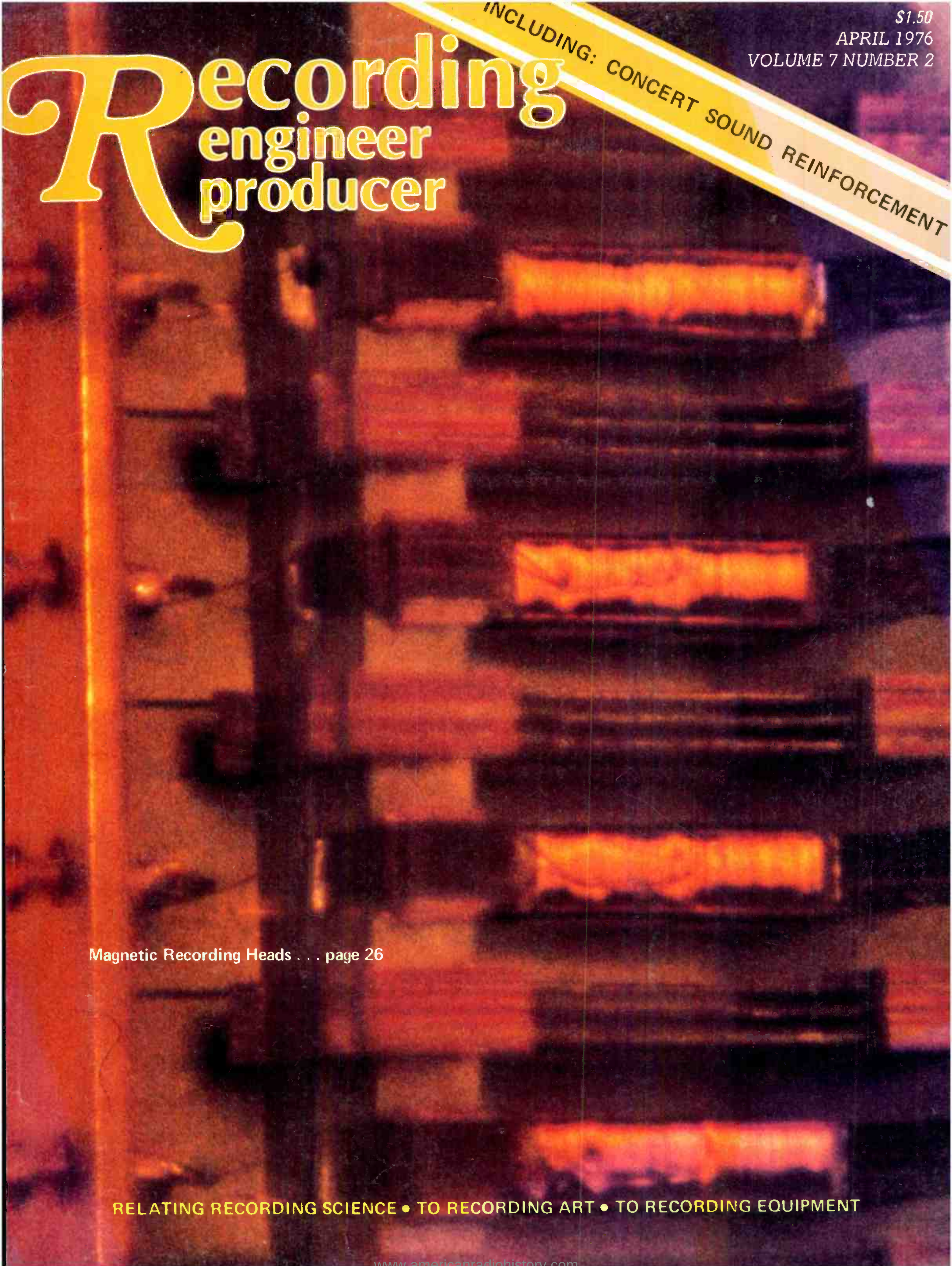
APRIL 1976
VOLUME 7 NUMBER 2

Recording engineer producer

INCLUDING: CONCERT SOUND REINFORCEMENT



Magnetic Recording Heads . . . page 26

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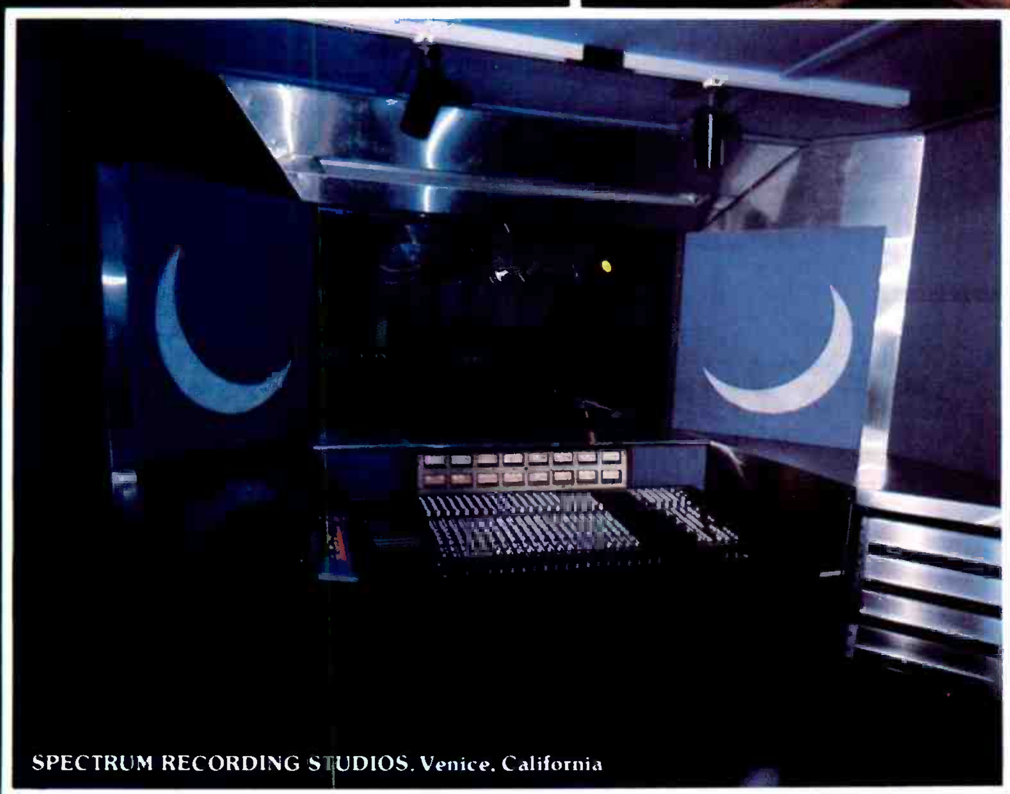

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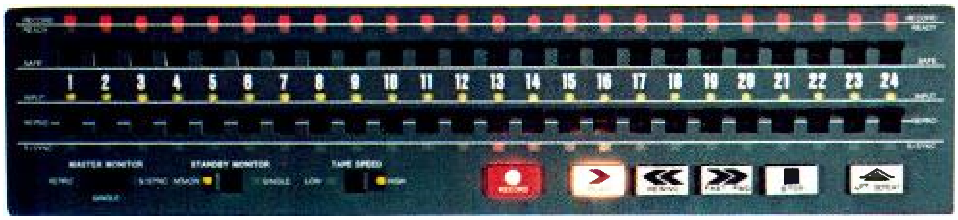
But one fact stands above all the rest. The MM-1200 is a matchless recording machine. It captures what you feed it, and it plays it back the way you put it in. You can take the MM-1200 for granted.

A recorder that grows.

If your budget limits you to 8-track work at first, you can always plug in more channels and change heads

Multichannel Endpoint



later. The MM-1200 head assembly is attached with a single screw. The basic chassis is identical for 8, 16, and 24-track configurations.

New controls, new convenience.

A newly designed control panel makes the MM-1200 easier to use than any other multichannel recorder/reproducer. It lifts out for remote use without loss of functional control. There's also an optional

remote for transport functions only. On the machine, LED indicators give a bright display of every function called up for every channel. The electronic tape timer also uses LEDs for a digital readout.

Search-to-Cue capability is standard equipment on the MM-1200. Put a cue anywhere on your tape and you'll be able to return to that same spot, from either direction, at the touch of a button. At 15 in/s, cuing accuracy is within plus or minus a half second. Sel-Sync monitoring of every channel on the MM-1200 equals normal reproduce excellence. You'll find "ping-pong" work as easy as any other technique.

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Technical specifications for the MM-1200 would fill a book. So we've written one, and you can get a free copy. Call your nearest Ampex sales office, or write to us in Redwood City. Even if you don't think you're ready for the MM-1200 right now, the facts will help convince you that the time is getting closer.

AMPEX

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Audio-Video Systems Division
401 Broadway, Redwood City,
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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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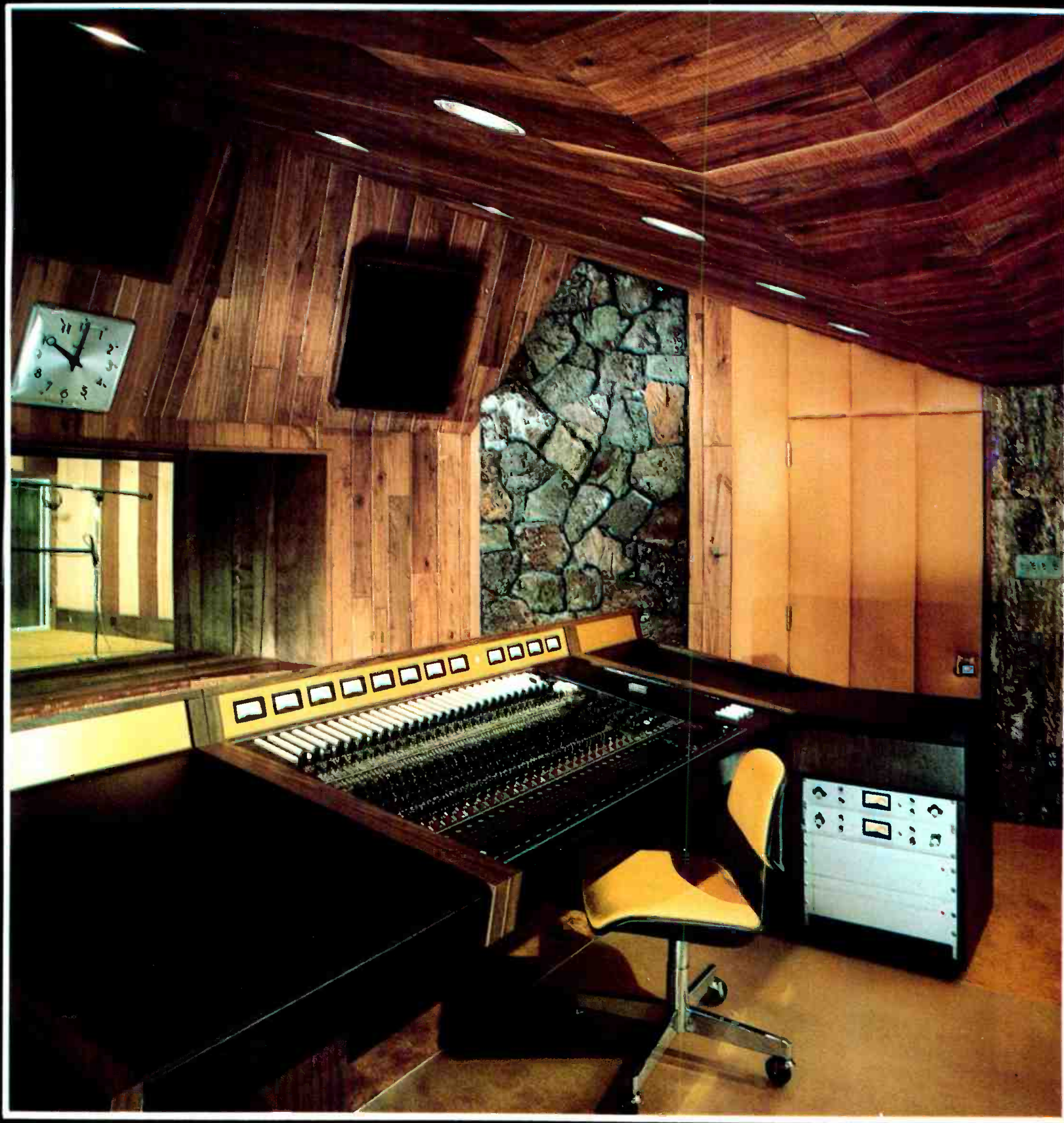
April 1976
Volume 7 — Number 2

The Cover . . . magnified 2,000%, Gary Kleinman's 35mm photo is a colorful microcosmatic view of a mag head before final assembly.



A SPECTRA SONICS Model 1024-24 was recently installed in BONNEVILLE PRODICTIONS' multi-track control room. Mike Collett, Director of Engineering, notes that the high reliability and outstanding performance of SPECTRA SONICS control consoles and audio products that the firm has been using over the past nine years, was the deciding factor in their choice. The Salt Lake City production house was designed to provide artists and agencies *state-of-the-art* electronics in a stimulating environment.

SPECTRA SONICS appreciates the confidence displayed by BONNEVILLE PRODUCTIONS. Address inquiries to:

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Introducing The Ice Cube.

It can go all day and all night and still keep its cool. Here's why:

One, there's a super quiet, thermally activated two-speed fan that runs low most of the time, but kicks into high when the going gets hot. (And, at a short 5¼" tall, The Ice Cube is perfect for stacking.)

Two, there's an absolutely exclusive 2000-watt solid-state inverter power supply instead of those massive transformers you're used to. Total weight: 35 pounds!

There's more. 300 watts RMS per channel, both channels driven into four ohms from 20Hz to 20 KHz, at .05% or less total harmonic distortion.

Color-coded peak reading lights step up and down so you're the first to know if it's clipping.

Go see The Ice Cube. It's formal name is the JBL 6233 Professional Power Amplifier. Bring $1500 and it's yours.

JBL

Circle No. 105

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 4x4 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

Box 6600, Buena Park, CA 90620

Circle No. 106

LETTERS and LATE NEWS

from . . . BRUCE LOWELL
NBC–TV
BURBANK, CA.

"Apparently writer Berliner does not have sufficient information as to the wonderment and complexity of the subject of television audio. To say that 'the audio portion of TV holds few surprises for the recording engineer portrays misinformation to the reader. Nearly all end users and distributors of video gear can write a short article about video theory as many texts are readily available. However articles on audio for television is not a subject that one can find current literature in great abundance.

Aside from simple situation comedies using two booms to a mixing board, most audio requirements for TV specials, game shows, variety shows and musicals are areas which require specialists in just the field of TV audio. The techniques are entirely different from recording, and as a former location recordist I can attest to the fact that techniques and skills of record recording have no bearing whatsoever in television. The requirements are completely different. Although we may use the same commercially available equipment, our approaches to its useage is completely different. The techniques used in providing separate control for the simultaneous on-air mixing and house sound reinforcement for both performers and talent stymie the imagination when taking into consideration the elements of time allotted for production and the pressures that occur, and lack of rehersals. The audio engineer in TV works under a set of rules and with a crew that understands television, not record making.

Instead of mixdown we go directly to tape with a product that needs editing and then sweetening. Both the processes of editing and sweetening have come a long way in the past two years due to mini-computer assistance; it would behoove the reader to refer to many articles and papers on these latter subjects."

ED — Thanks, Mr. Lowell for your capsule view of some of the significant differences between record and TV recording.

However, the article "TV FOR THE AUDIO MAN" was plainly not meant to explore the authors' expertise, or lack of it, in the field of *TV recording and mixing*. Perhaps, reader Lowell misses the point of the article by Oliver Berliner and Rob Lewis. The intent of the article, as the authors plainly pointed out, was simply to address that portion of the audio community who have little opportunity to work in the TV discipline; affording them a familiarity with the principles of

TV TRANSMISSION. What was described are just the basics . . . and are certainly less than must-reading for those whose everyday work involves the intricate technical process by which TV's non-prescription sedative is delivered to the populace.

Although not so stated, the article is the first of a number which will explore TV recording and mixing, in some detail. . . . and, although there is no question that the subject is indeed *wonderous and complex* the editors can't help but observe that the recording studio community having so far been *the* audio innovators, should not, indeed, be unprepared for the kind of surprises which TV can have in store for them.

NATURAL SEMI INVADES SUBJECTIVE MARKET WITH SOLID-STATE GNOMONIC UTOPIAN AND RECONDITE ULTIMATE DEVICE

SANTA CLARA, Calif. – *April 1, 1976* – In the serene tranquility of the corporate ashram meditation room, amid wafting incense and the sounds of sitars, technologists of Natural Semiconductor Corporation today quietly unveiled an ultimately evolutionary solid-state device that will open vast new unmanifest frontiers to the field of integrated electronics.

Named by its inventors as the "Gnomonic Utopian and Recondite Ultimate" Device (known by its initials as the "GURU DEV"), the advanced electronic circuit is the first example of CSI (Cosmic-Scale Integration, which is infinitely better than large-scale stuff).

The "GURU DEV," which has been designated as the model "OM-108," is expected to find wide use in providing subjective evaluation and advice for people in all walks of life – much as the computer provides objective evaluation of data.

Dr. Morris Breakthrough, for whom many technical advancements have been named, explained that he and his colleagues, Dr. Poisson d'Avril and Swami Tritop Catchabanana, have been working for years on methods of applying what they call the Principle of Bestowed Objectivity.

"You see," Dr. Breakthrough said, "in our laboratory push at the limits of objectivity, the barriers between us and the unknown grew so fine that they became interrelated with us and with our perception of them. When we dismissed them from our minds and shut off our computers, the limitations vanished.

"It was a perfect application of the Principle of Vernal Equivalence to the age-old problem of selective indifference, and through these mechanics of inactivity we found that our knowledge of objectivity depended on our state of mind and was therefore wholly subjective.

"Through a rather ingenious application of the Heissluft Theory of Positive Retrogression (HTPR), we were able to build a mathematical model of Platonic Subjectivity. It was then a simple matter for d'Avril and Catchabanana, here, to reduce the model to practice in the form of the GURU DEV Model OM-108 integrated circuit."

In a typical Empathuter*, the GURU forms the heart of a section known as the "Compressed Narthex Shrove Unction System" (named by its initials CONSHUNS). When CONSHUNS is consulted for subjective evaluation, the GURU accepts input from the CNPU (Central Non-Processing Unit). The CNPU essentially does nothing and thus prevents distortion and worsening of the situation.

Within seconds, the GURU circuit electronically requests comparison information from the FOM (Fish-Only Memory), which fishes for rembrances of things past. Results of the comparison are electrically transferred to the WOM† (Write-Only Memory), where they are stored permanently and never seen again.

*Note: an Empathuter is analogous to a "subjective computer," such as IBN's Subjugator TM System LXXXVI.

†WOM is a trade mark of Frenetics Corp.

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WAKEFIELD ANNOUNCES PREMIUM DISC LINE

Wakefield Manufacturing, a manufacturer of quality records for the past 20 years, has announced the establishment of its highest quality disc line, called "TQ."

"The TQ line was established for those clients who require the highest quality, wide dynamic range pressings, that the state of the art permits," explains Richard Wakefield, President and General Manager of the Phoenix based firm. According to the firm, the difference between their regular pressings and the TQ pressings will not lie so much in the audio quality of the final product but in the use of very stringent quality controls and procedures to be followed for the TQ disc throughout the manufacturing process.

These procedures begin in Wakefield's mastering department where the lacquer to be used is selected on the basis of 1/3 octave noise measurements and listening tests. The cut lacquer is then prepared for plating and within one hour after cutting is in a specially controlled plating tank. According to the company, the immediate processing of the freshly cut lacquer helps insure quiet surfaces and minimizes groove echo problems. The metal parts produced are then sound checked and if found to be acceptable are used to press the record. The initial pressing, again made in specially controlled presses for the TQ line, is completely sound checked, measurements made on surface levels, and only after all TQ criteria is met is the production run started. Hourly sound checks are made during the run, stamper changes are made frequently, and all pressings are visually inspected a second time before being sleeved and jacketed.

"While in some cases the finished pressings will not exceed the quality we normally produce, these new procedures assure that a job that has to have our best product gets it," explains Mr. Wakefield.

SPECTRA SONICS EXPANDS HOLLYWOOD OFFICE, PROMOTES BRUCE BALL TO DIVISION SALES MANAGER

As announced by Spectra Sonics president William Dilley, Mr. Bruce Ball has been promoted to Division Sales Manager while Mr. Brian Morze and Mr. Steve Cannon have been added to the engineering sales staff. Mr. Cannon will be specializing in Sound Reinforcement having formerly been Chief Sound Engineer for Forest Lawn Memorial Park where he was responsible for over 200 events (symphonies, concerts, etc.) in the Hall of Liberty and various other installations within the park complex.

Mr. Morze formerly with General Electric Corporation's X-Ray Equipment Division, has a Bachelor of Science degree in Physics from University of California at Los Angeles and has an extensive background in the audio field. Additionally, Ms. Julie Wahnsiedler has also been added to the Hollywood staff in the capacity of receptionist/secretary.

Spectra Sonics Hollywood Division is located at 6430 Sunset Boulevard, Hollywood, CA 90028, (213) 461-4321.

SYNERGETIC AUDIO CONCEPTS DESCRIPTIVE BROCHURE

Synergetic Audio Concepts has a new six-page brochure which describes their 3-day nationwide sound engineering seminars. The brochure includes a full schedule for 1976 seminars in 11 different cities.

These seminars, now in their fourth year, receive enthusiastic support from their graduates. About 1/3 of the graduates have attended the seminars twice, and many for the third time.

To receive a free brochure and for further information, contact Don Davis, **SYNERGETIC AUDIO CONCEPTS, P.O. BOX 1134, TUSTIN, CA 92680. Ph.** (714) 838-2288.

AUDIO ELECTRICAL SUPPLY INC., FOUNDED

Bill Jones, president of the newly formed Audio Electrical Supply Inc., announces the company's entrance into the professional recording equipment sales and studio construction field.

Several new products are also being introduced for the first time by the company, these include the all new APD1600-16/24 track studio recorder/reproducer manufactured by the Audio Products Division of Bouse Manufacturing Company, of Newport Beach, CA., and a studio monitor speaker system designed by George Augspurger of Perception, Inc.

AUDIO ELECTRICAL SUPPLY, 15466A CABRITO ROAD, VAN NUYS, CA 91406, (213) 787-3679/873-3929.

NORTRONICS ANNOUNCES NEW ALLOY EXTENDS LIFE OF MAGNETIC HEADS

Development of a new wear-resistant alloy now makes it possible to significantly extend the life of magnetic heads, according to Dr. Steven Bendson, Manager of Research and Development for Nortronics.

Dr. Bendson announced this week that after two years of extensive testing at Nortronics, he has confirmed the long-wear properties of the new alloy. The alloy, a high permeability type magnetic alloy named Wear-Resistant Hy Mu 800*, was developed by Carpenter Technology, of Reading, Pa.

Nortronics will be the first magnetic head manufacturer to offer the new alloy in its products.

"Tests have shown that the new material wears up to ten times longer than conventional high permeability materials with no significant sacrifices in head performance," Dr. Bendson said. "Heads made with this new alloy will mean more reliable equipment and less maintenance. Replacement heads will be required much less frequently, resulting in fewer service calls."

An outstanding feature of the new alloy, Dr. Bendson added, is its workability, which permits it to be used to form case shields as well as cores, a property not found in other long-wearing permalloys used in the industry now. This allows construction of heads with a larger percentage of wear-resistant material in contact with the recording media.

*Wear-Resistant Hy Mu 800 is a trade name owned by Carpenter Technology.

NORTRONICS COMPANY, INC., 8101 TENTH AVE., N., MINNEAPOLIS, MN 55427.

DUNCAN ANNOUNCES CREATION OF SIERRA AUDIO

The recent announcement was made to formalize the representation of Tom Hidley's EASTLAKE AUDIO in the U.S., Canada, Central and South America and Australia/New Zealand.

The goal of the newly formed Burbank based company is a fresh approach to package studio design, construction, and equipment installation. Among the unique services provided is "in house" training of studio maintenance and operating personnel for turnkey installations to learning a proven format for studio success.

Sierra additionally announces exclusive representation of SPHERE ELECTRONICS for studio and disc mastering consoles in the Western United States, and as foreign agent. The close affilliation will guarantee the desired state of the art and expedient delivery, according to the statement by Kent Duncan

Equipment distributed by Sierra includes 3M, Dolby, Orban, Inovonics, Crown, Shure, Roger Mayer Electronics, MicMix, Koss, Eventide, UREI, and dbx.

While studio design will center around the Eastlake Audio Monitor System, Mr. Duncan empahasizes that acoustical design of installations using other monitor systems is available.

SIERRA AUDIO, 625 SOUTH GLENWOOD PLACE, BURBANK, CA 91506 (213) 843-8115.

JAMES M. HOLLON APPOINTED SALES SUPERVISOR, PROFESSIONAL RECORDING & BROADCAST MARKETS, MAGNETIC AUDIO/VIDEO PRODUCTS DIV., 3M COMPANY

Hollon joined 3M in 1967 and has held sales positions in the Southern California area. His most recent assignment has been in Los Angeles serving professional recording and broadcast accounts.

AUDIO ENGINEERING SOCIETY

54th Convention Program

May 4 – May 7, 1976
Los Angeles Hilton Hotel

TUESDAY, MAY 4 –

9:30 AM
Session A: DISC RECORDING & REPRODUCTION

2:00 PM
Session B: AUDIO in BROADCASTING AM/FM/TV

Session C: ELECTRONIC MUSIC – 1

7:00 PM
Session D: ELECTRONIC MUSIC – 2

EXHIBITS OPEN: 1:00 PM – 9:00 PM

WEDNESDAY, MAY 5 –

9:00 AM
Session E: MAGNETIC RECORDING

1:30 PM
Seminar 1: SOUND REINFORCEMENT

2:00 PM
Session F: SIGNAL PROCESSING

3:30 PM
Seminar 2: SOUND REINFORCEMENT

7:30 PM
Session G: MOTION PICTURE SOUND

EXHIBITS OPEN: 1:00 PM – 9:00 PM

THURSDAY, MAY 6 –

9:00 AM
Session H: SOUND REINFORCEMENT

1:30 PM
Seminar 3: ECHO & REVERBERATION

2:30 PM
Session J: ARCHITECTURAL ACOUSTICS and ROOM DESIGN

3:30 PM
Seminar 4: COMPRESSORS

7:00 PM
SOCIAL HOUR & AWARDS BANQUET

EXHIBITS OPEN: 1100: AM – 5:00 PM

FRIDAY, MAY 7 –

9:00 AM
Session K: AUDIO STANDARDS

Session L: SPECIAL APPLICATIONS IN AUDIO

2:30 PM
Session M: TRANSDUCERS

EXHIBITS OPEN: 11:00 AM – 5:00 PM

STEREO MIKING TECHNIQUES USING COMPUTER PATTERN ANALYSIS

by
STEVEN B. FULLER

Often in recording orchestras, small groups or even solo performers I have used two mikes on the same stand at some angle to each other to achieve a natural stereo image. Typically I use the ORTF system where the mikes are spaced about six inches apart at an angle of 90 to 110 degrees. Other times, when the mikes are closer to the source, I might use the X-Y system where the mikes are at about 90 degrees but the elements are very close to each other. In using any system of two mikes close to each other to achieve a natural stereo image you are probably far enough from the source that the reverberant field and other sounds from the rear need to be considered. The problem then is to find out how to select mikes that best suit the acoustic environment.

The Author

STEVEN FULLER received his degree in electrical engineering from Washington University in Saint Louis, where he is currently a research associate in the Mechanical Engineering Department. His present responsibilities include management of an aerial air polution mapping system and its telemetry and computer interfacing, as well as design of improved air pollution monitoring equipment.

He owns MULTISOUND, a recording studio dedicated to the pursuit of excellence, through engineering . . . "before pressing the record button."

Mr. Fuller holds a patent on LDR expanders and compressors using incandescent light sources.

He also serves as advisor to University students in independent study of recording. He has been a consultant for several radio stations, colleges and other recording studios on acoustical and electronic design problems.

As a first order analytical look at the situation I decided to make graphs of the polar response of the sum of two mikes at some arbitrary angle to each other. In order to do this two things were required. The first was a computerized graphics routine which David Patterson, a graduate student in systems science and mathematics at Washington University, worked up. The second necessity was a mathematical expression of a mike's polar response. The expression, $1+B\cos\theta$ when plotted logarithmically in polar form, can be used to approximate a large number of real mike patterns by varying B.[1] If B is small or zero we have an omnidirectional or pressure mike and if B is large tending to infinity we have a pure bidirectional or velocity mike. I used B=100 as my "large" value. Fig. 1 shows a mike of B=100. If we set B=1 we have a perfect cardioid as shown in Fig. 2. So we see that the cardioid mike is someplace between the pressure and the velocity mikes. While I won't attempt to quantify it in this article, proximity effect increases as B increases (at low frequencies) though not necessarily in linear fashion. If we set B=1.4 we have the hypercardioid mike shown in Fig. 3. Obviously if we vary B between 1 and 100 we get various amounts of "hyper." If we set B=0.8 (Fig. 4) we find a cardioid that has a rear rejection that might be considered good in a real mike. So as we vary B between one and zero we see the cardioid lose more of its rear rejection.

In Fig. 5 we show the resultant sum of two cardioid mikes at an angle of 90 degrees. All of these plots assume the two mike elements are close enough together to ignore phase difference effects. Stated another way, the curves are valid for frequencies where the spacing of the elements is not a significant portion of a wavelength. The wavelength λ is given by

$$\lambda = c/f$$

where c=1129 ft/sec for air at 70 F.[2]

continued . . .

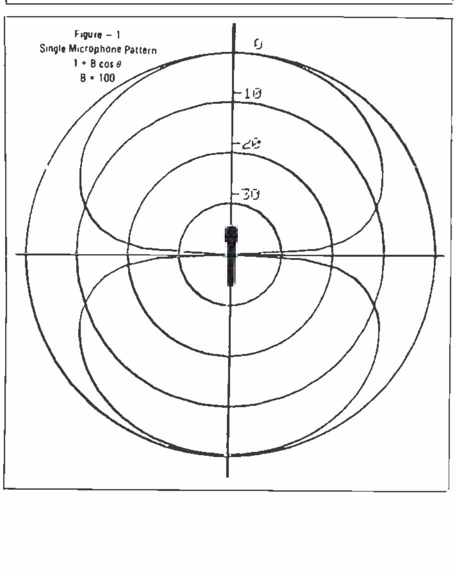


Figure – 1
Single Microphone Pattern
$1 + B\cos\theta$
B = 100

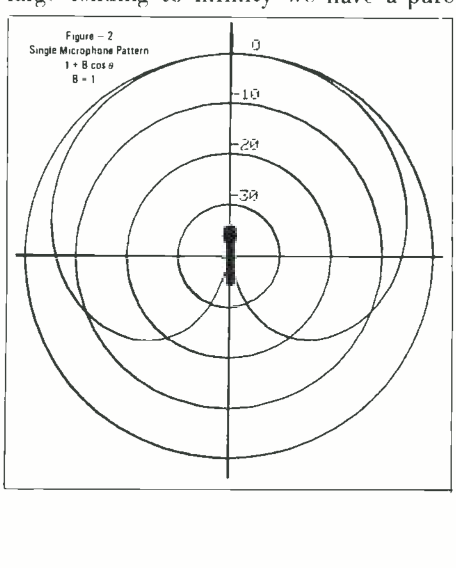


Figure – 2
Single Microphone Pattern
$1 + B\cos\theta$
B = 1

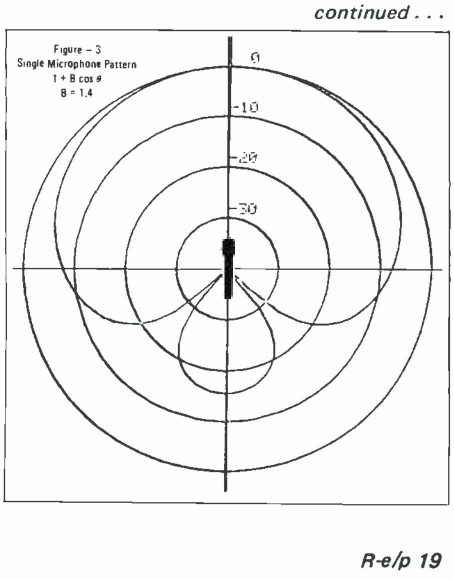


Figure – 3
Single Microphone Pattern
$1 + B\cos\theta$
B = 1.4

If we have two cardioids at 180 degrees we get a resultant omni pattern. In fact if we have any two similar mikes of the form (1+B cos θ) at 180 degrees, the resultant pattern is omni. As B gets large, however, the system gain approaches zero. This can be visualized by considering two figure eight mikes and recalling that the two lobes have opposite polarity.

Things got more interesting when we made the plot of a figure eight and a cardioid at right angles. Fig. 6 shows this sum which is one side of a matrixed MS system. Recall that in the MS system the left and right channels are synthesized by taking the sum and difference in such a configuration. Fig. 6a shows the difference signal. So far we have assumed the same gain and sensitivity for both mikes. Notice that the resultant hypercardioid looks much like the B=1.4 shown in Fig. 3 but displaced by 45 degrees. If we now take two B=1.4 hypercardioids at 90 degrees the resultant as shown in Fig. 7 is a cardioid. In fact it can be shown that for B= 2 the cardioid resultant is exact and that the MS sum or difference has B= 2 (again for equal sensitivities and gains). An interesting sidelight is that if we have back-to-back MS systems (or four B=1.4 mikes at right angles) we get a resultant omni pattern but quad pickup with excellent imagery.

If you are interested in economy of mikes, channels or phase you can do quad with only three mikes (one figure eight and two cardioids) and appropriate matrixing. Since there are fewer mikes where B=1.4 at all frequencies than good approximations to B=1 or B=100 at all frequencies, you will have to weigh the trade-offs involved in using synthesized mikes versus only fair approximations to B=1.4.

While we are on the subject of quad we should recall that two B=1 cardioids at 180 degrees have a resultant omni pattern with unity gain. Two B=1.4 mikes at 180 degrees will also be omni but with lower gain. So if we consider a cluster of four mikes at right angles for quad we find ourselves better off in the image department with four B=1.4 mikes. This is because it takes two right angled B=1.4 mikes to give one cardioid and the resulting two cardioids then give the omni pattern at unity gain. In other words we cover the circle once with the hypers and twice with the cardioids.

If we continue working with the MS

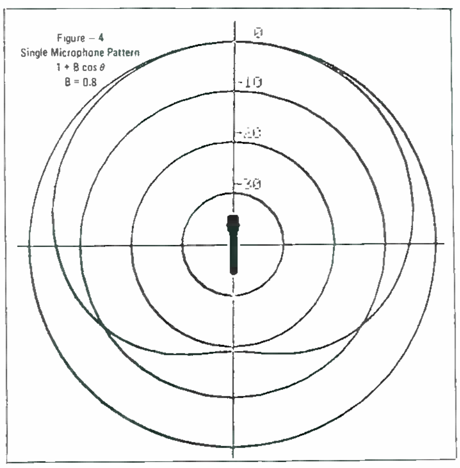


Figure – 4
Single Microphone Pattern
1 + B cos θ
B = 0.8

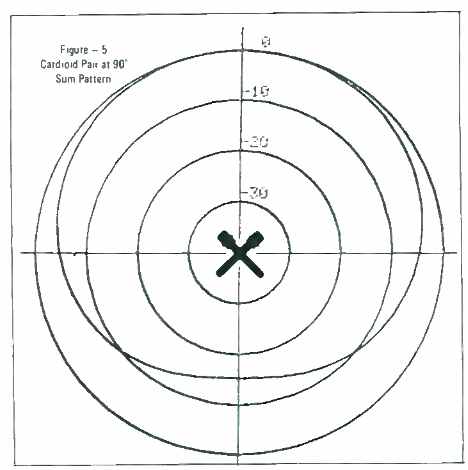


Figure – 5
Cardioid Pair at 90°
Sum Pattern

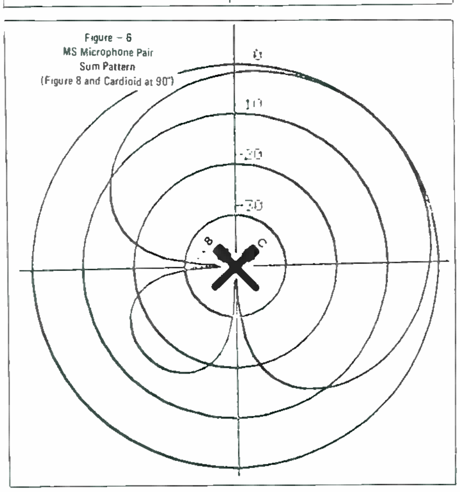


Figure – 6
MS Microphone Pair
Sum Pattern
(Figure 8 and Cardioid at 90°)

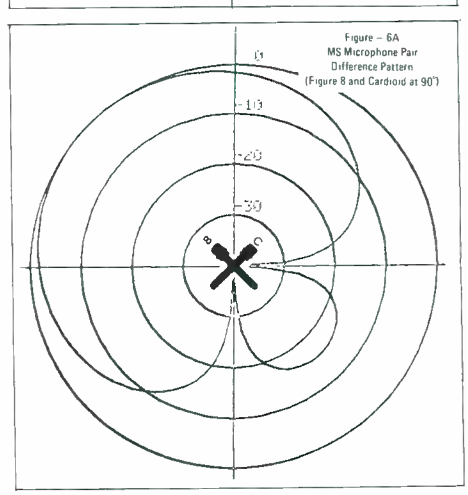
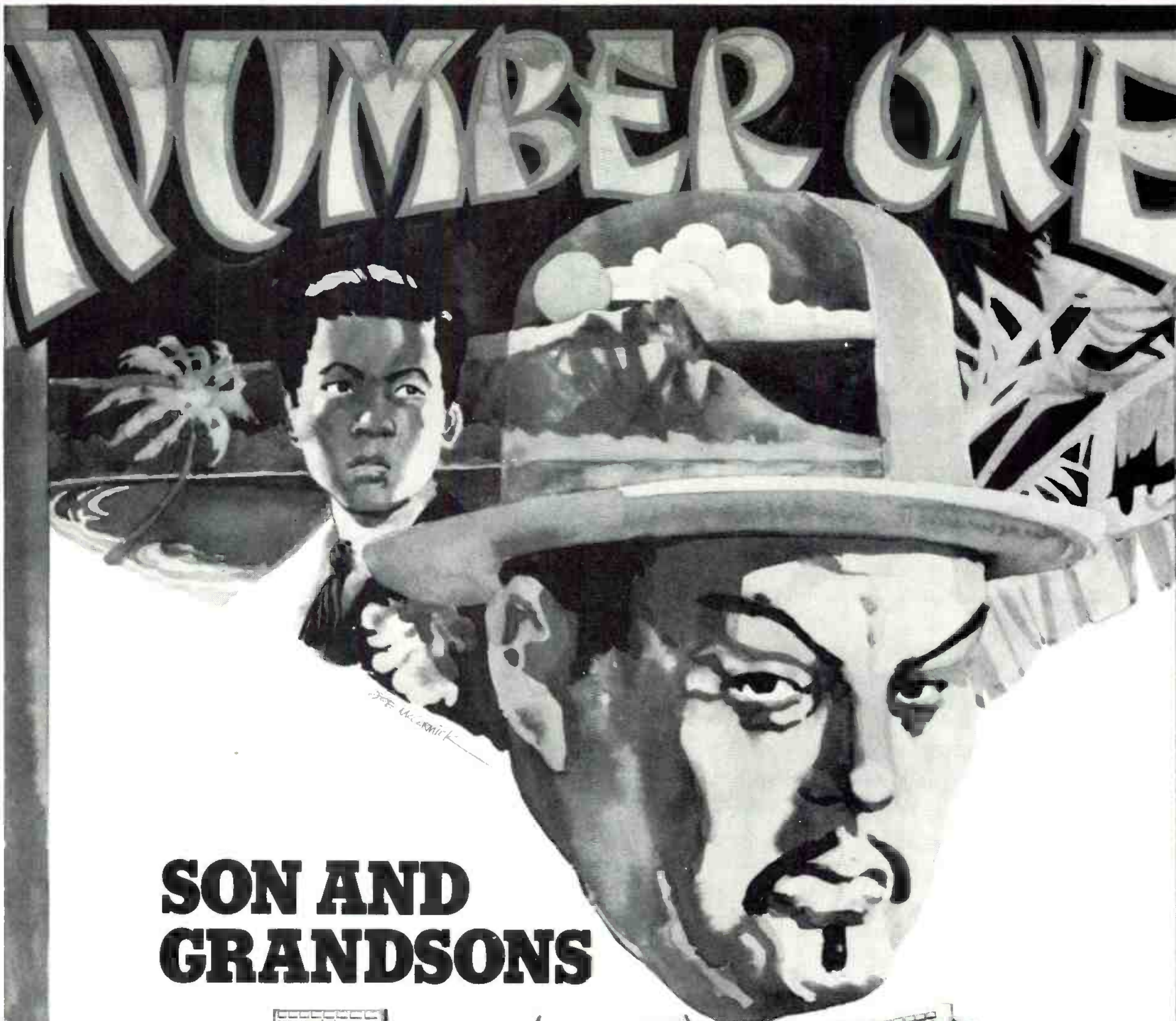
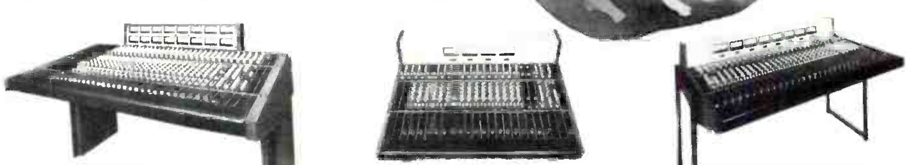
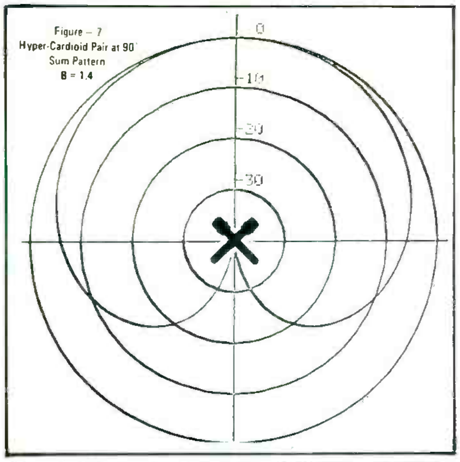
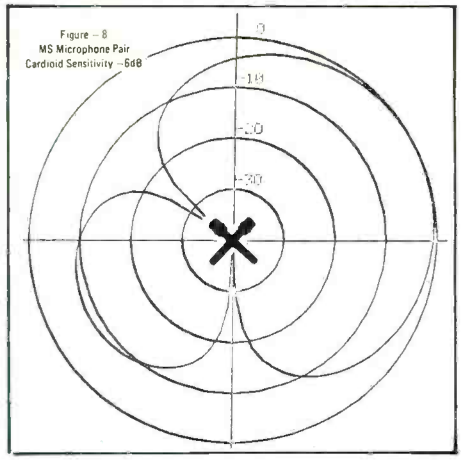


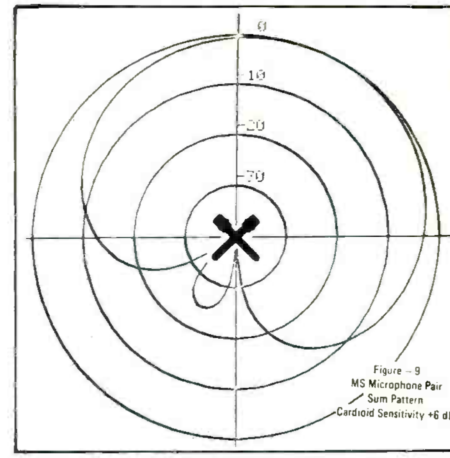
Figure – 6A
MS Microphone Pair
Difference Pattern
(Figure 8 and Cardioid at 90°)


NUMBER ONE









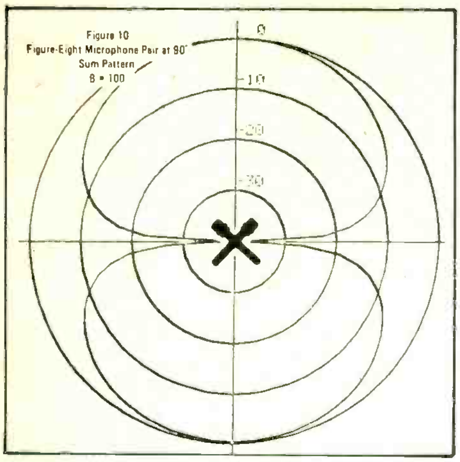
system by changing the relative gains of the mikes we see that reducing the cardioid (middle) mike by 6 dB produces a greater effective stereo angle between the matrixed channels and a greater rear lobe on the equivalent mike. (Fig. 8) If we increase the cardioid gain by 6 dB we narrow the effective stereo angle and reduce the rear lobe. (Fig. 9)

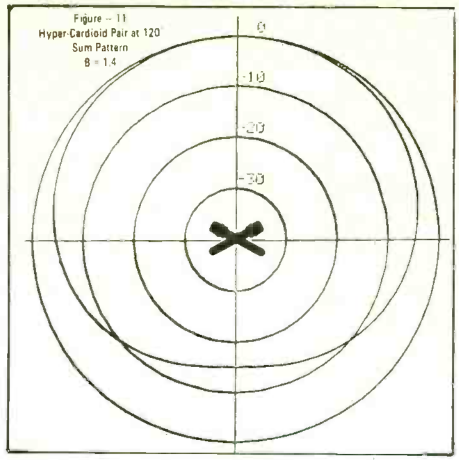
Now we come to the combination that surprised me most, two B=100 figure eights at right angles. The resultant sum shown in Fig. 10 is also a figure eight but with slightly reduced side rejection. If we calculate the B value of the sum we find it to be about 35. It may be well to mention here that the greatest sensitivity of the pattern to small changes in B occurs near B=1, hence the insignificant change between B=35 and B=100.

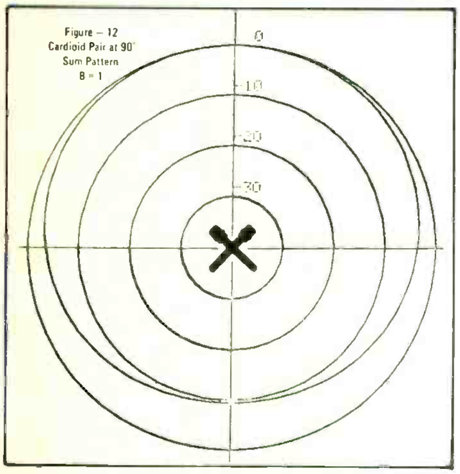
Having dealt with several major factors we can now look at more subtle effects. Figure 11 shows two B=1.4 hypers at 120 degrees. Note the similarity of this sum to Figure 5. Obviously, we have achieved the greater separation of the large angle but the same rear rejection of the cardioids at 90 degrees. For reference, Figure 12 shows two cardioids at 90 degrees. If we move the mikes together we approach the pattern of a single mike, and Figure 13 shows the intermediate position of 60 degrees for two cardioids.

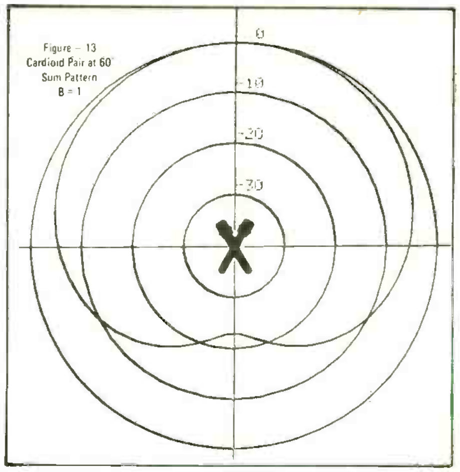
While it is obvious for certain cases that the patterns obtained for two mikes can be rotated around their principal axes for a correct three dimensional pattern, I am wary of any generalized predictions. We are now in the process of developing a three dimensional routine and hope to have the results available in the near future.

A practical application of these studies came up the first day we had the program







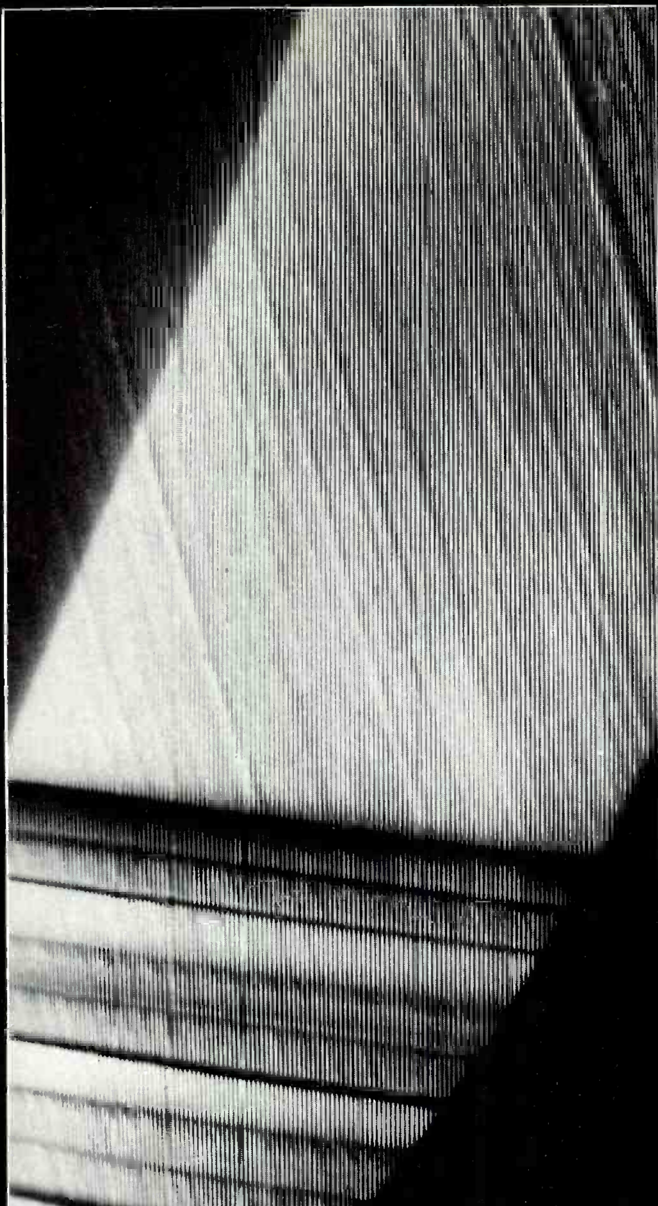


running in selecting the microphone type for the overall ORTF pickup for the St. Louis Symphony broadcast recordings. Gregory Gomard of the St. Louis Symphony Society came to see the graphics system in operation and decided to switch from cardioid mikes to hypercardioid (B=1.4) to see if such a change would eliminate some excess hall reverberation and audience noises. While a thorough evaluation is not yet complete on the change, the audience noises have been audibly reduced. There is better string tone and the balance is more even. The improvement in the strings may be partly due to more accurately placing the mikes at 90 degrees when the patterns were changed.

The entire exercise had given me a new respect for both the hypercardioid and figure eight microphones. The hypercardioid really does have a purpose other than feedback prevention, or leakage prevention. For me it was very exciting watching my imprecise notions about multiple mikes disappear, and I hope others will find this work useful.

1. L.L. Beranek, Acoustics, New York: McGraw Hill Book Co., 1954, p. 181
2. Ibid., p. 7 & p. 10

Circle No. 118



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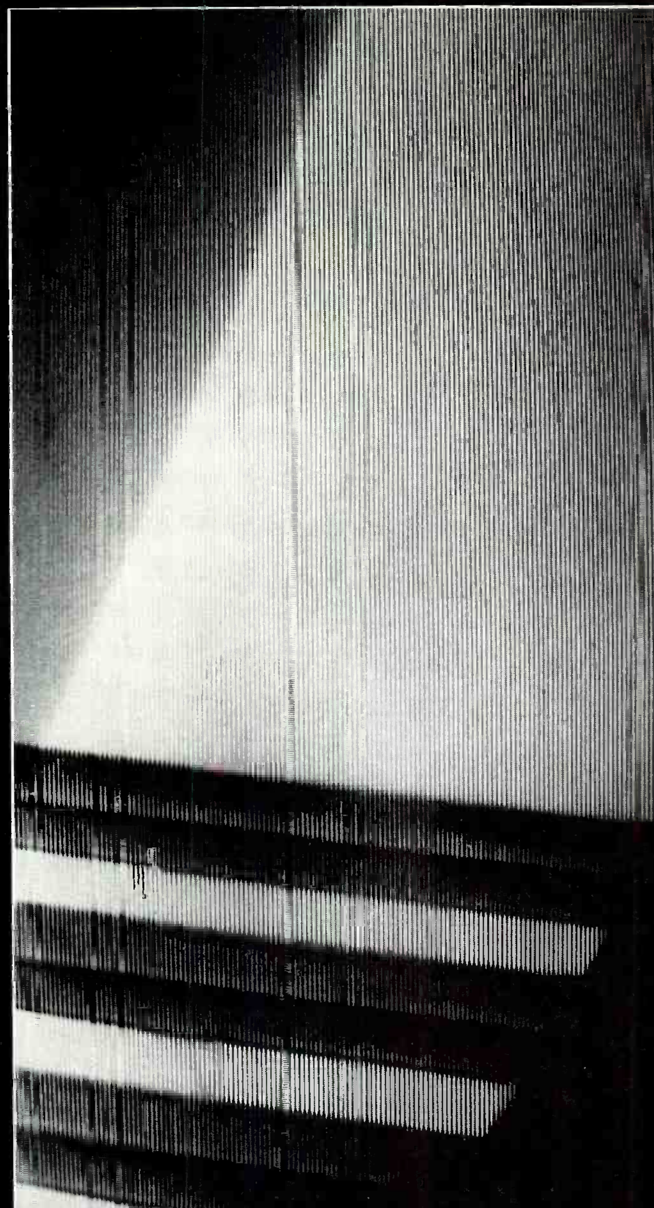

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Circle No. 119

Magnetic Tape Heads — FERRITE vs METAL

by
Walter Scott
Cheif Engineer
Saki Magnetics
Santa Monica, California

One of the more important advances in magnetic tape recording in recent years is the development of the ferrite head. This head differs from a conventional, laminated metal head in two important respects. Ferrite and non-magnetic materials which form the tape contact surface of the new head are hard, low porosity ceramics. These materials wear more slowly than metals and retain smoother surface finishes. While metal heads are held together by heat sensitive epoxies, ferrite heads are bonded with hard, stable glass.

The greatest advantage of ferrite over metal is cost. While the former may cost 2 to 3 times the latter, the life of a ferrite head is 10 times that of a comparable metal head.

Ferrite heads, by virtue of the physical hardness and high electrical resistivity of the core material, avoid many of the defects of metal heads. Figure 1 is a magnified view of the face of a metal, high speed duplicator head after several hundred hours of use. The vertical line at the center of the picture is the recording gap. The three gray, horizontal bars on either side of this gap are core laminations. These laminations are .002 inch thick in order to reduce head core, eddy current losses within the head.

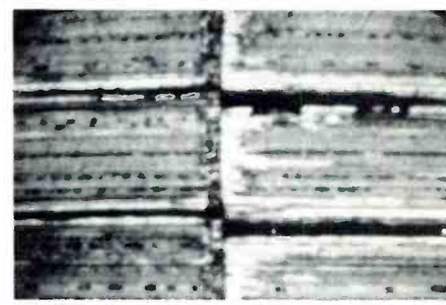


FIGURE 1

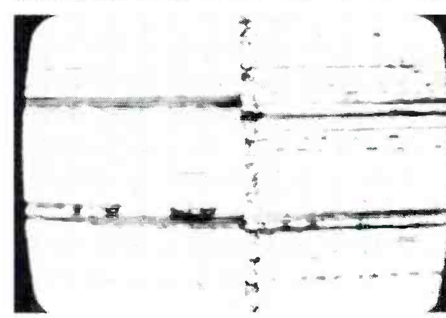


FIGURE 2

The black horizontal lines between laminations are glue lines, adhesive layers which bond the individual laminations into a core and provide electrical insulation between them. The important fact to consider in viewing this picture is that recording does not take place at any region of the gap intersected by a glue line. Much of the available width of the track made by this head will not be recorded. The playback output from this track will be lower than it could be had there been no glue lines.

The light areas to the right of the gap-line reveal another defect common to metal, high speed duplicating heads. The core edges are light because they are not contacting the tape. Heat produced by the high frequency bias signal core losses

ANATOMY OF A MAGNETIC HEAD

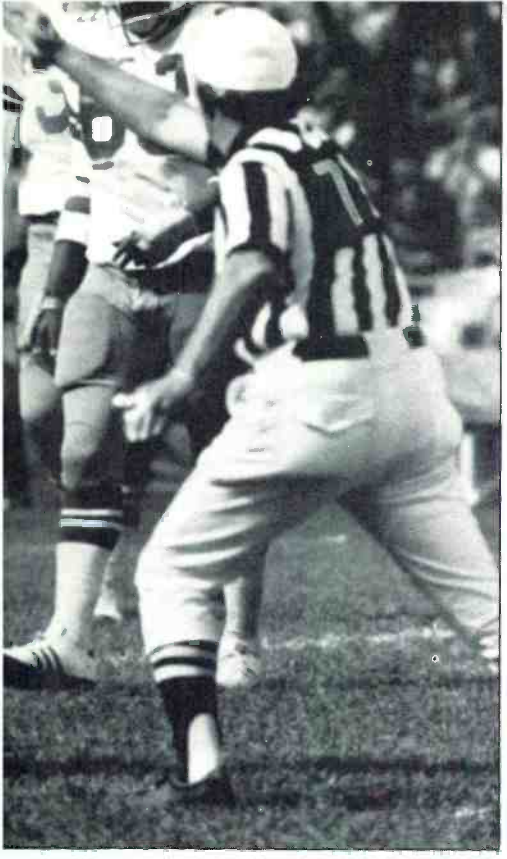
by
Peter Butt

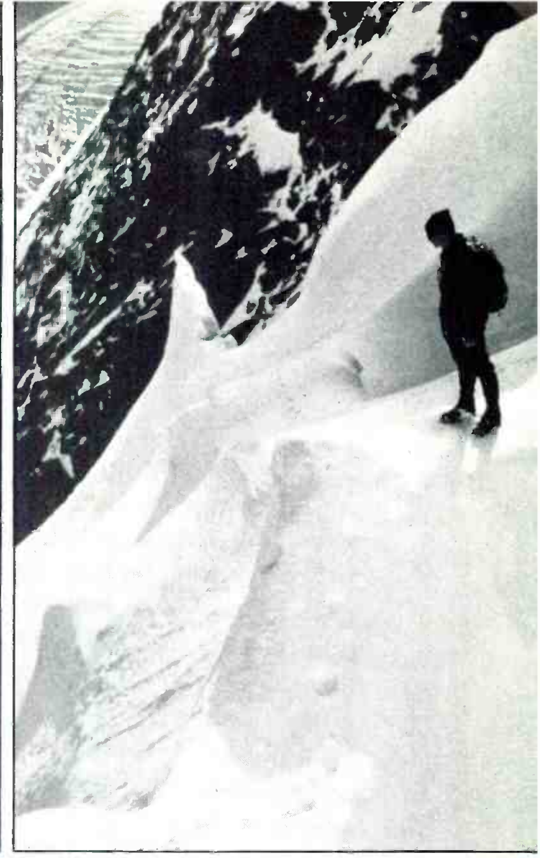
Magnetic tape heads are devices constructed for the conversion of dynamic magnetic fields into electrical energy and to convert electrical energy into a magnetic field. The ability of a magnetic head to accomplish this in a manner appropriate to the specific application is based upon magnetic, electrical, geometric and mechanical factors of the head design. These are highly complicated considerations about which much can be said. There are general aspects of magnetic heads designed for magnetic recording and reproduction of sound that can be described in fairly loose non-technical terms.

Most magnetic heads used in audio applications are of the "ring" type. That is, they are constructed as a ring-shaped core of some material that is capable of sustaining a magnetic field within it very easily compared with the ability of air or free space to do so. The ability of a material, magnetic or otherwise, to permit a magnetic field to exist within it is called its "permeability." This property is analagous to conductance in the case of an electrical conductor.

The ring-like head core is generally constructed in two symmetrical halves rather than in the form of a continuous torous. Each semi-torous is wound with a number of turns of wire. It has been observed that the existence of a magnetic field implies the existence of an electrical current and that an electrical current flow implies the existence of a magnetic field. The turns of wire, then serve as a means of generation of a magnetic field within the magnetic core by the passage of an electrical current through the coils. These same coils will respond to the change of the magnitude of a magnetic field within the core by yielding an open-circuit voltage that is proportional to the rate of change of that field. This two-way situation permits much of this description to apply to magnetic heads for recording as well as for reproduction.

The idea behind the design of a record head or erase head is to convert electrical energy passing through the head core windings into a magnetic field that can be used to induce some degree of relatively permanent magnetization to some kind of magnetic emulsion that has been made to adhere to a ribbon of flexible plastic material that can be moved across the head in close contact with it. A reproducing head is designed to perform the reverse process. If a magnetic field retained by such a magnetic emulsion system is passed over the head core in such a way that the changing magnetic emulsion field can be captured by the head core, the changing core field then causes a small open-circuit voltage to be generated in the core windings.





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have acted upon the heat sensitive epoxy which holds the head together. One side of the head has moved relative to the opposite side across the gapline. Because the tape is now separated from the core on one side of the gap, the recording field is no longer symmetrical to the gap. This field has been shifted from its original location so that the point at which recording takes place has been moved. A step at the gap of one head in a multi-head stack is rarely accompanied by identical steps at the gaps of other heads within the stack. This condition produces phasing errors between stereo pairs.

A higher magnification of the same gap is shown in Figure 2. Smearing and erosion of the gap edge are caused by recording tape abrasion. The lack of straightness of the gap edge produces a degradation in short wavelength response. The core edge on the right side of the gap is relatively straight because of the absence of tape contact. A metal head without the defect of a step at the gap will have irregular edges on both sides of the gap.

The tape contact surface of the metal head is rough and grooved as a result of tape wear. The dark shading over the core area is a film produced by the tape oxide and/or binder system reacting with the core metal. This film cannot be removed except by repolishing the head. The film and roughness of the head face wear the tape and produce a tape motion variation known as scrape flutter.

Figures 3 and 4 show the face of the ferrite head used as a replacement for the metal head of Figures 1 and 2. The corresponding magnifications are equal. Both heads have had equal use. Note the straight edges of the ferrite cores at the gapline. Scratches across the gapline are visible at the edges indicating that no step exists at the gap. The ferrite core is solid so there are no glue lines intersecting the gapline. For equal tape distortion and track width, the ferrite head track will produce a greater playback level than the metal head track because the entire width of the ferrite head track is recorded.

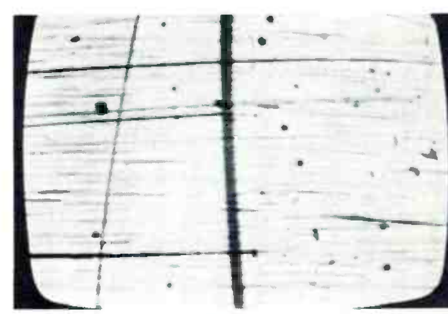


FIGURE 3

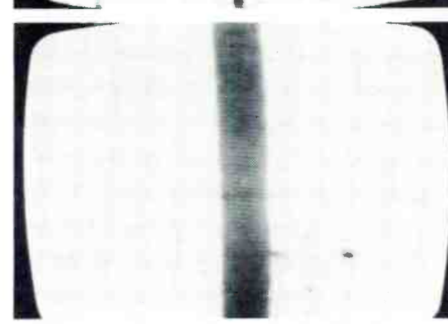


FIGURE 4

The straight edges of the ferrite head will produce better short wavelength response than the irregular edges of the metal head's gap. The ferrite cores are smoother than the metal cores, and no film adheres to the ferrite. The ferrite head will therefore not wear the tape as rapidly as the metal head. Also, scrape flutter due to the ferrite head will be measurably less than that due to the metal head.

With these advantages plus lower cost per hour of use, it may seem strange that all recorders are not equipped with ferrite heads. Some knowledge of the evolution

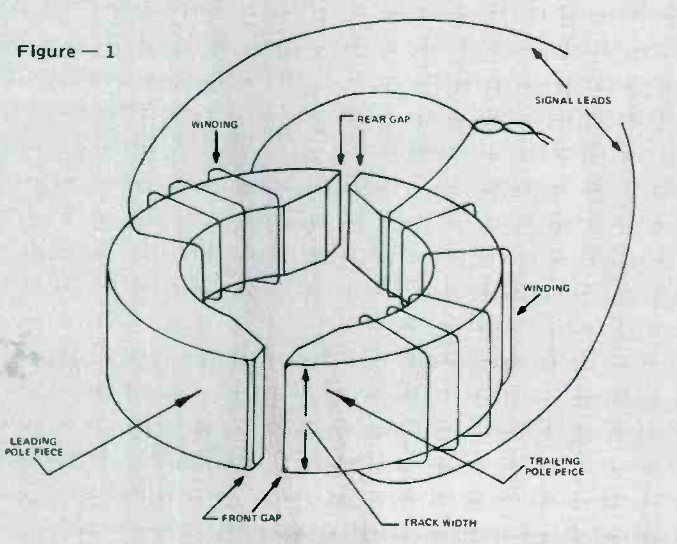
It has been found that a greater proportion of the magnetic field induced within a record head can be exposed to the aforementioned magnetic emulsion system, if the record head core is not a continuous construction of permeable material. If a slot is cut completely through the core at a point where it will come in contact with the tape, a fairly large portion of the magnetic field within the core can be caused to escape from the core and to penetrate the magnetic tape. Similarly, in the case of a reproducing head, the existence of such a slot in the permeable ring core will tend to permit a portion of the tape magnetic field to enter the reproducing head core. This captured magnetic field will then serve to induce an electrical open-circuit voltage in the head core windings. This opening in the magnetic head core is commonly referred to as a gap.

Although fairly small, physically, the dimensions and geometrical characteristics of the gap opening are of critical importance in the performance of the magnetic head. Additional factors contributing to the performance of the head are the magnetic properties of the core material, its means of construction, its geometry, the existence of any other gaps within it, and the nature of the windings around the core.

The drawing in Figure 1 shows a generalized magnetic head having a core composed of two symmetrical halves. Each half has a winding about the long dimension of the core. In practice, the core halves are made separately and then fitted together to form the ring configuration. The gap is formed by inserting a thin sheet of non-magnetic material between the edges of the core halves at the point(s) where they would contact each other otherwise. The spacer is always present in the gap that is intended to be in contact with the tape. A spacer may or may not be included in the gap at the rear of the head core assembly.

The head assembly of Figure 1 is shown without any of the external construction that retains its geometry in a fixed state and protects the core assembly from mechanical damage. This external construction also serves the purpose of providing the means of mounting the head to the tape system of which it is to become a part. The packaging may also serve the function of contributing to the shielding of the core assembly or assemblies within it from external magnetic and electrical influences. This, again, is another subject that could be extended considerably, but will be neglected for this discussion.

Figure — 1



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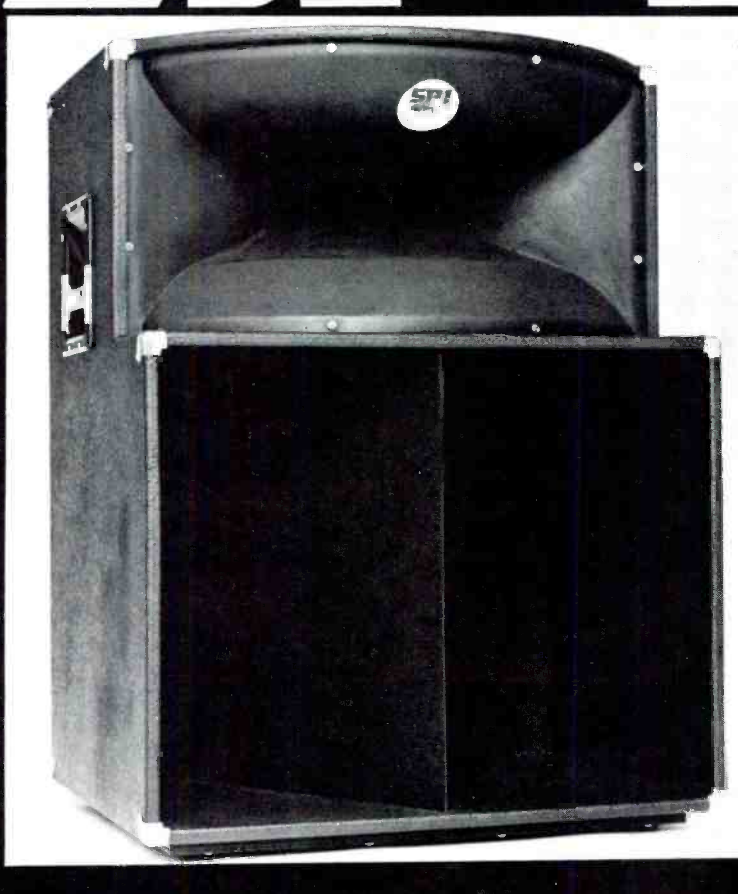


Circle No. 121

of the ferrite head may explain this inconsistency.

When ferrite heads were first developed in the early 1950's, gaps crumbled in less than a hundred hours. Head faces contained holes due to voids in the ferrite, and crystals of the weak ferrite were pulled out of head faces by tape friction to produce more holes. These problems were overcome by a new type of gap and development of stronger ferrite materials.

GLASS BONDED GAPS

The fault of the early ferrite head gap spacer was that it expanded with temperature (due to head-to-tape friction) at a much higher rate than the ferrite. The result was like water freezing in cracks of concrete sidewalk. The expanding spacer fractured the ferrite. The problem was solved by a device used in ceramics industries, a temperature compensated glass seal. By using a gap spacer composed of a glass adhesive that expands at the same rate as the ferrite, forces which had eroded the gap were eliminated.

STRONGER FERRITES

Most ferrite is made by the high temperature chemical reaction of thoroughly mixed metal oxides in the exact amounts required to produce a desired ferrite composition. The resulting material is then crushed and milled to fine powder, mixed with a binder, and molded in a press to a desired shape. It is then sintered in a furnace where the binder evaporates leaving voids in the material. During sintering, ferrite particles form crystals of the size required to produce desired magnetic, electrical and physical properties. The double firing is necessary to reduce shrinkage.

A significant improvement in ferrite for magnetic tape heads is a material called *isostatic pressed ferrite.* In manufacturing this material, no binder is added to the milled ferrite powder. Instead, the powder is placed in a sealed rubberlike mold and subjected to very high hydraulic pressure in a liquid filled vessel. The pressed ferrite is strong enough to maintain its shape after removal from the mold and through sintering. Although superior to prior ferrites, isostatic pressed ferrites are still subject to cracks and voids caused by shrinkage during sintering. However, many ferrite heads made for the consumer market today contain material manufactured by this process.

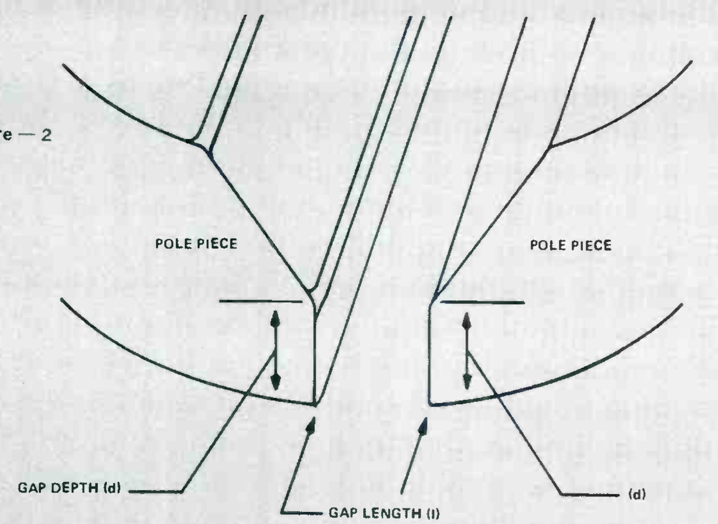
A second advancement in the method of sintering ferrite produced *hot pressed ferrite.*[1] In this process, the powdered ferrite is compressed by a ram *during* sintering. Not only is the resulting ferrite stronger than the isostatic (cold) pressed material; it is more free of cracks and voids. It is this material which best eliminates the problems of earlier ferrite heads, gap erosion and surface porosity.

Unfortunately, at the time ferrite heads reached this stage of development, the design and construction of new recorder and duplicator systems had declined due to reduced customer demand. Today, many owners of equipment designed for metal heads are replacing these heads with ferrite heads for lower head replacement cost and better system performance. In the changeover, equipment owners and head manufacturers have been forced to solve a number of compatibility problems caused by two electrical differences between ferrite and metal cores.

The first of these differences is eddy current loss. The second is a loss of core permeability with increasing frequency.[2] Both losses are negligible in ferrite, but substantial in metal.

Core permeability loss at high frequencies is not as well known as eddy current loss and deserves a closer look. Permeability loss occurs in magnetic materials which are good electrical conductors. An alternating magnetic field produces eddy currents in proportion to the conductivity of the core material. These currents which increase with frequency produce a magnetic field opposing the original field,

Figure — 2



Continuing the discussion of the gap, there are matters of general nomenclature concerning the geometry of the gap itself. The length of the gap is a highly important factor in head performance that rarely escapes mention in literature related to magnetic heads and magnetic recording. The length of the gap is measured along the direction of tape travel, as shown by the dimension, l, in the gap detail drawing of Figure 2. The gap depth, the thickness of the core material at the gap edge, is shown as dimension d. The width of the gap is generally specified indirectly by reference to the "track width" of the head. The gap must necessarily extend across the full width of the area to be recorded or reproduced on the tape.

Magnetic head gap lengths are fairly small in comparison with other head dimensions. Audio heads for use in reproducing applications have gap lengths of from about 200 micro-inches to 50 micro-inches or less. The ability of a head to reproduce recorded wavelengths that are very short is inversely proportional to its gap length. Record head gap lengths range from about 200 micro-inches to about 1000 micro-inches or greater. The ability of the record head to saturate the entire thickness of the magnetic tape oxide coating is directly related to the size of the gap length dimension. Gaps that tend toward the magnitude of the tape oxide thickness used are generally more desirable for recording than those with small gap dimensions. Record gaps tend to be larger than their corresponding reproducing counterparts, as a general rule. Gaps found in conventional audio heads are generally too small to be observed in detail with the unaided eye.

The head core is constructed by one of two methods. In the case of a metallic core material, the metal is formed into thin sheets of about 1 or 2 mils of thickness. These thin plates are then cut to size and stacked upon one another until a gross thickness equaling the required track width is obtained. The thin metal laminations are bonded together by an adhesive between the plates that also serves to insulate them from each other.

Ferrite heads are generally made from a single piece of ferrite material that has been formed to the proper size and shape.

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an effect strongest at the center of the lamination cross section. At high frequency, the magnetic field of a metal lamination is confined to its surfaces. The higher the frequency, the less deep the field penetrates the lamination surfaces. In other words, the effective cross section of the core decreases as frequency increases. In practice, the inductance of a laminated metal head may decrease to one third of its signal frequency value at bias frequency. Compatibility problems derived from differences in behavior of the two core materials differ from record to playback heads.

RECORD HEAD

A metal core record head can be represented by the circuit of Fig. 5.

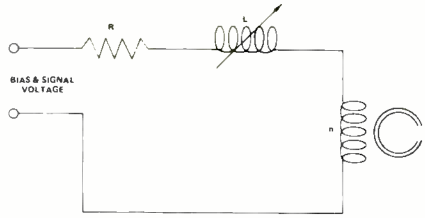


FIGURE – 5
METAL RECORD HEAD EQUIVALENT CIRCUIT

The recording field at the head gap is proportional to current passing through a head coil of n turns. Variable resistance R represents frequency dependent head loss. Variable inductance L represents head coil inductance change subject to core permeability loss. This inductance may vary over a 3 to 1 range between signal frequencies and bias frequency.

Head inductance must satisfy a bias driving circuit in at least two ways. Head impedance at bias frequency must be low enough to produce the required bias current from the available bias voltage. The resonant frequency of the head and associated cables should exceed the bias frequency. A third requirement exists if the head is tuned at the bias frequency. It must resonate with the bias coupling network.

When replacing a metal record head with a ferrite head, *inductance* at bias frequency must be duplicated if the metal head was resonated at bias frequency. If resonance is not required, identical bias frequency *impedances* of the metal and ferrite heads will best load the bias driver. Inductance decrease at bias frequency occurs in a metal head, but not in a ferrite head. Therefore, equal metal and ferrite head inductances at bias frequency require that the ferrite head have fewer turns than its laminated counterpart.

Bias fields produced by both kinds of heads are of equal magnitude because turns reduction in the ferrite head is offset by reduced losses in that head. However, this is not true at signal frequencies where losses in both heads are small. At these frequencies the ferrite head with reduced turns requires more current than the metal head because recording field is the product of current and turns. Signal electronics must be capable of supplying this extra current. Signal current requirements are small, and extra current is not slways a problem.

In some cases where a record head is *series* tuned to the bias source, the low losses of the ferrite head present too low a load impedance to the bias driver and distort the bias waveform to produce increased bias noise on tape. This problem is corrected by adding a resistor in series with the head coil in the head connector. The resistance is low and should not change signal drive requirements.

A potential disadvantage of ferrite often cited is its low maximum flux density, B_{ms}, of approximately 4000 Gauss. This is less than half of the B_{ms} of 4-79 Permalloy, 8700 Gauss. The latter material is used in almost all laminated metal heads. The argument against the use of a core material with a low B_{ms} is the possibility of saturation of the recording head pole tips at the point of highest flux density, the gap edges. Pole-tip saturation limits the effective bias field that the head can produce.

When pole-tip saturation occurs, the following phenomenae are observed. The bias level-versus-play-back output level curve is unusually broad. Record level set to produce a specified tape distortion (i.e. 1% third harmonic distortion of a 1-mil wavelength signal, etc.) will produce less playback level than normal.

The pole-tip saturation characteristics of ferrite high-speed duplicating heads appear to be superior to these characteristics for laminated metal heads. Specifically, ferrite heads used with high coercivity tapes of 540 Oersteds show no indication of pole-tip saturation. This apparant inconsistency might be explained by the fact that eddy-current shielding in metal head laminations restricts the bias flux to the surface of the laminations. This reduction in available cross-sectional area of the magnetic circuit increases the flux density in the usable flux path. This condition does not exist in ferrite for reasons discussed above. In this manner, the lower B_{ms} of the ferrite core appears to be offset by a greater available cross-sectional area at the high frequencies used by high-speed tape duplicators.

PLAYBACK HEAD

A metal core playback head can be represented by the circuit of Fig. 6.

Flux change from tape passing across the head gap produces a voltage which appears in series with frequency-dependent inductance L. Frequency-dependent loss, R, shunts the head output voltage,

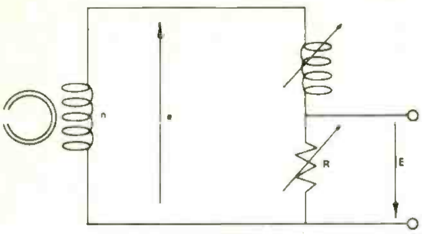


FIGURE – 6
METAL PLAYBACK HEAD, EQUIVALENT CIRCUIT

E_O.

In a metal head, frequency dependent core permeability provides a desirable effect whenever head turns are limited by head-input circuit resonant frequency. The inductance decrease permits a metal head to have more turns, and thus a higher output voltage, than would be possible otherwise. A ferrite head does not exhibit much inductance decrease at high frequencies. When a ferrite head must have a resonant frequency greater than the metal head it replaces, the ferrite head is limited to fewer turns. Ferrite head voltage output at mid and low frequencies will then be less than that of the metal head.

The combination of L and R for a metal head attenuates high frequencies. Playback equalizers designed for metal heads correct this loss. Ferrite heads have less high frequency loss and do not require as much equalization as metal heads do. With ferrite head the reproduce equalizer may be set to reduce high frequency gain and thereby reduce high frequency system noise. Also, the peak frequency of the equalizer may be increased to extend system bandwidth. The combination of these two advantages permit use of record and reproducer equalizations having less reactance for the case of a ferrite recording system as compared with a metal head recording system. This would imply that a ferrite-equipped record/reproduce system could be expected to exhibit less phase distortion than a metalic head system.

If a playback equalizer lacks the adjustment range needed to compensate a ferrite head, the circuit can be modified, unless prohibited by company policy. In the latter case, a resistor may be added across the head, preferably in the head connector. Head inductance and the added resistive load produce a low pass filter to pull high frequency output back into equalizer range. Electrical advantages of the ferrite head have been sacrificed, but long life and other advantages of the ferrite's physical properties remain.

CONCLUSION

By now ferrite replacement heads have been designed for many popular studio recorders and tape duplicators. However, when a recorder owner needs to specify a replacement head, the head manufacturer should be sent the characteristics of the original metal head and the circuits used with that head. Type and length of cable between head and circuit should be stated. If a playback head is tuned, frequency of resonance should be given. Bias current and voltage should be stated with units of measurement qualified as rms or peak to peak.

Lower head cost and a competitive demand for extended recorded bandwidth are currently accelerating the use of ferrite heads. A knowledge of differences between ferrite and metal heads, and potential compatibility problems, can help the recording engineer bridge the changeover at minimum cost and lost time.

1. H. Sugaya, "Newly Developed Hot-Pressed Ferrite Head," IEEE Trans. on Magnetics, Vol. MAG-4, No. 3, pp 295-301, Sept.
2. V. G. Welsby, "The Theory and Design of Inductance Coils," John Wily & Sons, (Great Britain), 1960, p.94.

The Author

Since his graduation from the University of California at Los Angeles in 1961, WALTER SCOTT has worked as an electronics engineer specializing in the design of magnetic recording heads for Westrex Co., RCA, and Ampex. He has been cheif engineer of Saki Magnetics, Inc. since 1972.

AUTOMATED MIXDOWN

the second generation

by
ALLISON RESEARCH

Programmable mixing techniques have presently reached the point of industry acceptance where I think it is safe to assume that the art is here to stay.

In the design of our first generation systems, we had no previous experience to draw upon, so we went ahead in the direction we saw as most fit. At that time, our design goals were, basically, to provide the audio system with a means of remembering the positions and motions of the console controls. It was not difficult for us to foresee at that point that the industry would demand automation far beyond the control of levels, so we configured a system which could handle up to 256 analog functions, a number which represented just about the maximum possible capacity, consistent with usable scan times.

The workability of Allison Research first generation systems can be attested to by literally hundreds of hit records produced on consoles whose automation ranges from "level only" through nearly "total automation," in most continents of the world.

The purpose of this article, however, is not to discuss past performance, but rather, to pick it apart and find the shortcomings, in an effort to prescribe a more refined second generation approach.

SPEED vs. CAPACITY

The paramount key to improved system performance lies in the method employed to store the data itself. In early devices it was logical to time-share the functions, in a simple sequential fashion, and serially apply the scans of data to the tape machine for storage. The limiting factor here, of course, is the length of time required to make one scan of the console elements.

More functions = longer scan times.

It was this speed versus capacity situation which caused us to employ the unique quinary (5 level) cooling in our

continued on page 36 . . .

by
AUTOMATED PROCESSES

Automation, or more specifically remix memory, is now well established as an effective aid to the recording engineer in optimizing the artistic and technical aspects of a mix derived from a multitrack master. In effect, remix automation provides the extra hands. the infallible memory, and the split second response necessary to build up a perfect "no compromise" mix in a minimum of time.

In January of 1973, Automated Processes, Inc. delivered the first fully programmable audio mixing console. In the intervening years this console and many others of similar design have accumulated tens of thousands of operating hours in prominent studios throughout the world, with success proven by hundreds of hit recordings. The operating experience of scores of mixing engineers, maintenance technicians, and studio operators working with this automated remix equipment under actual field conditions, has resulted in several suggestions for improvement while verifying the validity of many of the original design concepts.

These comments and suggestions have been incorporated into a newly designed automation system called AUTOMIX.

The design of Automix was based on the following concepts:

1. The console must function as a conventional high quality manual system. Performance characteristics must be consistent with the current state of the art with respect to distortion, frequency response, noise, etc.

2. All controls must respond in a familiar and comfortable manner. It should not be necessary to "learn to play a new instrument" in order to take advantage of automation techniques.

3. Reliability and maintainability must not be sacrificed. Not only must the system be built of proven reliable components, it must also be designed with redundancy to permit patching

continued on page 46 . . .

by
RUPERT NEVE

What makes a 'Hit'? Is it the music, the personalities involved or maybe the promotion it receives? How significant is the balance engineer's contribution? Can mixdown be considered as an art form, as truly creative?

A consideration of the power that the modern recording console confers upon the balance engineer prompts these questions, for there can be no doubt that he msut be considered a partner in the creative process.

Has "Automation" any place in mixdown then? In the full sense of the word, not at all, since if we allow the engineer a creative role, technology must not be allowed to eclipse it.

Nevertheless, the fact that one man, however skilled, has to cope with such a large array of controls has inevitably given birth to a number of more or less successful systems aimed at assisting in this task, to give the engineer more hands and a better memory.

Before embarking on the design of any such system, it is, of course, essential to study the main tasks involved in mixdown, so as to distinguish between those involving skill and artistic judgment, and those properly the province of an electronic aid system. A brief summary of such a study will help in the appreciation of the system which has been designed to fulfil the requirements emerging from it.

The Mixdown Task:

It is frequently the lot of a balance engineer to be given a tape to mix which he did not record. Thus at the first stage, he will be concerned to listen to the music, perhaps to identify the track content and gradually to build up a mental picture of what he seeks to achieve.

At this stage, tracks will be sampled individually so as to set equalization, and pan position where appropriate. At

continued on page 52 . . .

continued from page 35 . . .

first generation programmers. The combination of quinary encoding and word by word validation produced accumulated delays on the order of one-sixth that obtained by bi-phase, frame validation methods, and made complex degrees of automation possible.

However, the somewhat analog nature of the code placed rather stringent requirements on the alignment and general quality of the storage medium and, of course, the delays, while drastically reduced, were still a limiting factor.

ENTER PRIORITY ENCODING

Upon re-examining the data requirements for mixdown purposes, we found that simple sequential scanning is not the ultimate answer. At any given instant, most parameters of the console will be found to be stationary, or not moving. Thus, there is no point in wasting valuable time in constantly refreshing these static parameters with new data, until such time that they are changed.

This premise holds true regardless of the complexity of the system. If we were to configure a system which was able to detect changes in the console controls, and give changing parameters priority over the static ones, there would be no degradation in the effective speed of the system as more and more functions were employed.

To further understand this principal, let's analyze the operation of a really complex console which has, say, programmable levels, echo, panning, E.Q., noise-gating, limiting, assignments, patching, etc.

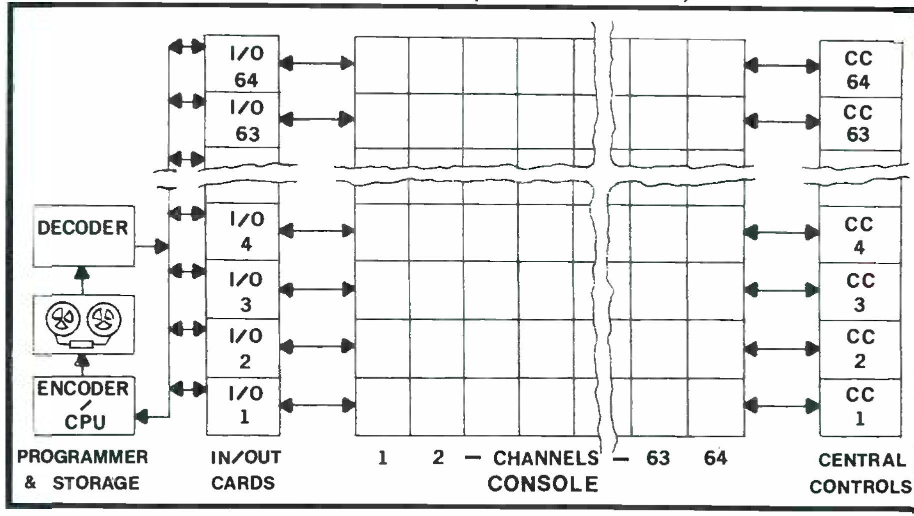
In spite of the fact that several thousand automation functions might be employed, operation would still consist of changing a level here, an equalizer there, and so forth. It is unlikely to expect more than two or three parameters to change at *any given instant.* Even in the unlikely event of the simultaneous change of 20 or 30 parameters, the delays would be negligible.

With this improved base to work with, it now becomes practical to utilize a more stable form of coding the data on the magnetic medium without encountering the previous problem of speed vs. capacity.

THE ALLISON 65K PROGRAMMER

In configuring the new 65K programmer, we chose to define the overall system capacity as 65.536 binary bits, a number which, although certainly sufficient to handle the most complex console, we do not consider as unrealistic in view of today's rapidly changing technology. The data stream consists of individually addressed and validated digital words, each word containing a 12 bit address, 16 bits of data, a parity bit and a parity time interval. Each such word is normally associated with some particular location in the console such as "channel No. 3 level, mute, solo and group assignment sub-section." The 12 bit address defines the systems capacity as 4096 sources of data, each source having a 16 bit data capacity.

Typical configuration of Allison Research MEMORY PLUS system operating at maximum capacity (65,536) bits). Each of the 4096 console blocks represents a 16 bit sub-system.



The addressing system is broken down to matrix form such that the console, or other programmable source, is thought of as having 64 rows of data producing elements, each row having 64 discreet 16 bit locations. (64 rows X 64 columns = 4096 locations.)

For purposes of standardization, we define the first 48 columns of any row as being audio channels No. 1 through No. 48, while the remaining 16 columns are designated as master functions. (E.G. group masters, echo return channels, etc.)

Standardization is further implemented by defining the function of each row, as far as present peripherals allow. Row one, for instance, is designated to perform the task of programming faders, mutes, solos, input assignment and group master assignment. Row two handles echo systems, Row 3 performs the panning functions, etc., etc.

In short, the programmer is configured to handle consoles which, in standard form, contain up to 48 channels, and which may contain as many as 64 programmable sub-systems per channel. Obviously, by employing double rows, a 96 channel console may be implemented.

PRIORITY DETECTION

In order to determine which, if any, data sources are changing, it is necessary to examine all data in the system at periodic intervals. To achieve the minimum possible delays, we chose to examine the entire console at a rate coincident with the time required to encode one word onto tape. In other words, as one word is being encoded we search the entire system for changes, so that any such change may be encoded as the next word.

The time alloted for this tandem cycle is 3.2 milliseconds. (Since the thought of analyzing 65,000 bits in 3 milliseconds brings certain speed limitations to mind, a parallel processing approach is indicated.)

In practice, each row of data is processed by an individual in/out port which loads up the 64 potential 16 bit words once each 3.2 milliseconds. As each word is loaded it is compared, bit by bit, to the corresponding word received on the last examination cycle. If any difference exists, the address of the word in question is notated in a Random Access Memory (RAM), while the data itself, for each word, is stored in a second RAM.

Concurrent with this loading and comparing action (a parallel operation performed by the in/out ports), the master encoder continually searches the priority detection RAMs of all in/out ports, in successive approximation fashion, and locates the address of the next word to be encoded. When encoding time comes, the address is known and the latest data pertinent to that address is available in RAM memory. If multiple priorities exist, they are serviced in essentially the same order as received, with data encoded being the latest data pertinent to the changing parameter.

A secondary function of the priority system is to locate and notate word locations which have corresponding console sub-systems, so that unnecessary time is not spent in processing words

Two levels of priority then exist, "Priority one" being words which are changing and "priority two" being static words which are really there.

In order to assure that a constant stream of changing words cannot completely lock out the encoding of static parameters, the encoder is configured to allot at least one word in four to the performance of a sequential encoding of static words.

The net result is that any change in any console sub-system will be recognized and encoded within 3.2 milliseconds, assuming a singular change. In

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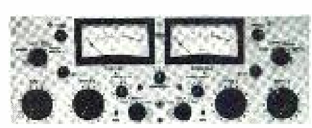

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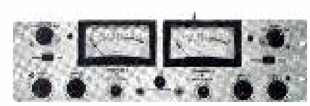

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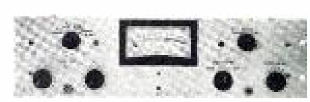

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the event of multiple changes, they will suffer a one time delay of 3.2 milliseconds per 16 bit word. As for the accumulation of delays with multiple passes of tape, the individual word validation method puts decoded data back into the console immediately. The total accumulated delay, taking into account the effects of the non-synchronous encode/decode cycle, comes out 4.8 milliseconds, regardless of the number of functions employed.

Although multiple simultaneous priorities undergo an access delay of 3.2 milliseconds per word, no additional accumulation delays are incurred beyond the nominally stated 4.8 milliseconds.

In comparing these figures with those of first generation Allison systems, it can be seen that 65K series programmers, even at full function capacity, operate at 1/20 to 1/30 the delay time of a 256 function first generation system.

It should be noted that this performance is obtained with simple data storage on spare tracks of the master music recorder itself, and without any synchronizing equipment or slave data storage mechanism. In facilities where synchronizing equipment is employed, accumulated delays may be eliminated entirely by the simple expedient of advancing the data playback head 4.8 milliseconds, with respect to the record head.

DATA STORAGE MEDIA

Basically, there are three catagories of data storage media from which we might chose. Each has its advantages and its draw backs. They are: Tracks of the master audio recorder, synchronized data tape recorders, and disk or drum memories. The master music machine offers the simplest method in terms of operating complexity and cost, and assures absolute synchronization, together with the guarantee that the data will not become misplaced in storage. The disadvantages are the imposed limitation on the number of separate mixes that may be stored due to track shortages, potential leakage of the code into the audio, and the necessity to give up at least two tracks of the audio machine.

Synchronized data tape systems alleviate some of the shortcomings of audio tape storage, but tend to increase the operation complexity of the system as well as to add costs for synchronizing equipment, digital tape recorder, and tape materials themselves. An often overlooked drawback of this storage medium lies in the inconvenience encountered in storing, filing and generally keeping track of which data tape belongs to which music tape.

While disc or drum memories are highly suitable for general purpose computer usage, this author is not strong on their use for the type of system herein described, because of their finite recording time, relatively high cost and, again, the storage and operational inconveniences.

With these thoughts in mind, Allison has chosen not to limit their second generation equipment to one specific storage medium, but rather, to configure it to work with any of the common data storage methods.

TAPE MACHINE REQUIREMENTS

When audio recorder storage is employed, the required bandwidth is 10 kHz (3dB point) the signal to noise or signal to cross-talk ratios must be in excess of 15dB, and speed variations requirements are plus or minus 50%.

In contrast to first generation programmers, which were somewhat prone to the production of error signals when subjected to excess amounts of cross-talk, improperly aligned tape machines, or defective tape, second generation 65K programmers are capable of tolerating the most adverse of conditions with absolute freedom from error production. Because of multiple methods of parity validation, on a word by word basis, operation is on a strictly go/no go basis, that is to say that the decoder will either produce the exact digital word that was encoded, or it will not decode at all.

It might also be noted that while first generation programmers wrere required to skip an entire scan in the event of a dropout, the 65K decoder, being individual word addressed, needs only to skip the word affected by a tape defect. This is coupled by its ability to properly decode drop outs of approximately 10 times the depth of 256 series equipment.

As far as the potential leakage of code into music, the 15dB signal to noise ratio requirement allows the code to be carried at extremely low levels, thus effectively eliminating this potential problem.

CONSOLE INTERFACE

Since first generation automation programmers were primarily designed to interface to adaptions of conventional consoles, their method of data distribution in and out of the console was in terms of analog control voltages, in the range of 0 to 5 volts. While this approach is probably the simplest and most understandable method, it is far from being the most desirable, particularly for complex consoles. The most profound shortcoming of an analog interface can be found in the path from the decoder – through the console – and back to the encoder. When multiple passes of the tape are programmed, any parameters which are left in READ MODE are passed through this decoder – console – encoder loop at each programming pass of the tape. While it is expected that parameters left in READ MODE will retain the exact settings to which they were programmed, this may not be the case with an analog interface. Accuracy errors of as little as .2% and or offset voltages of as little as 5 millivolts in the console path, the A to D convertors or the D to A convertors can cause the data to deviate by one or more steps each time it is subjected to the decode – console – encode path.

In first generation Memories Little Helper systems, this potential problem was dealt with by the expedient of employing .1% resistors together with offset adjusting potentiometers on each module. Although these systems do exhibit zero error accumulation with multiple passes, it is obtained only by a careful calibration of the entire system.

If the data is passed from Decoder to console to Encoder in digital form, the path is unquestionably accurate and totally stable without any need to resort to precision components or adjustments of any sort.

The second draw-back of an analog interface lies in the inefficient use of the available data. While a gain control, for instance, requires a finely resolved and essentially variable range of control voltages, many console parameters, such as switch functions, require but a simple on or off action – exactly that which is provided by one bit of direct digital interface. Other console parameters such as equalization can be much more flexibly configured with a digital interface, since the exact number of bits required for the job may be used.

In short, a digital interface allows the peripheral designer all possible control choices from 1 bit control (on-off functions), through 3 or 4 bits (for discreet 8 or 16 position switching), to 8 or more bits (for variable parameters such as level controlling). In each case, the digital interface allows an exactly defined parameter, which is absolutely repeatable and unaffected by component tolerances, offset voltages, ground loops and other analog pests which invariably appear in any console.

THE DATA BUSS

Now that we have settled on a high capacity (65.536 bit) programmer and a digital interface, the only logical next step is to employ a data buss approach in distributing data around the system. Buss oriented systems were created and perfected by the computer industry for the same reasons that indicate their use in programmable consoles. Instead of running tens of thousands of pairs of wires around the system, we can take a common wiring buss, containing only enough wires for the 16 bits of data and the 12 bits of addressing (28 wires), and time share it, in the same fashion that we time share the tape machine for handling all of the data. Since each console subsystem has a unique address (derived from the 12 address bits), it can be

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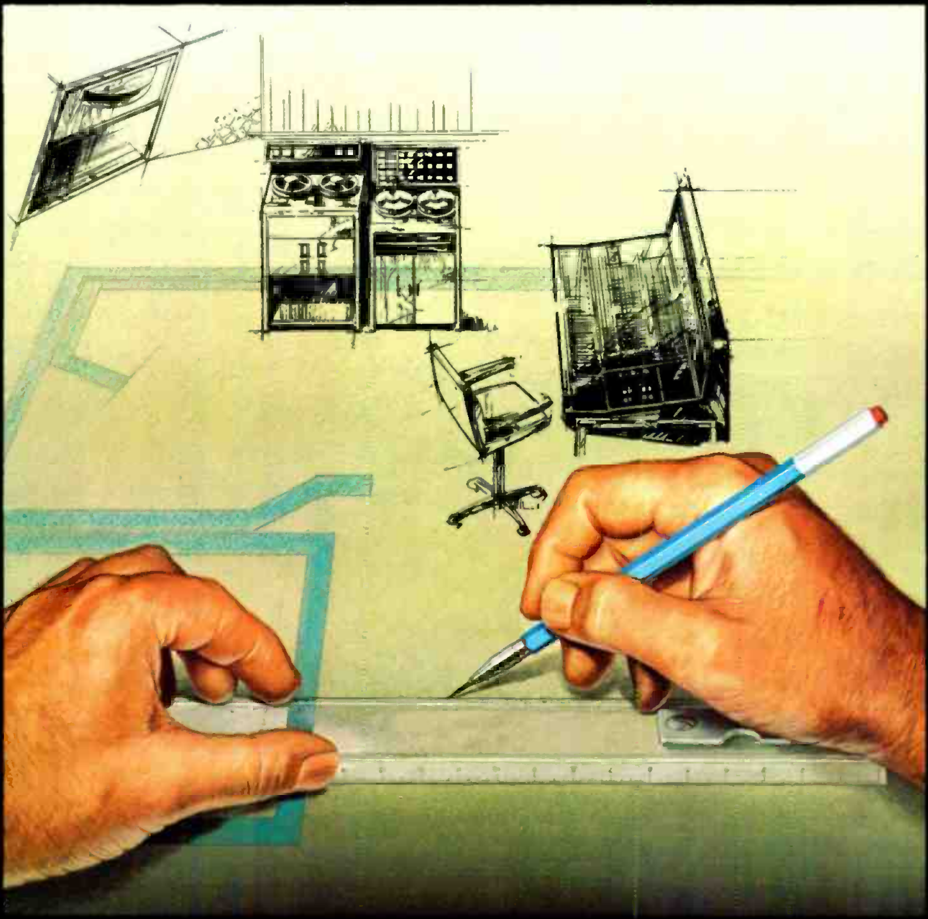
Acoustical Guarantee



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

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

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

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

Take care of Baker, he's incredible."

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

WE PUT OUR MONEY WHERE OUR MOUTH IS!

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

SYSTEM PERFORMANCE ACCEPTANCE

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

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

SOUNDS INTERCHANGE LTD.

By ____________

Dated DEC 6/74

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Continued from page 38 . . .

strapped across this common buss and instructed by two additional bussed wires. These instructions would be: Send data and receive data. The send data command means that the programmer wants the addressed sub-system to place its data on the buss for processing, while the Receive Data command indicates that the programmer is requesting the addressed sub-system to pick-up the data which is presently on the data buss. When a sub-system is neither sending nor receiving data, its connection to the data buss is effectively opened, thus allowing free use of the buss for other sub-systems.

A further benefit of such a system lies in the fact that, since all sub-systems share a common connection and have send/receive capability, it is now possible to re-configure the system, at will and without wiring changes, to transfer data from sub-system to sub-system or in and out of peripherals which may be added at some future time. This capability makes console design a whole new ball game, wherein early obsolescence may be avoided as newer forms of control are formulated. Instead of re-wiring, the console may simply be re-instructed.

While it is quite possible to connect an entire system comprised of 4096 sub-systems on one common 28 wire buss, we have chosen to employ multiple data busses in the structure of our second generation systems. Our purpose was two-fold. By using multiple busses, we are able to achieve faster processing speeds, and we are able to de-centralize the system for increased reliability and ease of trouble shooting.

As was discussed earlier, the 65K programmer defines up to 64 rows on each of 64 columns, or channels. In our final configuration, each row is implemented by an "in/out" card in the programmer, and a data buss per row. All level controlling sub-systems (Row No. 1) then, are on a common data buss, while echo sub-systems are on a second data buss and so forth. By processing all rows simultaneously, in parallel fashion, we are able to fully instruct the entire system once every 3.2 milliseconds.

In short, the data buss approach to console data distribution offers increased system flexability and markedly reduced cost, since control wiring is reduced to a small number of wires and connectors, which are bussed, rather than discreetly wired. System reliability can be increased owing to the vastly reduced number of wires and connectors employed.

REQUIREMENTS OF THE CONSOLE CONTROLS

In order to configure a truly usable second generation system, we felt the necessity to first define what should be expected of the console, from an operational standpoint. The following is a list of what we feel the requirements are:

1. All programmable parameters shall be individually accessible, in terms of functional control and in terms of mode (READ/WRITE).

2. The status of all parameters shall be visually monitor-able, whether in READ or WRITE mode.

3. The operation of all controls shall be instinctive, with no requirement for the operator to speak "computer language."

4. All controls shall be capable of being both electrically and *physically* operated by both the automation system and by the operator. This requirement is not met by null lights and metering devices, which simply instruct the operator to physically move the controls. The automation must actually make the control settings.

4A. In setting system controls, the automation must not employ servo motors, relays or other mechanical devices.

5. All parameters shall be capable of being controlled either one at a time, or on a master basis, as an adjunct to decreasing set-up time of the console. Examples of this would be "clear all mutes," or "assign all channels to Stereo Center," or "set all Equalizers to Flat Response."

5A. All automation modes must be controllable either individually or on a master baiss. E.G. "Master Write"

6. All controls shall be large enough for simple human intervention and shall be easily reached and identifiable by the operator.

7. The console face shall be small enough to allow the operator full control of as many as 64 channels or groups, without having to stretch or move from his operational seating position.

While these requirements might appear to be impossible to achieve, it is our conviction that each one of them is urgently necessary for the successful structuring of complex programmable consoles.

Now, let's analyze these requirements and see what the solutions might be. If we are to meet requirement No. 1, we must add READ/WRITE controls to *each parameter* which is programmable. No.5A rules out simple two position mechanical switches, since they cannot be conveniently operated from an over-riding force. No. 2 requires that we add some form of visual display to each parameter so that we may know its position when it is under automatic control. No. 4 wipes out most controls found on a conventional console, since it dis-allows any control whose electrical setting bears a direct relationship to its mechanical position. No. 5 only further enforces this implication. Since requirements No.1 through No.5 indicate that we must add controls to the conventional console, while No.6 and No.7 indicate that we must remove controls, the answer is very simple and undeniable: It is impossible to successfully automate a complex console without employing radical conceptual changes in its control.

THE ANTI-REDUNDANT CONSOLE

This problem of too many eggs for the basket to hold them, is by no means a new one. It has been solved time after time in the past, and those who have been unwilling to accept the transition have died. How many of you remember those old rotary calculators with about 15 rows of 15 keys, so that you could multiply big numbers like a million, at a cost of around $1000.00. One day some person figured out that you could multiply numbers like 10 to the 99th power, using a simple 10-key keyboard and a few other buttons, for $29.95. Needless to say, they don't make rotary calculators with a key per number anymore.

The same solution is painfully evident in the interface between man and his recording console. The proposition of a control per parameter is preposterous, both in terms of suitability and cost. Since man has but two hands, how can we sanely justify thousands of controls, too small to hold in the fingers and spread out over an area twice the reach of a man's arms?

Although a complete mixdown console could, and perhaps some day will, consist of a single keyboard coupled to c.r.t. displays, it is our contention that the industry is not prepared to go that far into the concept, this soon. (Requirement No. 3)

We feel that our starting point should be a physical position for each track. A position that the operator may identify with, and upon which he may pencil in the instrument, or other music component, which it concerns. Now, if this position includes, among other controls, a momentary push button, this single "button per track" may be used to assign a nearly limitless number of parameter changes to that track, via central control devices, which may be shared by all channels. This premise is particularly effective when applied to parameters which normally require large physical areas on the channel module and large numbers of controls, yet are parameters which are not adjusted very often in the course of a mixdown.

The first such parameter which comes to mind is the output assignment section. If a centrally located matrix of buttons, numbered 1 through 32 is employed, the operator may make his output assignment by the simple expedient of pressing the singular button on the desired track, and manipulating the central matrix for the desired configuration. The operation is totally instinctive, since the operator is allowed to think "assign overhead drums to outputs 7 and 11." If full time visual monitoring of the assignment is desired, a matrix of LED's or other indicators may be located on the channel strip, to indicate the assignment status.

By adding a few more buttons to the central matrix, we can easily satisfy all of the previously stated requirements. Read and Write buttons allow us to selectively place any channel in either mode. A "Clear output Assign" button allows us to clear all 32 outputs with one operation, while an "all channels" button allows all channels of the console to be ganged, or dealt with simultaneously. The singular operation of pressing both the "CLEAR OUTPUT ASSIGN" button and the "ALL CHANNELS" button can clear the entire console, a task which could take as many as 1024 operations on a conventional console. Care must be taken in the design of the central control, however, to make catastrophic changes of this sort well protected from accidental execution. This is a simple engineering job with many solutions.

The savings which can be directly attributed to this configuration of output assignments alone add up to approximately 4 square feet of console space, and the elimination of around 1000 switches, with an inherent increase in system reliability due to the grossly smaller numbers of parts.

From a monetary standpoint, the savings effected by the removal of mechanical parts will more than likely pay for the automation added.

In structuring our second generation system, we have chosen this central control concept as our method of achieving the requirements which we initially outlined above. While not all parameters (Faders in particular) lend themselves to the implementation of this technique, most of them do. While we, at Allison Research, are currently refining methods of dealing with all parameters of a fully programmable system, we will limit the remainder of this article to the two areas in which we have actually reduced theory to production line equipment. These areas are: The equalizer section, and the level controlling section (which includes muting, soloing, input assignment and group assignment).

THE GREAT EQUALIZER

In configuring a fully programmable equalizer, we first had to make the decision of discreet or variable control. We chose discreet selection of frequencies and gain parameters as being the only justifiably correct method of control, since the digital art is one which specifically deals with discreet steps. Even when seemingly variable parameters are controlled by digital means, they are in reality a series of discreet steps. Generally speaking, the Great Equalizer is an addressable device which is controlled by 32 bits of digital data, and is TTL or CMOS compatible. Its programmable parameters are as follows:

1. An 8 frequency 18dB/octave High Cutoff filter (1.2kHz to 12kHz)
2. An 7 frequency Hi Eq. section with peak/shelf selection and ±15dB of Equalization in 15 steps (820Hz to 12kHz)
3. An 8 frequency mid Eq section with shelf/Peak/shelf selection and ±15dB of Equalization in 15 steps (220Hz to 3.3kHz)
4. An 8 frequency Lo EQ Section with Peak/shelf selection and ±15dB of Equalization in 15 steps (39Hz to 560Hz)
5. An 8 frequency 18dB/octave Lo Cutoff filter (39Hz to 390Hz)
6. A phase reverse switch, and an in/out switch.

In attempting to interface this rather complex equalizer to the operator's hands, we find that our own requirements call for visual displays of all parameters as well as positionless controls. The "instinctive operation" requirement pretty much disallows the use of a calculator keyboard.

What we're left with is a matrix of illuminated momentary buttons, one for each position of each equalizer section. The Hi Cutoff section, for instance, can be implemented with 8 buttons, one per frequency.

The equalizer fortunately falls on our list of devices which may be centrally controlled. Here is how we handle it.

The central Equalizer control panel contains the above defined matrix of momentary switches, together with LED indicators. It additionally contains a READ button, a WRITE button, a CLEAR E.Q. button and a HOLD DATA button.

Pressing the singular momentary button on an individual channel module causes the equalization presently in that channel, as well as its READ/WRITE status, to appear on the LED indicators of the central control. Since the central control has large buttons and is graphically representative, the act of "looking at an equalizer" is, in our opinion, more accurate and probably faster than looking at a conventional overly dense and graphically inferior console equalizer. If it is now desirable to change the selected equalizer, the operator need only to continue pressing the singular channel button while he makes the desired changes on the central control. From an operational standpoint, the central control, indeed, actually becomes the channel equalizer.

When it becomes desirable to gang-control the equalizers, for instance, when clearing the console, the Hold Data button comes into play. Whenever the HOLD DATA button is held down, the central control is prevented from receiving E.Q. data from the individual channels, but in turn it transfers its data to any channel whose singular button is pressed. Therefore, the act of clearing the central equalizer control to flat response, then simultaneously pressing the Hold Data button and the ALL CHANNELS button causes all channel equalizers to be cleared to this flat position, or any other desired position.

Duplicate equalization to a number of channels may be accomplished similarly by setting the desired E.Q. on the central control, holding down the HOLD DATA button, then pressing the desired singular channel buttons.

As with the output assignment section, we have again taken hundreds of controls off the console face, as well as 3 to 4 square feet of area. We have also increased the reliability through the removal of mechanical parts and the money saved in the process has paid for the automation.

The system is inherently goof proof, since the unitentional operation of any single button will cause no change in the system parameters. Only an instinctive and intentional two handed operation can enter changes. Since all buttons are momentary, it is impossible to forget that some switch which might affect the data to be entered, was set to some unexpected position.

THE FABULOUS FADER

As you might have guessed, our requirement list necessitates replacing the normal console fader with something quite different. In our quest for a device which meets all of the requirements, yet maintains the approximate feel, throw and instinctive operation of a conventional fader, we finally settled on an approach which utilizes a continuous optically encoded belt as the human adjustment mechanism. Positional indication is provided by a 32 element LED array which is located beneath the belt. From the operator's standpoint, the device appears as a 3/4" x 7" panel, which contains a single LED illuminated momentary button, and a slightly recessed slot, measuring some 5/8" by 5". Appearing in this slot, is a single point of light, which indicates the present position of the fader. The light is passing through the optical belt, which is held flat within the slot, yet is moveable by the operator's finger. The surface of the belt is somewhat textured to present a non-slip surface to the operator, and the resulting feel, in terms of sliding force, vertical pressure and horizontal scaling are essentially identical to that of a very high quality conventional fader. The device is not touch sensitive, it is movement sensitive, and the operator need not put his finger at any particular location on the belt to operate it.

If, for instance, a finger is placed, say, an inch below the point of light and is moved up and down, the light point will track the finger movement, maintaining

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To Our Friends in the Music Business

As some of you are aware, Tin Pan Valley Corporation is dedicated to the establishment of the worlds first community specifically designed for people, like ourselves, who are actively engaged in the creative arts.

Our purpose in undertaking this task stems from our conviction that creative people, particularly music people, desire an alternative to the lifestyle and business climate offered from conventional sources.

Tin Pan Valley Corporation was formed some fifteen months ago, by active members of the music industry, to pursue these objectives. We have since located and purchased the site, a picturesque 234 acre valley, located at the outskirts of metropolitan Nashville and geographically isolated from the path of progress by high rolling hills and dense forests.

Our master plan calls for a maximum of 28 business sites and 25 homesites which are to be privately owned and legally deeded to their owners. There is ample acreage nearby for the future expansion of residential areas. The business sites are restricted to the general realm of creative arts, and are tightly controlled in terms of aesthetics and the enhancement of the natural beauty of the land.

A typical recording studio site will have 2 to 3 acres of land and will be connected to other sites by both natural walkways and by low density roadways. Eating facilities are planned, as are certain recreational facilities.

The focal point of the community is a 3½ acre community owned park, complete with a giant old tobacco barn which is much to beautiful to be torn down.

The general theme of the project is a low key, walk to work, somewhat old fashioned village affair which bustles with creative activity. A true working community whose inhabitants are proud of their environment and of their chosen line of work.

Since our project is one which we feel cannot be entrusted to conventional land developers and money men, we have spent the past 15 months learning the ropes of the real estate game. We have also spent substantial amount of our own money in purchasing the land and in engineering our master plan. Tin Pan Valley now has the support of the necessary political figures and we have gained the knowledge of what is required to make this most unusual endeavor a reality. This is where you come in.

If you believe in what we're doing, you can do us an incredible service by putting your feelings in writing and mailing them to us. If you are interested in joining us, we need to hear from you now! If you're planning a studio, a publishing house or a pottery shop for that matter, you owe it to yourself to find out more about what we're doing.

Whatever your interest might be, I can't emphasize strongly enough the importance of your relaying it to us without delay, as it could have a profound impact on the future of Tin Pan Valley. We will, by the way, be at the L.A. A.E.S. convention.

Thank you.

Paul Buff
Allison Buff
Bob Todrank

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Tin Pan Valley

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... continued from page 35

AUTOMATED PROCESSES

around malfunctioning components. Fault isolation must be simple and rapid. Electromechanical devices, such as servo drive mechanisms, which often have reliability and maintenance problems, we felt, should be avoided. A major advantage of the servo approach is the case of transfer from automatic to manual control during a mix. However, it is extremely difficult to smoothly return the system to automatic control since previous levels are not displayed by the control knobs until after the transition has taken place. Servo mechanisms can also have the disadvantage of unnecessarily limiting the response time of the system.

4. All normal mixing functions should be programmable; not just level controls but panners and equalizers as well. In this way time consuming control position logging is virtually eliminated, and dynamic effects previously not considered practical can be easily achieved.

5. The memory system (programmer) must provide unlimited capability to update or modify control settings with no accumulated time delay on successive passes. The system must permit virtually any number of controls to be actuated simultaneously without suffering the delays associated with non-synchronous priority logic techniques.

6. The Programmer must have virtually unlimited data storage capacity and must be capable of controlling all console functions regardless of the size or complexity of the system. It must be capable of permanently storing a number of complete mixes, but must not utilize more than one track of the multi-track master, and must be capable of operating from an external timing reference if NO track is available on the master tape.

7. The system must have the capability of controlling the master tape machine, and by means of the one timing track assigned to automation, to search out and cycle through any desired location on the tape.

8. The entire system and all of its major component parts must be compatible on a retro-fit basis with equipment currently in the field. For example: considerable functional improvement can be achieved by substituting the new synchronous Automix Programmer for the older non-synchronous unit.

9. The system must be cost effective. It must not only be designed to minimize initial cost, in addition it must be inexpensive to maintain, and capable of sustained operation with a minimum of costly down-time.

Before proceeding with a description of the Automix system, a review of automation technology seems in order.

The remix automation systems currently in use utilize Voltage Controlled Amplifiers (VCA's) in each audio channel, which respond to DC Control Voltage levels generated either by the appropriate console control or by the memory programmer. An individual wire connection is provided for each control function as illustrated in the Model 3224 Control Data Flow Diagram, Figure (1). The VCA is capable of 120 dB control range, 120 dB dynamic range, 0.1% harmonic distortion at peak signal levels, and tracking accuracy of ½ dB in standard production units. Since the VCA is also capable of changing attenuation at the rate of 5 dB per microsecond, signal limiting capability is incorporated into each channel without adding components to the audio signal path. As shown, audio signals pass through only the VCA's while all control functions are processed on a DC analog basis. The non-synchronous, continually scanning programmer is actually a multiplexer which sequentially digitizes each DC analog control voltage. This quantized data modulates an audio tone which is recorded on one track of the multi-track audio master tape. In order to retrieve the control signals, the tape is played back in "sync" mode and de-multiplexed by the programmer as new control data is recorded on a second audio track. This procedure of alternating between the two tracks on the master tape is repeated on each successive updating pass until the final mix is achieved.

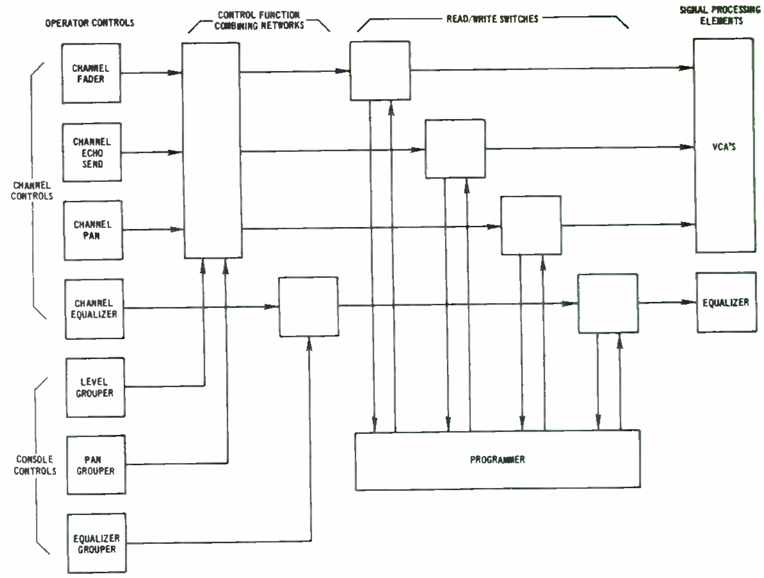


Figure – 1

This Automation system provides completely independent channel electronics so that manual operation of the console is possible without the programmer. Thus, all console faults except those associated with the power supplies and the programmer itself, can be isolated to a single channel.

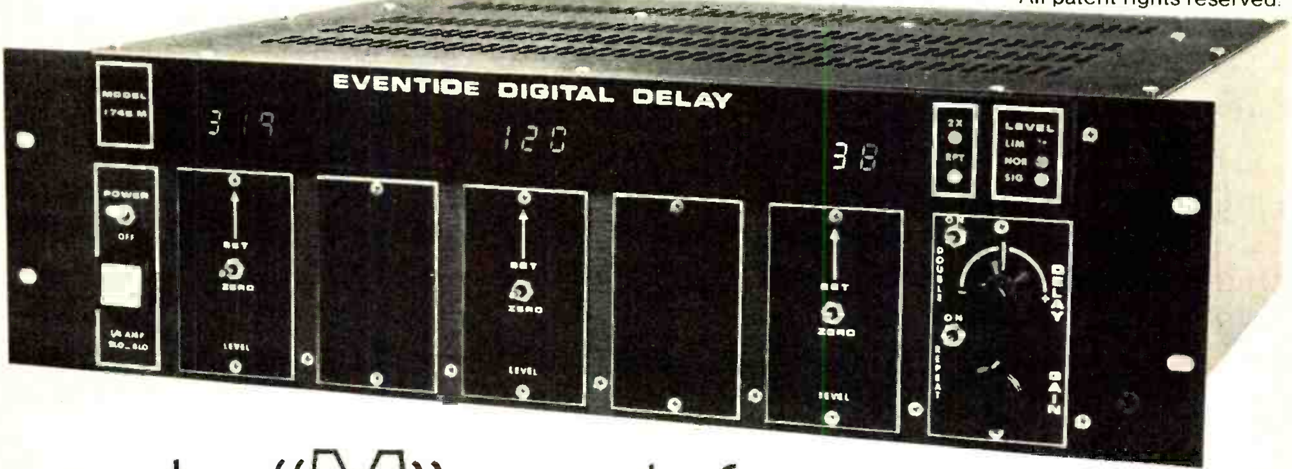
In the development of the Automix system, a number of alternative techniques were investigated. Foremost among these was bus oriented data transfer as utilized in modern digital computers. In the implementation of this technique, all data generating devices, storage devices, and receiving devices, are connected in parallel by a single network called a digital data bus. A group of identification digits or address is automatically added to each data word before it is transmitted on the data bus. Since each data receiver recognizes only appropriately addressed data, any transmitter may communicate with any receiver over a single data bus network.

In order to utilize the data bus technique most effectively, a mixing console would be designed with all controls equipped with, or functioning as analog to digital converters, sending data out onto the bus. The actual audio signal processing components as well as the programmer would receive control data in digital form, since all of these devices would be tied together by means of the data bus. Thus, computational and switching functions traditionally accomplished in each audio channel would be performed by a central digital processor programmed to arithmetically process the control data for the entire console.

As shown in Figure (2), this approach can greatly reduce the number of wires required to interconnect the system.

Although the simplification made possible by data bus approach appears to have a potential economic advantage, detailed system analysis continues to weigh heavily in favor of an improved version of the previously described analog system. For example, the digital system suffers from a lack of redundancy since

Eventide's NEW 1745M



the "M" stands for memory

EVENTIDE has developed a new idea in digital delay lines. Instead of shift registers, EVENTIDE'S MODEL 1745M employs Random Access Memory Integrated Circuits for delayed signal storage. This Memory system makes available a large number of delay taps and produces the audible illusion of continuously variable delay. And because there is little or no noise when changing settings, delay switching without transients can even be made during soft passages.

In one lightweight unit, EVENTIDE has incorporated a sophisticated digital delay line with several special effects, previously unavailable on standard delay equipment. The EVENTIDE MODEL 1745M provides:

- *flanging*
- *pitch changing*
- *recirculation*
- *tunneling*
- *reverb*
- *comb filtering*

And, if that's not enough, plug-in modules optionally available include a wide range pitch-changer/tempo regulator with drive electronics for Ampex, MCI and 60HZ machines, and an automation-compatible remote control.

The 1745M also features improved dynamic range (greater than 90 db) and more versatile signal level indication. Through the use of CMOS circuitry, the system's power consumption has been reduced drastically, which in turn reduces ventilation requirements.

Request a demonstration of EVENTIDE'S NEW MEMORY DIGITAL DELAY LINE from your EVENTIDE dealer.

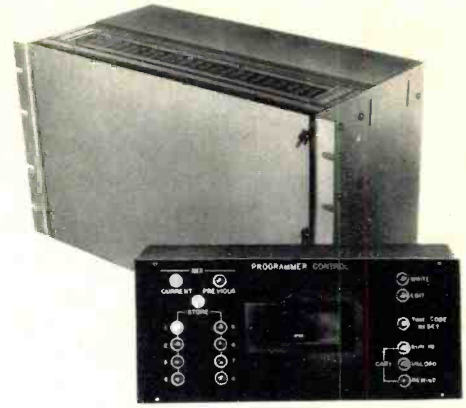
Eventide
the next step

We also manufacture OMNIPRESSORS and FLANGERS.

EVENTIDE CLOCKWORKS, INC. • 265 WEST 54TH STREET • NEW YORK, N.Y. 10019 • 212-581-9290 the real time control and computation signals are centralized within the data bus and a common data processor. This is in contrast with the analog approach in which each channel operates independently and does not require programmer connections for manual operation. Not only does this lack of redundancy increase the liklihood of total system failure in the event of a single component malfunction, it also makes trouble shooting and fault isolation by patching more difficult to accomplish. In addition, considerable difficulty can be anticipated in shielding the low level audio signal processing circuitry from the radiation generated by the high speed digital signal lines required throughout the system in the data bus approach. The relatively low frequency DC control lines in the analog approach present no similar problem. Of course, the all digital system also suffers from a lack of compatibility with existing automation equipment.

As a result of this analysis it was decided to design the new Automix remix console as a wired analog system coupled to an independent synchronous programmer of advanced design. The Automix Control Data Flow Diagram is shown in Figure (3). The console features full time programmable control of all remix functions including channel equalization, program and echo levels, signal routing, and panning of the echo busses in addition to the main quad busses.

Although developed concurrently with the Automix console, the Model 1024 Programmer is a totally independent and self-contained unit, and as such is not subject to the severe constraints imposed by the use of an audio tape recorder as a digital storage device. Utilizing state of the art microprocessor techniques, the



programmer is, in effect, a digital computer including the necessary interface electronics to communicate with the control elements within the console, and also containing sufficient memory capacity to store and retrieve all mixing control data.

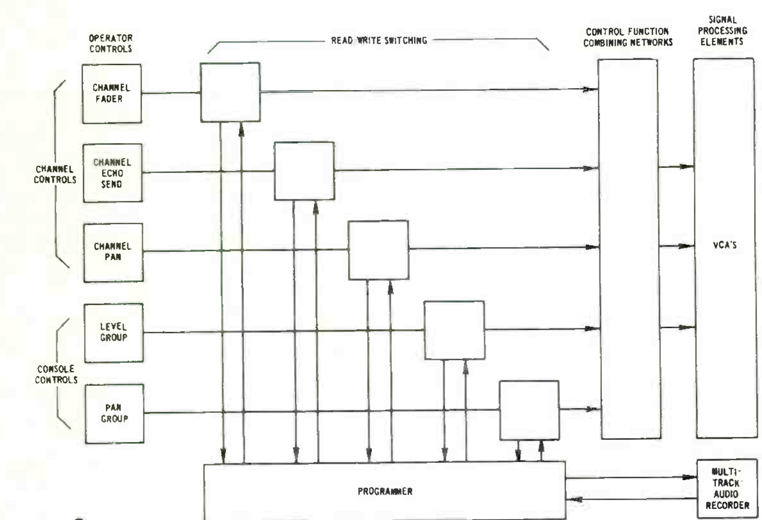


Figure – 2

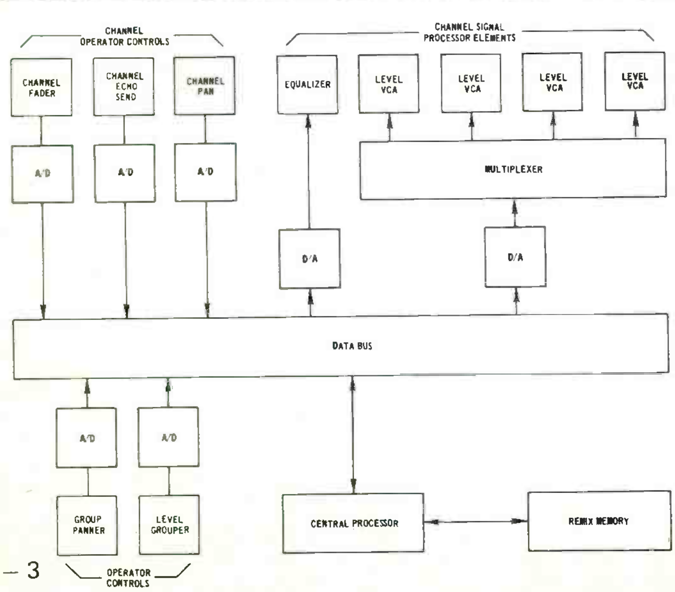


Figure – 3

In the practical utilization of the previously designed non-synchronous remix memory programmers, the high frequency data signal when recorded on the audio master tape, invariably generates undesirable crosstalk. Furthermore, a time delay occurs when data is played back from the tape and processed along with new data to be recorded on a second track. In a practical application, this delay will be several hundred milliseconds after only a few passes, making timing to a beat totally impractical. Additional delays are frequently encountered due to the inability of the system to retrieve valid data if the recorder is operating under less than optimum conditions with respect to machine alignment and tape quality. In an effort to reduce the data rate while adding additional programmable console functions, a modification of the non-synchronous system has been proposed in whch priority is given in the scanning sequence to any function which is changed. While this technique substantially reduces the accumulated delay, it does not eliminate it, and of course, several functions changing at once further aggravate the problem. In addition, the dropouts and errors, which are inevitable when using an audio recorder for digital data, can add substantial delays due to the much slower total scan time associated with this technique. This additional delay results from the fact that data is read in real time, so in the event of a detected error, old data is retained until new valid data is available.

In order to overcome these difficulties, the new Automix Programmer was designed as a synchronous system with random access memory. In this system one track of the master tape is set aside for a timing code, to which all occurances of events are referenced. During the initial "write" pass of the mixdown process, blocks of data are written into slots of the memory. Each slot is refer-

enced to the timing data on the master tape. On subsequent passes, data is retrieved from the appropriate slots, and reconstructed BEFORE it is required by the system. This data is then dumped into the output registers at exactly the right time. Any new changes will be rewritten into the same slot. At any point in time, the exact status of all functions can be determined by first reading the initial setting of these functions, and then substituting subsequent changes. In order to avoid long processing times, cyclic updates are interspersed with the motional data, and momentary status is reconstructed from the last cyclic update and subsequent changes. If any errors are detected during the read cycle, the data in the affected slot will be re-read. If, after a few times of reading the same memory slot the error persists, the data can be approximated by reading and processing other slots and adjacent cyclic updates.

Of all available memory devices with removable storage medium and low cost, the 3M type DC300A data cartridge, which has gained wide acceptance in the data processing industry, was selected. At first glance, a floppy disk drive seems more suited to this application, because it closely resembles a true random access memory. However, a single floopy disk has inadequate data capacity to handle all console control functions, while the data cartridge, with approximately ten times the storage capacity, is more than adequate for all anticipated applications. The longer access time associated with the data cartridge is accommodated by the use of "virtual" storage, whereby all data that is anticipated to be required for the next seven seconds is stored in a semiconductor cache memory. In the event of a detected error, this time is available to the programmer to conduct an error correction proceedure, and the cache memory that normally stores the anticipated read data, will now temporarily store new write data. Cyclic update information is interspersed with the motional data, and is repeated in its entirety over a period of 13½ seconds. Because of the serial nature of the mixdown process, immediate access to all data is not required, and the seven seconds available to the programmer will allow it to access points well beyond one minute removed from the location of the master tape machine.

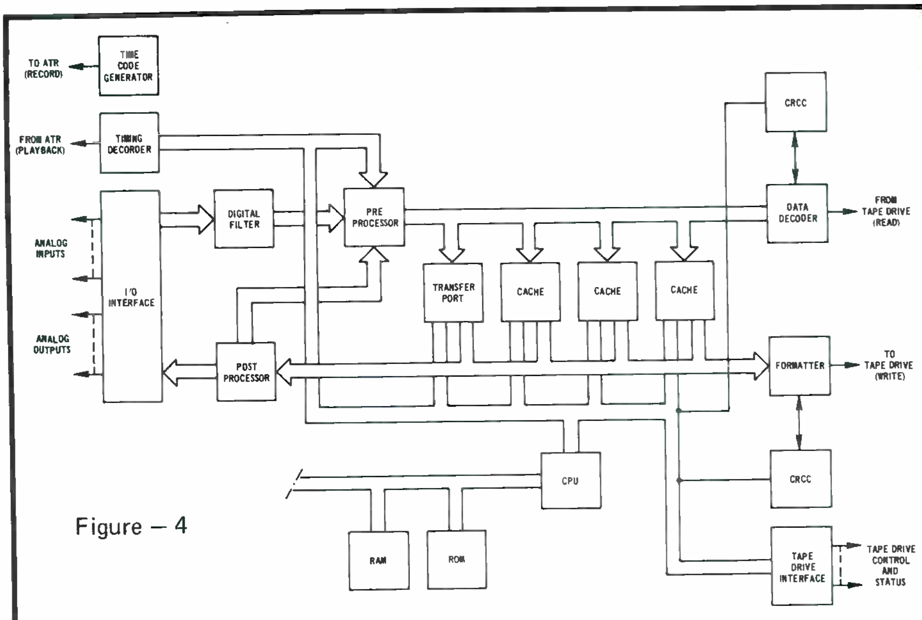


Figure — 4

Figure (4) shows a block diagram of the basic programmer system. The self-contained Code Generator produces the standard MagLink timing code which has found wide acceptance in the synchronization of audio tape machines.

The I/O Interface receives and digitizes analog control levels and also accepts blocks of digital data from the processor for conversion and transmission to the console. The Model 1024 Programmer will accept up to 1024 analog inputs, but is expandable beyond this number.

Input data, in its digital form, is processed by a Digital Filter. This filter compares the magnitude of each sample to the magnitude of the following eight samples of the same function, and decides on the validity of each sample.

The Pre-Processor retrieves valid motional data from the filter and adds timing, addressing and additional status data. When the tape drive, which is under the control and supervision of the Central Processor Unit (C.P.U.) is ready to accept "write" data, the pre-processor will send its data to the Formatter, which adds the proper pre-amble, post-amble, and Cyclic Redundancy Check Characters (CRCC). The data is recorded in blocks of 16,384 bits, not including pre-amble, CRCC and post-amble, but including status information, timing reference data and housekeeping data. Each block of data corresponds with approx. 1.7 sec. of real time. Data is recorded on the cartridge in accordance with the proposed ANSI standard @ 1600 bits per inch, phase encoded, using a standard 3M type DC300A data cartridge and a four track tape drive. A cyclic redundancy check is performed on read-after-write data. If a write error occurs, the system will record the data on the next available block of tape.

When reading data from the cartridge, the Data Decoder will convert the tape output signal into a clock and a decoded data stream. Because the system reads data long before it is needed, each block is temporarily stored in one of three cache memories. A cyclic redundancy check is performed by the Data Decoder and when an error is detected, the programmer will conduct an error correction routine. Meanwhile, operational data for the system will be retrieved from the cache memories, and new data, generated by the pre-processor, will be stored in available caches until the tape drive is available to accept more write data.

Among the operational advantages of this system are the following:

1. Only one track of the multi-track master tape is required and the timing data on this track has a lower frequency content than required for either SMPTE time code or for actual data recording, and is therefore less likely to present crosstalk problems. Crosstalk can be further reduced by recording the timing track prior to recording audio on the adjacent tracks.

Timing data may also be derived from a deck mounted tape timer, the tape machine tachometer or any other time related reference, thereby eliminating the need for a spare track altogether. This feature is in the processing of previously recorded masters, which do not have a track available.

2. To eliminate wear of the master tape, a work print may be made by dubbing the original onto a second machine along with the timing code. The work print can then be used to generate the remix data, and when the process is completed, the original master is used to generate the final mix.

3. Errors caused by the audio tape recorder will have little or no effect on the operation of the system, since only timing data is involved, and the timing decoder of the programmer has the ability to free-wheel when invalid data is received.

4. Precise timing may be achieved by entering data statically with the audio tape stationary.

5. Although the system will automatically update the previous pass, as many as eight complete mixes can be stored simultaneously and recalled in part or in their entirety at will. In this way, a complete mix can be assembled from several previously stored partial mixes, or the best parts of several mixes can be combined into a single final mix at the touch of a button.

6. The use of the MagLink code permits the system to control the audio tape machine in order to automatically search and cycle through any desired portion of the master tape.

As a total system, the Automix console and programmer have achieved all of the design goals previously outlined, while maintaining the concepts and components which have achieved time proven performance and reliability.

Circle No. 131

. . . continued from page 35

RUPERT NEVE

a second stage, the engineer will progress from a trial static balance, introducing such dynamic changes as the music and the producer's aim demands. These moves are, of course, restricted by the simple fact that he has only two hands and he must therefore rehearse so that at the final stage a master copy can be taken in which all the necessary actions are perfectly executed.

A number of points emerge from this brief description. Firstly, the engineer has to learn the music, and this is one task which cannot be delegated to any machine, even if a number of expedients may be adopted to make the task easier.

Secondly, the engineer must practice and remember his own actions during the mix, and this is the aspect upon which all existing systems have concentrated.

Thirdly, in the process of rehearsing or perfecting the mix, much winding and rewinding of the tape is involved. Of course, many excellent tape location systems are available, but ideally a comprehensive mixdown aid should store start and finish locations along with the other mix data, so that "Back to the top" will involve only one command without the engineer having to remember a time index or footage reading.

Essential Criteria:

A more detailed study will reveal other criteria for a truly helpful mixdown aid.

For instance, it is not enough simply to store actions during an attempted mix, it must be possible to correct and improve during recall of those actions, wihtout having to employ additional indicators or controls. That is 'Up-date' must be instinctive. Again, recall of all stored attempts must be possible not only the last but one. After all, not all engineers progressively refine a mix, some will wish to make a series of alternative trials, returning to any one at a later stage.

It has already been emphasised that only the engineer can remember the music. But a well designed system will give him flexibility in carrying out the mix and thus assist in even this task.

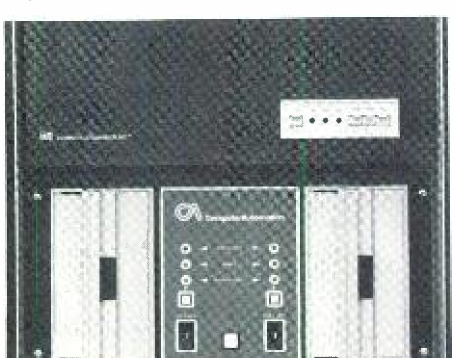


FIGURE 1

For instance, it should be possible to break the operation into short time segments, so that the amount to be remembered at one time is reduced. Alternatively, some engineers may prefer to concentrate upon one section of the band at a time, both to reduce the memory load and to secure detailed concentration on that section. In either case, some means will be necessary whereby the various part attempts may be 'Merged' together into a whole, with special provision to ensure that the 'joins' come right in the process.

Often an engineer is faced with a series of very rapid cues, as when sorting out a series of vocal overdubs for instance. If the aid system can confer some freedom from 'real time' such passages could be handled much more easily.

An aid system which incorporates the features listed above, and which is simple to operate and understand will confer upon the operator the ability to 'manage' the mix process in accordance with his own ideals and thus truly free his creative ability.

NECAM – A system designed to meet the criteria just described:

At the heart of NECAM, though decently hidden away in any practical installation, is a powerful mini computer (fig. 1). This undertakes all decisions and superintends the storage or memory of all controlled functions. One great advantage of the use of a computer is the flexibility that it confers. The sets of instructions or rules by which the computer operates are called software, and software can be changed to enable alternative modes of use without modification to the hardware. This means that many features of NECAM can be tailored to the specific needs of a particular engineer or studio.

Associated with the computer is a data store in which all the settings and actions that take place during a trial mix are 'remembered'. This arrangement means that data is not laid upon tracks of the master tape. To interface between console and computer, and between tape machine and computer, two small card racks are necessary. To complete the installation, there is a small control unit complete with indicator display (fig. 2) with which the operator communicates with the system, and it with him. The display is a 32 character alphanumeric unit and provides a continuous indication of status, will indicate the next action, and will politely refuse impossible commands.

Using NECAM, the engineer is always aware of something rather more than a mere memory machine.

Time Code:

All stored activity in the system is related to a digital code which must be

FIGURE 2

recorded on one track of the multitrack tape. To facilitate working with existing systems, including VTR and Film, the internationally accepted SMPTE edit code has been adopted. This code identifies any point on the tape in terms of an 8 digit number which reads Hours, Minutes, Seconds and Frames.

The code need only be written once on the tape, and no special equipment is needed for this purpose other than the code generator.

The NECAM code read system incorporates a verification program so that drop outs, splices or lengths of leader can be tolerated. Once the code is on the tape it cannot slip, so that all cues remembered by the system will be reproduced at the correct time irrespective of the number of updates.

Since all cues are related to this code, the system will operate at any speed at which the code can be read. Thus to cope with a series of rapid cues, the tape may be run at half or even quarter speed. When played back normally the stored actions will take place as required.

Labels:

To avoid the use of 8 digit numbers to identify any point on the tape to which the engineer might wish to return, such points are described by Labels. These are simple numbers issued by the computer whenever the operator presses the 'Label' button. The computer 'ties' the label number to the time code found on the tape at that point, and the operator has only to remember a number of at most 3 digits. (999 labels is the maximum permitted.) Labels may be defined with the tape stationary or when in motion, so that musical cues may easily be marked for future reference.

Tape Locate:

The time code, and the labels are used in the tape location function of NECAM. Any label may be returned to simply by entering its number and pressing a 'Go' button. The location mechanism involves a sophisticated application of the computer both to eliminate the need to run the tape in contact with the heads during fast wind, and to account for differences in tape machine performance due to spools either partially filled, or of differ-

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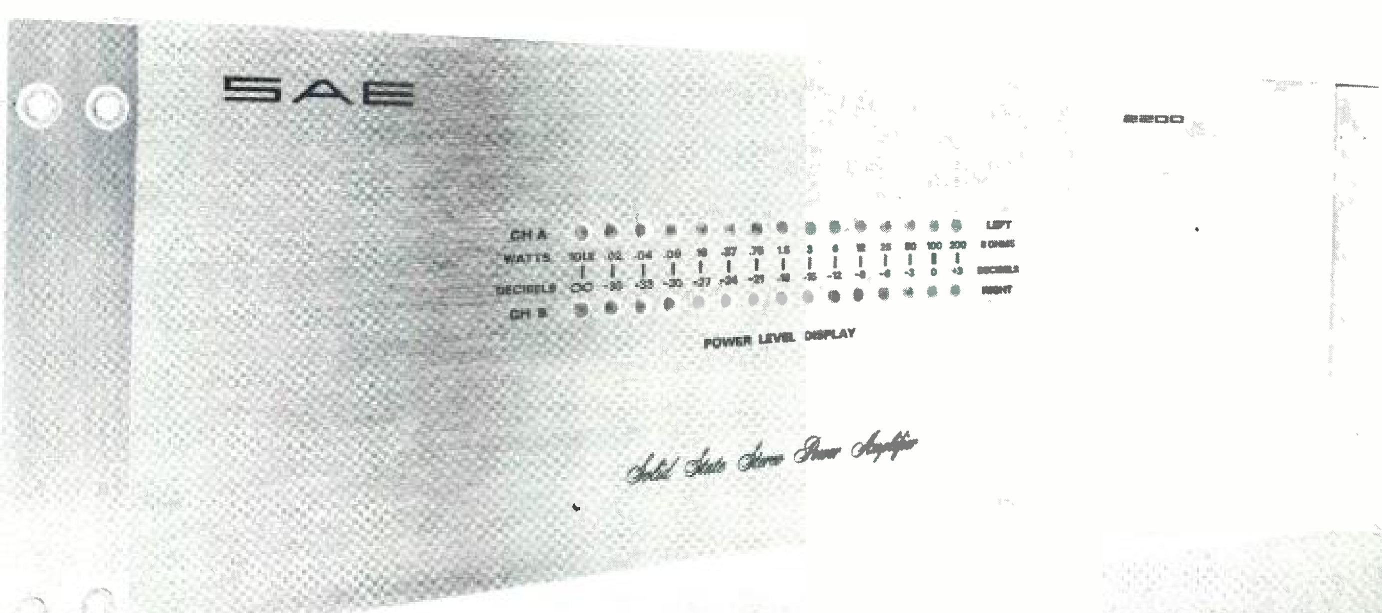

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Circle No. 132

ent diameters. In action, once the locate command is given, the tape machine first verifies the present tape location by a very brief 'play' action, then fast wind starts, with the tape free of the heads, followed after a calculated period by another brief check, then at the minimum distance from the location, the tape is again brought up to the heads and the transport is slowed down smoothly to park the tape a second or so ahead of the chosen location.

The programme is 'adaptive' that is to say, the computer literally 'learns' the characteristics of the machine and progressively improves its performance over a number of runs. Thus widely different transports may be used without the need for adjustments.

Mix Segments:

To facilitate 'management of the mix', the system is designed to be able to operate in segments, which are of course defined by 'Labels'. The start of a segment by a 'From' label and its end by a 'To' label.

The locate function parks the tape ahead of the 'From' label, then on pressing the 'Play' button, the system is automatically in the "Assisted" mode. There is no distinction between 'read', 'write' and 'up-date' in NECAM, all are combined in one 'Assisted' mode. If a 'To' label has been entered, the tape will stop when it is reached, and at this stage a decision must be made as to whether to remember whatever has been done, or to discard it. This is in contrast to other systems where the decision to 'write' must be taken first. Pressing the button marked 'Keep' will store the start and finish labels of the segment as well as the control movements made, and will allocate a "Take" number.

To replay that segment, with all stored actions, it is only necessary to press in succession the 'Go' and 'Play' buttons.

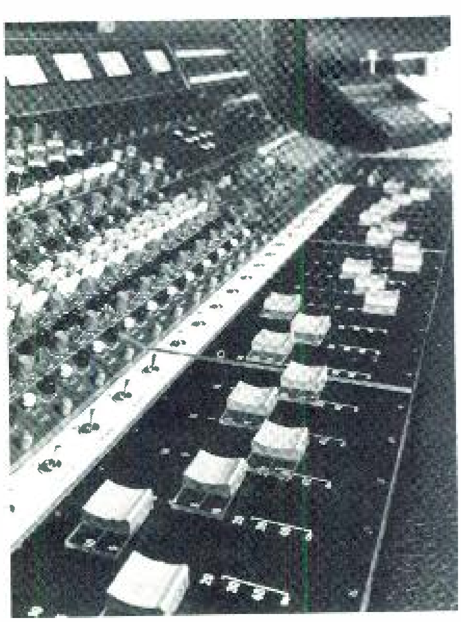


FIGURE 3

When several segments or attempts have been stored, each may be recalled by entering the 'Take' number before pressing 'Go'.

Display:

To keep the operator informed of progress, an alphanumeric display is provided. This shows which take is being played as well as the labels defining its start and finish. At other times, instructions or explanations are offered: for instance, entering a 'Take' number which does not exist will produce the response "Take 'N' has not yet been defined." Again, when undertaking data manipulation, a polite message such as "Merging takes please wait" will appear. Simple terms have been used for all control buttons 'Keep' 'Group' 'Delete' for instance, in a deliberate attempt to avoid 'Computer' type jargon.

Faders: (Fig. 3)

In designing NECAM, much emphasis was placed on the need for 'Instinctive Up-date,' but both common sense and practical economics dictate that the same argument, and the same solutions, should not apply to all controls of a mixing console.

In the NECAM system a radical solution was adopted in the case of faders which are clearly the most important controls on the console. A servo drive system has been incorporated so that on replay of stored actions the faders move, as if by hidden hands, in exact repetition of the operator's actions. The knobs are made touch sensitive, so that to improve or correct, it is only necessary to grasp the fader, which will automatically secure up-date.

This is not such an unusual idea s may first be though, it is indeed a well proven system used, for instance, in the Auto-pilot systems of the thousands of commercial and military aircraft in service to-day. The advantages are as follows: The fader retains its dual role of indicator and control, the operator has no special decisions to make, and the system is fail safe, since the audio control still resides in a fader track which will operate quite normally with the computer switched off.

It may be added that there is a certain magnetism about the sight of faders moving on their own, and this feature may well attract a certain amount of studio custom for its own sake!

For controls less likely to require dynamic up-date, simpler memory systems are to be preferred. A simple logging system enabling the operator to set up 'pre set' type controls to a stored pattern is one method. Alternatively, if VCAs are acceptable, the stored data can easily be used to preset auxiliary controls in an instant. Neither of these methods is truly capable of dynamic up-date, but a moment's reflection will show that to make use of a true dynamic up-date facility on all auxiliary and E.Q. controls implies a mix complexity that can never be economic in either equipment or time terms.

The memory capacity of NECAM will cope with any of the systems, and being designed essentially for custom applications, it will be up to the purchaser to define his needs and to justify the costs involved.

Sub Groups:

Servo controlled faders may be linked to a master fader just as can VCA controls, but in the NECAM system a more elegant sub-grouping method is adopted. A button marked 'Group' is pressed, which produces the message "Please touch required faders." Then by simply touching each fader to be grouped for an instant, and then pressing 'Group' once more, the selected faders will now move together, whichever one is operated.

Should the internal balance of the group require adjustment, touching any two of the group at once will temporarily break the grouping to permit the adjustment.

Coupling within the group is so proportioned that all will fade to silence together, preserving the balance in the process. The only limit on group size or on the number of groups is the number of faders available. Cancellation of a group is very simple; 'Group' is again pressed followed by touching one member only which effectively establishes a group of one.

Up-date Modes:

Three modes are provided for update, which are selected on individual toggle switches associated with each fader.

'Manual' mode disconnects the stored instructions altogether but allows a new set of actions to be remembered. This is of use where total cancellation of a series of actions is required without having to 'hold the fader' throughout the take.

'Normal Mode' accepts any correction made, then on releasing the fader knob, the last position is retained until a further stored instruction arrives. From then on, the original movements will be repeated.

This enables the level during one section of the music to be adjusted without affecting the remainder.

'Relative Mode' on the other hand will effect a proportional correction throughout the take, and is useful when it is desired to correct the level of one instrument relative to the remainder all the way through.

Transition from one mode to another can be made at any stage, thus a premature cue (which would be stored along with the correct actions) can be cancelled by holding Manual mode until after the incorrect cue had passed.

Merge:

The most sophisticated feature of NECAM is the ability, conferred by the use of a computer and of an independent storage medium, to manipulate the stored data and thus perform "Off Line Edits."

Suppose that an operator wishes to divide his mix into time segments – perhaps related to significant instrumental entries or to the choruses of a song. All he needs to do is to define the segments by issuing labels at the appropriate points, then perfect his mix progressively over each such segment. Finally, the stored actions of the best takes may be joined together in the 'Merge' action – without moving the tape at all.

The system has facilities for ensuring that the control positions at the end of one segment may be exactly recalled to form the start of the next so that no discontinuities exist, in other words, the computer 'looks after the joins.'

This is equivalent to making 2 or 4 track copies of each mix segment, then splicing them together. It is not essential to plan this action beforehand; suppose a mix of some length is attempted, which goes wrong when three-quarters completed. The 'Keep' button may be pressed at once, which will stop the tape, issue a label defining the point, and define a take number also.

The tape may then be wound back to any convenient point prior to the mistake, a new label defined and the controls brought into the correct positions automatically. Then a new attempt may be made which hopefully is satisfactory. Afterwards, the two 'good' sections are merged to make the complete take.

A further, more complex, merge is possible. When the 'Merge' command is given – by just entering a 'take' number, then pressing 'Merge,' the display will read "Please touch required faders." This may be ignored, and the button pressed again which will result in all fader movements of that 'Take' being incorporated in the merge. If, however, it is desired to include only some tracks, then the appropriate faders are touched before pressing 'Merge' a second time when only the data relevant to those controls will be merged.

Thus it is possible to lift the control movements for strings from take 1, with those for brass in take 2 and those for drums in take 3 to build a composite take 4.

All of these actions are performed "Off line," that is with the tape stationary and at no stage is any data lost – all the original takes are still in store and may be recalled at will.

This feature enables a balance engineer to 'Manage' his mix activity in a wide variety of ways, entirely at his own discretion.

Storage:

The data store in the NECAM system is a "Floppy Disc" which is a magnetic storage medium based on a flexible, coated plastic disc housed within a protective cover. The system has good "Random access" capability which enables any part of the stored information to be recalled without, for instance, waiting for a tape to wind through. Two discs are used in a dual drive system, one stores moment to moment activity, acting as a 'scratch pad,' the other is the permanent store. It is this dual disc system that enables the 'Keep' decision to be made after a take instead of before, and which enables the more complex operations of 'Merge' to be undertaken.

Once the mixdown is complete, the data disc (which is about the size of a 45 rpm single) may be stored inside the multi track tape box, thus associating the stored information with the music to which it belongs.

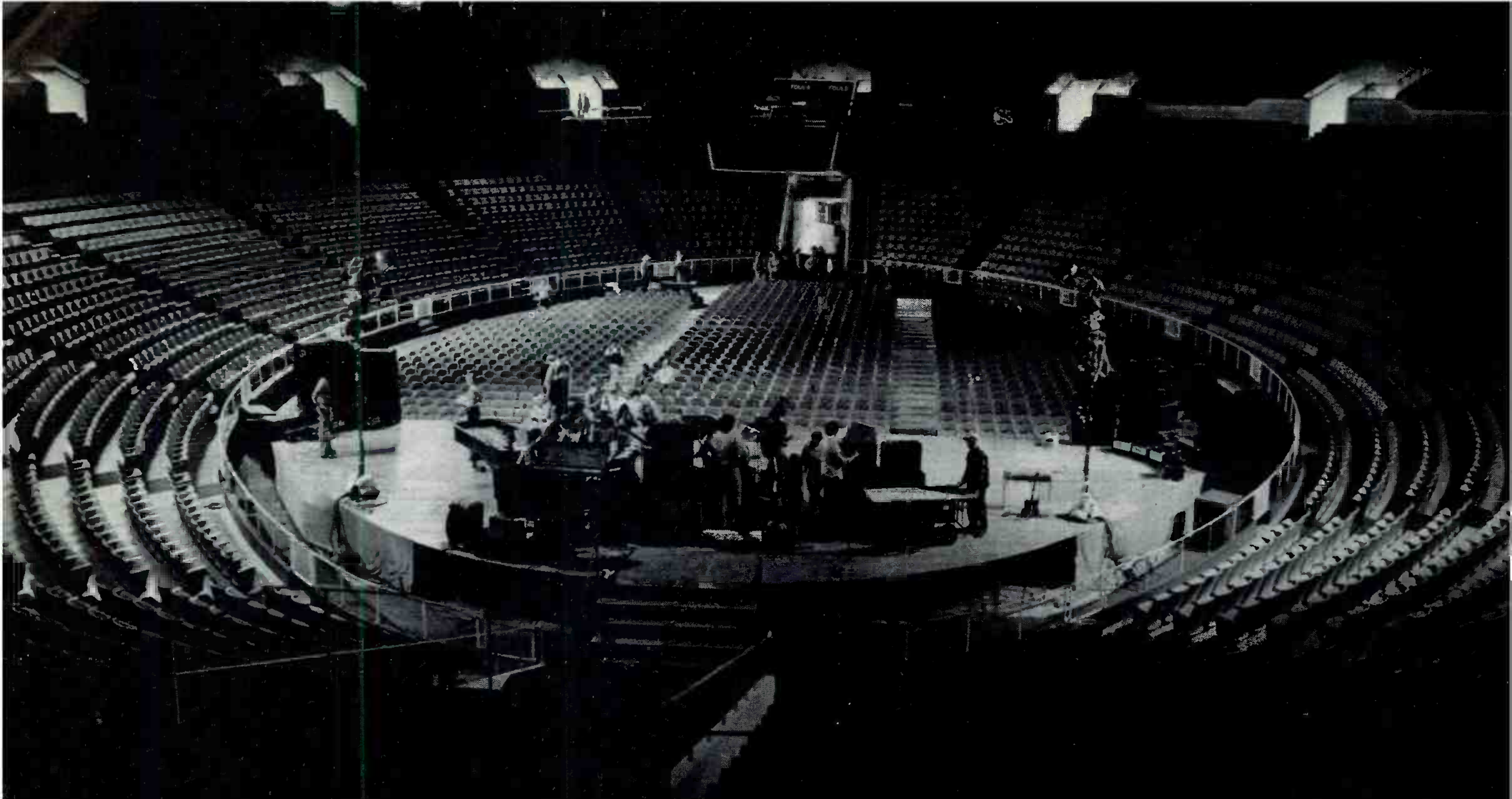
As much care must be taken with the disc as with the master tape, and for exactly the same reasons.

There is ample storage capacity on one disc for the activity likely to be associated with one or indeed several musical numbers. The capacity used up depends on the number of stored attempts and the amount of activity involved in them. As a precaution, the system provides the operator with a check of capacity used up each time the 'Keep' command is issued, and should it prove necessary, unwanted 'Takes' may be deleted when the computer will repack the remaining data so as to provide room for more.

Conclusions:

NECAM has been designed to meet the balance engineer's real needs and to free him from unnecessary decisions whilst leaving him in command of the situation. The use of a computer means that changes to the mode of operation, and extensions of use, may be added later without modifications to the hardware. NECAM has thus a built-in longevity even in this fast changing world.

NECAM hasn't made any 'Hits' of course, but balance engineers using it are certainly going to.



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SOUNDSTREAM
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by ROBERT EASTON

Salt Lake City has long been known as a place that defines realities in its own terms. But that practice has been extended to professional audio, and with empirical results that should attract the interest of every quadrant of the music business. Soundstream, Inc. is the organization in question, and recording/signal processing by purely digital means is their business.

> *Biographical Sketch*
> *"SOUNDSTREAM'S"*
> **Dr. THOMAS G. STOCKHAM**
>
> *Dr. Thomas G. Stockham has been active in the audio field since 1949. He received a Ph.D. in electrical engineering from M.I.T. and taught there for eleven years. While at the Lincoln Laboratories at M.I.T., Dr. Stockham originated techniques using digital signal processing to model auditory perceptual processes. Dr. Stockham came to the Department of Computer Science at the University of Utah in 1968. He has been instrumental in developing one of the country's formost digital signal processing facilities there, and has continued his research in visual and auditory fields. Over two dozen papers carry Dr. Stockham's name, and products such as the Bose Model 2201 loudspeaker system and dynamic noise reduction systems are based on his research. Dr. Stockham is a member of I.E.E.E., A.E.S., A.C.M., and served on the "Watergate Committee" analyzing the "Presidential Tapes." He is currently on leave of absence from the University of Utah, and is the President of Soundstream, Inc.*

Digital audio, as an industry buzzword, has come to mean a variety of things to different people. But in the context of Soundstream's endeavor, digital refers to the possibility of performing all studio operations, from the output of a microphone preamplifier through power amplification, by purely digital recording and signal processing techniques. Soundstream's objective is to realize a set of performance criteria that have long been known to be obtainable from this new approach.

Digital recording of audio can produce a result having a dynamic range at least 30dB greater than analog recorders, and with distortion of vanishing proportions, even at peak levels. Since the signal has been placed in the digital realm for recording purposes, it can be left there for mixing and signal processing operations considerably beyond the capabilities of a practical analog studio system. But perhaps the most persuasive argument for the departure to such a scheme is the ability of a system to be defined, structured, and altered by writing new "software," rather than by discarding equipment and replacing it with newer! better! grander! versions when frustration sets in.

Traditionally, studio operations have been pretty much dictated by a compromise between what producers and musicians want, and what is technically feasible. As analog systems, recording facilities have evolved to a plateau where they are both marvellously extensive and confounding because of it. But the prime feature of today's studios is perhaps their unsuitability to condensation and simplification of control functions. In contrast, the "digital," or programmable, studio system offers an entirely different approach to the architecture. While accomplishing the same tasks (plus new ones), it provides a degree of control and display of information that may be impractical to achieve by analog means.

Translation of audio signals from an analog format to a digital one provides advantages in signal-handling not common to the analog original. Numerical representation is used to describe amplitudes and other characteristics. In binary form these numbers have an almost total noise immunity. The notion of distortion, gain errors, and signal losses has no relevance to the binary medium, leaving us with an extreme degree of "absolutism" in handling the original signal.

Tape recorder "defects" tend to fall into categories that do not affect binary signals; the fact that what comes off the tape is perhaps a distorted version of the input doesn't matter to the encoding schemes used with this system.

Audio signals are converted to binary form as soon as possible in Soundstream's system. An analog-to-digital converter (ADC), of Soundstream's proprietory design having 16-bit resolution performs this task. It accepts line level inputs, and has outputs compatible with computer hardware. The numbers from the ADC represent the instantaneous amplitude of a waveform, sampled at periodic intervals. (Figure 1) Greater amplitudes produce larger numbers, and one measure of a digital systems' fidelity is its ability to produce numbers which are both accurate and very large. 16-bits produces 2^{16}, or 65,536 different descriptions of possible amplitudes; the dynamic range of the conversion is therefore 96dB. Similarly, the noise is one part in 65,536, or 96dB below clipping, and such errors produce a distortion of about .0015%.

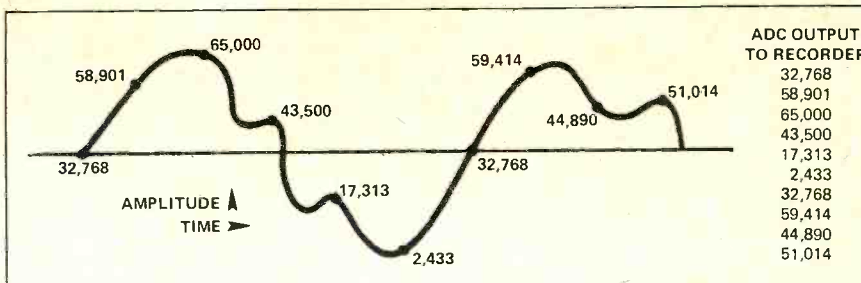


FIGURE 1: Amplitude quantization of an audio waveform by a 16-bit ADC. An opposite operation by a DAC reproduces audio from numbers.

This level of performance can be very closely approached in a well designed digital system, without the use of compressor/expandor noise reduction systems. And although compandors do provide a steady-state improvement of signal-to-noise with conventional recorders, the digital representation can allow undistorted recording of transient information that sneaks past all analog noise reduction devices. Modulation noise, too, is simply not present in digital recording, since the tape medium is used only to record full amplitude pulses ("square waves"), whose purity is entirely unrelated to the fidelity of the result. Tape and transport problems, too, can be minimized. Normal flutter is removed, and error-detection schemes are used to restore data damage caused by tape dropouts. Although the price of tape isn't the largest part of a recording budget, it's nice to know that 24 tracks could very probably be put on 1" tape, travelling at 30 IPS.

The tape recorder isn't the only method of storing digital data. In fact, an 800 megabit disc file is employed for the storage of recordings ranging from opera to the Beach Boys at Soundstream. But they expect that tape will remain the lowest cost recording medium for some time. By a wide margin tape is the densest storage medium possible today. However, for fast access of arbitrary locations of sounds, magnetic disc files and computer mainframe memory can be loaded from a portion of a tape. Editing, repetitive examination of areas, and extensive signal processing schemes are best accomplished through instantaneous access to material. At this time, it takes 100 mSec to access any location within the 16-minute capacity of the disc file. Material can be retrieved from other storage areas virtually without any delay at all.

Soundstream intends to show, and market, recording systems based on their prototypes later on in 1976. But that turns out to be only a part of their current activity. Although qualitative considerations served to focus attention on digital recording in the past, today there are reasons why audio should be left in that format for all subsequent operations. An extreme case of good fortune provides that extensive "mixing" and related signal processing operations involving digitized signals can be performed in large part with off-the-shelf computer hardware, and at a manageable cost. After several decades of independent development, general purpose computers have evolved into the "Minicomputer" – a broad capability machine with high speed, simplified programming requirements, and an ability to grapple with mathematical operations on a large scale. The Minicomputer plays a central role in Soundstream's concept of the future digital and programmable studio system. And at this juncture, the similarities of the digital audio system depart from all analog counterparts.

Let us digress briefly, and play devil's advocate for today's way. We'll briefly (and probably incompletely) enumerate everymans gripe with his up to $50,000 mixing console and its hardwired philosophies.

1. It does only what the original specifications forsaw as the necessary tasks. Restrictions were placed liberally by the buyer for cost reasons, and by the manufacturer out of a desire to construct it readily and profitably.

2. Half the expense of an audio console is associated with modules that appear on, say, all 28 input positions as a matter of convenience. But at any given time, only a small percentage is in use. Furthermore, controls may be many feet from the operator, with their settings completely invisible to him.

3. Technical characteristics of equalizers and other fixed items are not particularly alterable. Obsolescense and rapidly changing requests from the Producer's corner threaten the studio owners position from the day of installation. Yet capital equipment amortization prevails in the main, and the owner plows ahead toward a replacement date three years hence. "Boy, I know just what I'm going to get next time," he says.

4. The best part of being a recording engineer, and the scariest part too, is turning out a sound that is well liked by the client. Now, seasoned mixers know that a hell of a lot of variables enter into that picture; being "good" in that chair is a measure of how well one normally pleases – politically, if not sonically. In truth, the mixing art is based on an ability to operate well, having virtually *no* real information about what is coming up the mike cable, and remarkably little ability to do anything about it with the equipment at hand – even if one *did* know what the real problem was with a sound.

Most of us having spent time working in different studios will not differ greatly on the final conclusion of this technical ring-in-the-nose, and know quite well what the *real* problem is. A variety of "Automation" systems have been devised to alleviate certain studio procedural difficulties, but the approach proposed now embarks on the road of total programmability – in both the audio signal path and its controlling functions.

Dr. Stockham describes Soundstream's approach to the digital studio system by suggesting an intriguingly abstract vision, where all the articles in a studio merge into one ultra-capable entity, that can adapt itself to whatever form is best for the task at hand. This new article is the Minicomputer, its associated peripherals, including the digital audio recorder, and suitable software written to implement needed operations in the studio. In much the same way that an I.B.M. 370 computer is used by one party for galvanizing action from credit-card deadbeats, and by another for writing Reader's Digest Sweepstakes letters, smaller machines can be programmed to perform a very specific 24-track mixdown, or for generating complex corrections for groove-dynamics problems in disc cutting. It's the same machine in either application, but with different software trotted out.

The future attractiveness of digital signal processing of audio rests on the provable assumption that once purchased, the format of a studio system can be altered at will. Specialized signal modifying areas can be created by simply stating what is wanted. This is done by using software. It's the program, or set of instructions, that tell a computer system "who it is." Computer-based systems have no suitability to do any task whatever, except as instructed by software. It is partly supplied by the manufacturer of the system, and partly written by the user for his specific job. In short, it emulates the function of any real or imaginary hardware article. A reasonably brief set of instructions may create a non-existent filter or reverberation chamber, while physically building it might be a task that would never happen.

On first considering the idea, such a system seems pretty far removed from the way things are accomplished today. Yet the case of a professional recording studio is one involving considerable expense directed toward the solution of tasks that are, compared with other computerized business, only moderately complex. The primary objective expressed by the folks at Soundstream is not to do what is already done, but to provide a basic system that will do that and virtually anything else that arises as a consequence of possessing the means to perform arbitrary functions. It is intended that digital systems being developed can be used by virtually anyone, while the technicalities of understanding exactly how it all works will reside largely with this new sort of "console company."

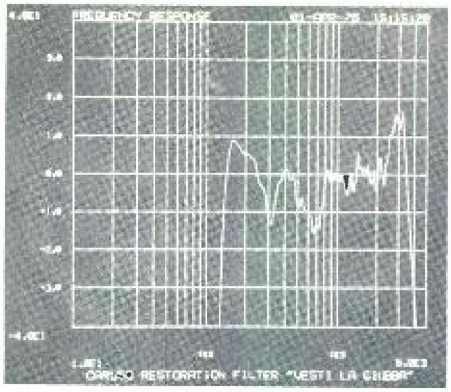
The physical and operational form of a digital audio console is significantly different from convention, though fairly straight-ahead in its basic concept. On a conventional console, knobs and switches "stay put" when adjusted, and thereby "remember" the desired settings. You stand up and look all over the board to observe where you set things previously. Sometimes you get down on your knees and fiddle with the patch bay spaghetti to see where everything is going, too. In the digital counterpart, extensive video graphics show the relative status of useful information. The multiplicity of mechanical controls is dispensed with in favor of a smaller number of controls. The settings of these controls are held in memory and shown on displays.

The instance of an "echo send" function is illustrative, since there are normally many echo send knobs present, most of which are never changed much during the mix. In the case of a digital console, some convenient control can be called "echo send" for awhile, and used to set all echo send levels, which will be shown graphically on a color display and stored in memory. Required changes can be made in rehearsal and stored. They could also be made manually during the mix. The ability to make any change at any time is present to a greater degree than before.The need to have 500 or so controls present has been eliminated. The number of controls actually present on the console is flexible. It ends up being a quantity slightly greater than the number of events one might want to change live and simultaneously.

Such a system is very closely related to its tape (or disc file) recording medium. It embodies many of the concepts devised for present day "automation" accessories, in the sense that changes in any setting are associated with a location in the musical performance. A time code is used to identify specific points along a take. Stored settings (even if continually changing) for control functions are applied to those audio signals. In this way an indefinitely large number of signal-modifying commands may be applied at once to a multitrack situation. At selected times each of the commands may be written, updated, or simply read at will.

Combining a large amount of computing power with graphic displays can overcome one of the primary difficulties in day-to-day studio practice: knowing what the original signal was, and what effect a modifying operation has on it. Figure 2 shows the filter transfer function employed in producing a particular form of equalization used to correct the spectral anomalies caused by an old metal acoustical recording horn. Visual display of its characteristics is revealing to the engineer, as relationships can be perceived between the display and what is

Tom Stockham with some of his digital recording goodies



heard. This example is a set of corrections determined by a computer program designed to locate unnatural resonances. computer. Restoration of archival recordings was made by processing them through this rather specific audio program. Another program could just as well have been written to allow subjective determinations of suitable equalization — to be executed by adjusting knobs, viewing the result and hearing the effect at the same time. Perhaps for a simpler problem however. The need for arbitrary adjustment of a large number of filter parameters is common to many audio tasks. The best solution to this problem is, practically and ideally, through the use of some method that lends itself to responses as variable as the requirements dictate.

As of March 1976, most of the capabilities discussed in this article already exist in prototypical form. We heard real 16-bit recordings played back for us, programs to control level, pan, and equalization are operative. A graphics terminal can display waveforms, control settings, equalization curves, and printed information as well. It's a 2-channel stereo operation at present, and uses magnetic disc files as the recording medium. In the near future, Soundstream Inc. will possess a digital tape recorder of their own design. This will allow recordings to be made away from their facility and brought there for subsequent signal processing work. Before too long, today's digital hardware will be extended to a multitrack configuration for studio use. Dr. Stockham believes that his sytems will be entirely affordable. Based on what was seen and heard, there is good reason to wish for continued and rapid development, and early installation of the complete system. R-e/p will continue to report on this new way of doing things, as events progress.

This article, then, is the result of the author's recent visits to the Soundstream Studio, and is based on conversations with Dr. Thomas Stockham, the president of Soundstream, and Robert Ingebretson, Director of Computer Services of the studio, as well as the author's exposure to the digital audio material produced at the facility.

AUTOMIX (with standard Model 2488 Console) installation at Technisonic Studios, Inc., St. Louis, Mo.

Programmer, Model 1024.

A studio with AUTOMIX* reaches a New Standard of Recording Excellence.

*THE COMPLETELY PROGRAMMABLE MIXDOWN CONSOLE.

For years, Automated Processes has been anticipating the future by designing and fabricating the most advanced consoles, components and synchronizers for the professional recording engineer.

Now we have combined the incomparable AUTOMIX Mixdown Console with Automated's own Model 1024 Programmer, to bring you total automation . . . a completely integrated system of matching software and hardware capable of extraordinary performance. Yet this ultra-reliable, ultra-flexible system carries a modest price tag.

The AUTOMIX Console captures every nuance and detail of a mix from up to 32 inputs and 4 echo channels. Each element of sound on any track may be improved individually or in combination, and the mix can be repeated as often as desired. Audio paths are isolated from DC control circuits, and voltage control amplifiers and state variable devices provide for all functions . . . level, localization (program and reverb), reverb level, and equalization. AUTOMIX permits optimum artistic capability, giving the mixer total flexibility in improving the mix for each product or media.

The Model 1024 Programmer uses a digital cartridge to store programming information, and therefore requires only one program track of MagLink timecode which will not crosstalk into the program tracks. During playback, data is recalled from the cartridge in sync with program tape or other timing source **with no cumulative time delay** since timing is included in the cartridge data. Data is always retained, together with new data recorded at each updating pass, to permit returning to the previous mix at will. Moreover, up to ten final mixes may be stored for later recall. Spliceless editing between mixes is easily accomplished.

Program tapes made with the Model 1024 are totally compatible from studio to studio, and the programmer can be retro-fitted into present installations.

But the most important Automated advantage is our experience. Why not let us put it to work for you. Contact us for full technical literature. Our staff is ready to discuss your personal requirements.

Visit us at Booths 104/105 at the AES Convention, Los Angeles Hilton, May 4-7.

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789 PARK AVENUE, HUNTINGTON, NEW YORK 11743 • 516-427-6024
A.P. Ltd., 17 Erncroft Way, Twickenham, Middlesex TW1 1DA, England · 01-891-2770, and Respected Dealers Throughout the World.

Circle No. 142

New Products

dbx 162 STEREO COMPRESSOR/ LIMITER WITH STRAPPING FOR MULTI-CHANNEL TRACKING

dbx has broadened its line of signal processing accessories with the introduction of Model 162 stereo compressor/ limiter. Similar in capability to the firm's 160 single channel compressor/limite, the new product offers true RMS level detection with the threshold variable from 10 millivolts to 3 volts. LED indicators show above and below threshold operation.

The Model 162 offers compression ratios variable from 1 to 1 up to infinity and maintains low distortion performance even at high compression ratios. The system maintains proper stereo image location by using the sum of the channel signals as the control voltage. Four or more channels of operation are achieved through strapping so that a single voltage proportional to the sum of the channels controls all the outputs of two or more 162's..

Meters are switchable to read input or output level or gain change over a 60dB range and zero level may be set anywhere from minus 10 to plus 10dBm. The 162 accepts input levels up to +26 dBm and will drive a 600 ohm load at +24 dBm.

The Model 162 is a fully professional, rack mount compressor/limiter with barrier terminal strip connectors and is ground loop compensated and protected against power turn-on, turn-off transients.

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

Circle No. 143

MM1200 NEW AMPEX MULTI-CHANNEL RECORDER ANNOUNCED

The MM-1200 is available in 8, 16 and 24-channel versions and is designed to work with various synchronization equipment in professional audio-video dubbing and mixing operations.

Standard on the MM-1200 is a common chassis and transport, plus a new electronics packaging which permits the 8 and 16-channel recorders to be easily upgraded to 24 channels.

Coupled with optional EECO time code accessory equipment, the MM-1200 offers precise synchronization of multi-track audio recorders with quad or helical video recorders, or for synchronizing two or more audio recorders.

EECO components available include wide range synchronizer, dual cue controller, edit code generator, edit code reader and video character generator.

The MM-1200 offers an unrestricted choice of operating modes, including individual selection of Sel-Sync, input, and repro for every channel, according to Charles A. Steinberg, Ampex Vice-President/General Manager of the Audio-Video Systems Division.

Every individual channel has a separate LED indicator light for each function and operating mode.

Standard on the MM-1200 is a built-in search-to-cue which permits the operator to set a cue point anywhere on the tape, then return to that exact point automatically from either direction at shuttle speed. This exclusive feature is accurate to ±0.5 seconds at 15 ips, and searches smoothly and precisely with no overshoot.

Also standard is a sophisticated new control panel with optional remote capability, an electronic tape timer, and a digital readout that registers up to ± one hour, 59 minutes, 59 seconds. NAB or CCIR equalization standards are also featured.

A universal power supply for 105-250V, 50-60 Hz, is offered in all versions of the MM-1200 and a newly designed ventilation system has been built in.

The new machine operates at speeds of 7½, 15 or 30 ips and is capable of handling 14-inch reels.

The MM-1200 features new rotary tape guides which eliminate friction between tape and guides. A special mu-metal shield in the head cover protects the heads against prevalent 50/60 Hz hum fields.

According to Steinberg, all electronics in the MM-1200 employ state-of-the-art backplane type wiring assemblies for better reliability and easier service. New improved connectors are used throughout and tape tension adjustments are conveniently located on top of the machine.

The head assembly can be changed rapidly by loosening a single screw. When one-inch and two-inch heads are interchanged, the tape tension servo is automatically switched to the correct range.

A special accessory for the MM-1200 is a Pick Up Recording Capability (PURC) which permits the editing or dubbing of new material over previously recorded material without creating errors at either end of the new insert.

Other accessories include a video layback head for sweetening or mixdown of audio tracks for video tapes, and Auditec II which allows the multiple tracks of voice, music and sound effects to be recorded and mixed in synchronization with the video.

In addition the EECO time code synchronization systems, the MM-1200 also offers as optional equipment a VS-10 variable speed oscillator, sync lock, overhead accessory bridge assembly, WBP-2 SMPTE wideband preamplifier, and two-inch stainless steel tape splicer which fits on top of the head assembly for editing convenience.

Price for the MM-1200 range from $17,000 to $32,000. Delivery is 30 days ARO.

A color brochure describing the MM-1200 and its operation is available by writing and requesting brochure A-757.

MARKETING COMMUNICATIONS, M. S. 11-12, AMPEX CORPORATION, 401 BROADWAY, REDWOOD CITY, CA 94063.

Circle No. 146

TAPCO 4400 REVERBERATION SYSTEM

The 4400's two independent channels each have a four band graphic equalizer that allows the user to tailor the reverb sound to simulate the qualities of virtually any room. At a pro net cost of only $389.00, the 4400 is said to be in a price class by itself.

Tapco claims to have eliminated the problems associated with conventional spring reverbs. The ping-pong effect that has always identified the sound of a spring reverb is caused by sharp high energy signal pulses that actually over drive the reverb tank. The Tapco 4400 has built-in signal processors that smooth out transient peaks, without disturbing the dynamics of normal signals. These unique circuits, combined with dual differential constant current drive amplifiers, produce, it is said, a quality unequalled in low cost reverb systems.

All functions necessary for everyone from rock musicians to recording engineers are provided on the 4400. The input level to each channel is set by

Circle No. 145

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the console company,
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Circle No. 147

SPHERE ELECTRONICS
THE CONSOLE COMPANY
20101 G PRAIRIE AVENUE
CHATSWORTH · CALIFORNIA · 91311
PHONE: (213) 349 · 4747

AutoPad volume control circuitry, and constantly monitored by the VU meters. Each channel even has its own four band graphic equalizer. By carefully tuning the equalization the operator can create the sound of any room, from huge concert halls to ordinary living rooms. Reverb percentage mix controls govern the amount of reverb in the output signal, and output level controls set the overall volume. Tapco's exclusive input mute switch momentarily quiets the input signal so the reverb sound can be evaluated by itself, while the equalization is adjusted to create the perfect ambience for any situation.

The 4400 is designed to be compatible with all professional and semi-professional audio gear.

TAPCO, 405 HOWELL WAY, EDMONDS, WASHINGTON 98020 (206) 775-4411

Circle No. 148

BALANCED IN-OUT 600-OHM STUDIO EQUALIZER FROM SOUNDCRAFTSMEN

The new Twin-Graphic Equalizer utilizes four Light-Emitting-Diodes to provide a visual front panel display for zero-gain input to output signal ratios. Other features include two completely separate ten-octave equalization panels, with plus or minus 12dB boost and cut provided individually for each octave. Separate equalized signal zero-gain controls are used for each channel, enabling exact balancing of input to output with a plus 6dB and minus 12dB range. For precise balancing, Light-Emitting-Diodes are used in conjunction with the zero-gain level controls so that visual as well as audible balancing can be accomplished quickly and easily.

Front panel pushbuttons provide selection of either an equalized or unequalized output, low and/or high shelving, and zerogain lites on or off. Separate terminations are provided for input and output of Section A and Section B.

The TG2209-600 is ideally suited for professional use in mix-down rooms, Tape-to-Disc transfer, Radio & T.V. production, P.A. Feedback Suppression, Environmental Equalization, and Sound Reinforcement.

The new unit features balanced 600 ohm op-amp input, balanced 600 ohm op-amp output, and switch selection for low or high impedance input or output.

Price is $550.00.

SOUNDCRAFTSMEN, 1721 NEWPORT CIRCLE, SANTA ANA, CA 92705 (714) 556-6193.

Circle No. 150

NEW APD1600-16-24 TRACK RECORDER BOWS

Bouse Manufacturing Company's president, Thomas Bouse announces that his Audio Products Division's model APD-1600-16-24 track studio tape recorder has completed its final production testing and will be introduced to the industry at the May, A.E.S. Show in Los Angeles.

The new recorder is the end product of nearly four years of intensive effort by the Bouse organization to provide the

industry with an extremely rugged, reliable tape transport and sophisticated state-of-the-art signal electronics.

Marketing of the new Bouse recorder will be in the hands of Bill Jones. Audio Electrical Supply, Inc., will distribute the product in the Western United States and Canada, with other area sales representatives to be announced soon.

AUDIO ELECTRICAL SUPPLY, 15466A CABRITO RD., VAN NUYS, CA 91406 (213) 787-3679 / 873-3929.

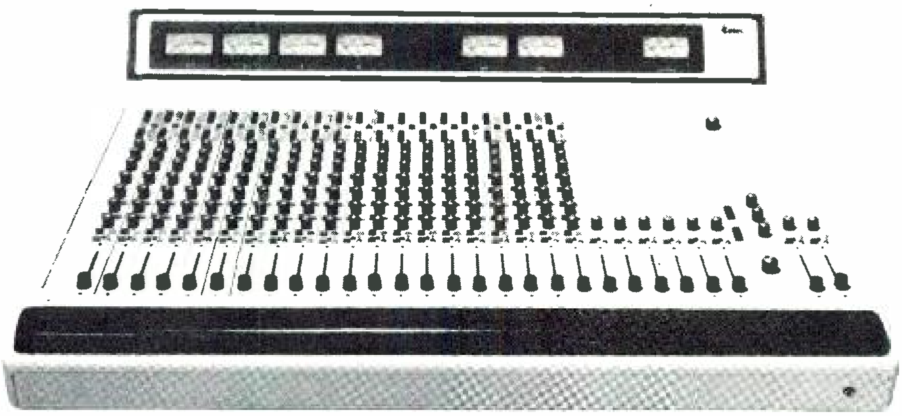
Circle No. 151

CETEC SERIES 20A AUDIO CONSOLE

North Hollywood, California, Cetec Audio has announced its Series 20A Audio Console. The Design incorporates systems innovations that allow it to meet the real-time demands of television production, sound reinforcement, and theatre effects – together or separately. The Series 20A is modular, with four chassis/enclosure sizes available: 2½' with 21 module positions, 4' with 31, 5' with 39, and 6' with 47 positions. The modularity goes beyond the benefits of serviceability or expandability. The operator may plug modules into any position to customize the arrangement for a particular show without affecting their function.

The emphasis is on human engineering

Circle No. 149



and ability to pre-determine program content and distribution in order to simplify real-time operation.

INTERSTAGE PATCH POINTS are mounted at the top of input and sub-master module for easy identification.

PRESET ON selection on input modules allows sources to be activated in predetermined groups.

VARIABLE MICROPHONE PRE-AMPLIFIER provides smooth control of signal overdrive conditions with live mics.

PEAK INDICATOR aids in adjusting mic sensitivity by flashing occasionally at normal signal levels while preventing over-load by more consistent illumination at higher levels.

Modules available in the Series 20A line include:

INPUT, which includes balanced pre-amp and line inputs, equalizer, straight-line attenuator, stereo panner. Three multi-purpose (cue and echo) sends, and 4 or 8 submaster mix busses assign switches.

SUBMASTER, which contains the summing amp with echo receive input control, straight-line attenuator, and line output amplifier.

PAN SUBMASTER, which is the same as above but with a stereo panner.

MASTER, with all monitor, solo, echo receive, and combined output electronics and controls.

In addition, the PRE-SET DISTRIBUTION SYSTEM (PDS) module is provided to control the outputs of a Series 20A console where required. Especially useful for theatrical productions, PDS makes it possible to rehearse, preset, and perfect the mood, movement, and even the physical space relationships of voices, music, and effects. For television studios, PDS provides a separate sound reinforcement mix matrix for controlling feedback when microphones are used with a live audience.

For further information, contact Mr. Bob Slutske,
CETEC AUDIO, 13035 SATICOY ST., N. HOLLYWOOD, CA 91605. Telephone (213) 975-1900.

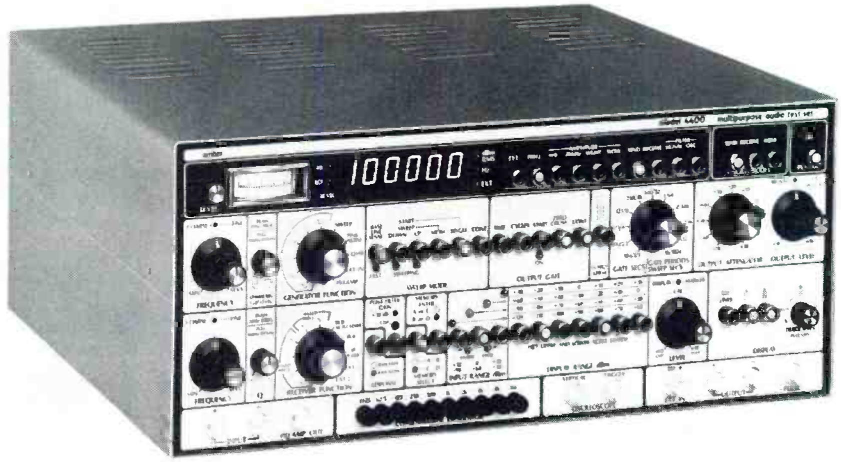
Circle No. 152

AMBER MODEL 4400 MULTIPURPOSE AUDIO TEST SET

This comprehensive and powerful instrument is the result of an extensive development program to produce a complete, yet cost-effective test facility for the professional audio industry.

The instrument consists of a generator and a receiver. The generator section incorporates a multi-waveform function generator, pink noise source, log sine wave sweeper and comb generator. It has facilities for tone bursts, a +30dBm balanced output capability and very low distortion.

The receiver section contains an auto-ranging digital level meter reading in dBm



RMS, an autoranging digital frequency counter, a spectrum analyzer, a wave analyzer, a band pass, band reject, high pass, low pass filter and a four channel digital memory to store response plots.

The instrument is used with any DC oscilloscope to measure level, gain, noise, crosstalk, distortion, frequency and phase. Four digital memories permit plots of amplitude or phase versus time or frequency.

Applications for the instrument include equalizer and filter measurements, console verification, tape recorder line up, room equalization, production line testing, microphone and speaker response measurements, spectral analysis and transmission line testing.

The 4400 is priced under $3,000 and deliveries will begin May 1976.

AMBER ELECTRO DESIGN LIMITED 1064 CHEMIN DU GOLF, MONTREAL, QUEBEC H3E 1H4 (514) 769-2739.

Circle No. 154

VEGA INTRODUCES NEW DIVERSITY WIRELESS MICROPHONE SYSTEM

This new system utilizes the VEGA PRO Series transmitters and receivers in a diversity reception mode that it is claimed, virtually eliminates all fades and dead spots. Fades and dead spots are caused by interference between direct and reflected radiation that cancel, resulting in loss of signal. The problem is most prevalent in "studio" operations, but also occurs outdoors.

In the diversity mode, two VEGA PRO receivers, placed three feet or more apart, both receive the transmissions. Because the two receivers are more than one-half wavelength apart, both will not have signal cancellations at the same instant. Both receivers feed a Model 62 Diversity Combiner that selects the receiver with the best signal strength within microseconds. The switching is immediate and noiseless. The resultant audio, the best of both receivers, is noise free and drop-out free.

The VEGA Diversity System is composed of Modesl 54 or 55 Transmitter, two Model 58 Receivers and Model 62 Diversity Combiner.

VEGA, DIVISION OF CETEC CORP., 9900 BALDWIN PLACE, EL MONTE, CA 91731, (213) 442-0782.

Circle No. 156

CL&S INTRODUCES NEW BRH90 RADIAL HORN

Community Light & Sound presents BRH90 Radial Horn now available from stock. Manufactured with throats for both 2" and 1.4" bolt-on drivers, the

BR1190 has a flare rate of 240Hz, which permits a crossover as low as 500Hz, and horizontal dispersion of 90 degrees at all frequencies of operation. The horn is constructed of black, hand-laminated, non-resonant fiberglas.

COMMUNITY LIGHT & SOUND, 5701 GRAYS AVE., PHILADELPHIA, PA 19143, (215) 727-0900.

Circle No. 159

RICHMOND PORTABLE STEREO MIXING SYSTEMS

Among the exclusive features of the new M82 II / S42 II models is a completely voltage controlled gain stage in each input channel allowing direct interfacing with automated mixdown equipment and electronic music synthesizers. This also provides facilities for virtually unlimited submaster combinations, allowing level adjustments by subgroup as desired.

Beyond this, flexible equalization, foldback and echo send channels, and complete monitoring facilities combine with rugged construction and professional specifications to make an ideal Portable Stereo Mixing System; available in 8, 12, 16, 20, and 24 input channel formats, expandable as desired.

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

Circle No. 160

SHURE OFFERS FREE SOUND REINFORCEMENT COMPONENT APPLICATION GUIDEBOOK

Shure Brothers Inc., has prepared a new, 16-page brochure describing how products in its SR line of sound reinforcement components handle critical sound requirements ranging in size and complexity from those presented by mammoth outdoor concerts to those in intimate lounges.

Shure's SR component line now includes an audio console, two power amplifier models, an extended range speaker system, a portable speaker column, a speaker column for permanent installations, an electronic crossover, and an array of custom accessories.

The new brochure contains case histories of how these components are being used as complete systems or integrated with other equipment in Las

Vegas and Reno showplaces; outdoor concerts; theatres and nightclubs; theme parks; meeting rooms and restaurants; houses of worship; auditoriums; and traveling sound reinforcement systems.

For your copy of this free brochure (AL525), write:

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

Circle No. 163

PACIFIC RECORDERS AND ENGINEERING NEW TVI (TAPE VELOCITY INDICATOR)

Now available, the digital variable speed unit (±33%) is designed for use with the MCI JH-110, JH-100, and JH-114 tape recorders.

Ten-turn vernier speed control provides the fine adjustment capability to shave time off a "61 second" spot, sharpen a flat, compensate for off-speed tapes, and accomplish any number of special effects, such as flanging and delay. Front panel button switching selects between fixed (crystal) and variable speed.

The TVI is a completely self-contained unit drawing its power from the decks servo connector, and can be remoted up to 33 ft. with an accompanying cable. Unit can be built into your console or can be enclosed in a formica cabinet.

PACIFIC RECORDERS AND ENGINEERING CORPORATION, 11100 ROSELLE ST., SAN DIEGO, CA 92121 (714) 453-3255.

Circle No. 164

TAPCO MODEL 2200 GRAPHIC EQUALIZER

TAPCO's new 2200 Graphic Equalizer is designed for professional recording and sound applications. The 2200's two independent channels have ten equalization bands, each controlling one octave of the audio spectrum with 15dB of boost and cut. The 2200 comes factory equipped with VU Meters, both balanced and single-ended inputs and outputs, and 19" rack mounting package. The pro net cost is only $289.00.

Because equalizers find use with practically every kind of audio gear, TAPCO has designed absolute compatibility into the model 2200. Built-in line drivers allow the 2200 to be interfaced directly with hi-fi equipment and low level mixing systems like the Tascam Model 10. The equalizer's balanced and

single-ended inputs and outputs can be used in any combination. That means the 2200 can be used in place of transformers between pro and semi-pro equipment. Separate EQ In/Out switches allow each channel of the equalizer to be programmed independently. And the 2200's VU meters can be set for any reference level with the calibration controls, accessible through the front panel.

Maximum input level is better than 10 volts RMS, and maximum output level is greater than +12dBm into 600 ohms, or 10 volts RMS into more than 2000 ohms. Harmonic distortion is below .06% at 10 volts output, and signal to noise ratio is specified at better than 80dB, typically 85dB.

TAPCO, 405 HOWELL WAY, EDMONDS, WA 98020, (206) 775-4411.

Circle No. 165

ENGLAND'S SOUNDCRAFT SERIES II CONSOLES AVAILABLE FROM SYSTEMS & TECHNOLOGY IN MUSIC

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Included in the new 111B is the unique "floating threshold limiter" which minimizes "spring twang" and provides absolute protection from overload. Also retained from the previous model are line-level balanced outputs and smooth four-spring (per channel) sound.

The Model 111B comes in a standard 19" rack mount and is 3½" high. Price for the dual-channel 111B is the same as our old single-channel 106CX: $695.00 (including power supply).

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Circle No. 167

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The Model APM 176 Audio Wattmeter is a precision broad band, solid state, instrument said to be ideal for measuring the 'frequency vs. power delivered' signatures of 4 to 16 Ω load loudspeakers and audio systems at full scale values from 3 to 300 watts, in 5 ranges, plus a 0 to 10 dB reference scale.

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Circle No. 168

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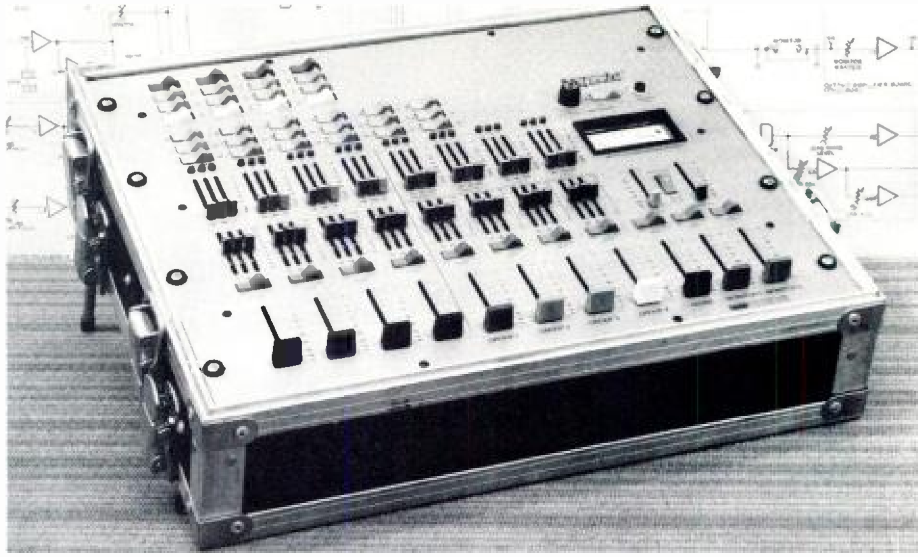
Called the TROUPER IV, the new system has been designed to fit the needs of the most demanding requirements. The console is composed of Output Control and Input Expander modules weighing less than 30 pounds per piece and measures 15¾ x 19 x 4½ inches.

Each input contains: level control, solo switch, 3-band graphic eq, two monitor and one echo send controls, COMPUGUARD Peak Limiting with 3-level peak indicator which displays over a 70dB range, mic pads and mic/



line switch combination which allow you to pad a total of 70dB in 10dB steps, and subgroup assign switches. The output section features include: 4 subgroup level controls, 3-band graphic eq on each subgroup, echo receive control, House and Monitor master level controls, solo preview system with switchable VU meter and peak indicator, preview selector switch for listening pre or post faders in the earphones, headphone jack and level control, independent announce input level controls for house or talkback to the monitor systems, 2 line input level controls for background music or on stage cues, front panel mounted power switch and fuse.

Back panel features: XLR type connectors for all balanced inputs and outputs, on/off switch for built-in phantom power supply for condensor microphones, external equipment jacks to inject signal processing equipment into the Trouper system, echo send and receive jacks, individual subgroup outputs with external equipment contracts, subgroup defeat switch which inactivates the subgroup system for mono operation (subgroun output may still be used for recording feeds or monitor channels), console interconnect for the additional Input Expander Moduales (38 Inputs Total), and 3 external equipment power jacks for Uni-sync accessories (ie: Meter Line Amp Package, Dual Graphic Equalizer with COMPUGUARD, Stereo Pan Package, Quad Limiter/Meter Package, and Spring Reverb Unit).

The Trouper IV is rack mountable in a low profile all metal console cabinet or available in durable flight cases.

UNI-SYNC, MARKETING DIVISION, 5559 CAHUENGA BLVD., NORTH HOLLYWOOD, CA 91601.

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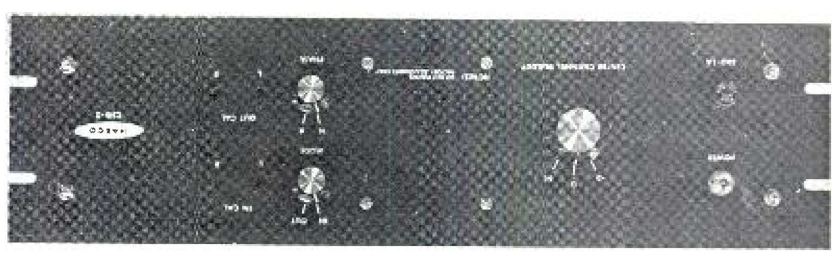

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. . . continued from page 44

ALLISON RESEARCH

the one inch offset exactly. The point of light, of course, is indicative of the audio gain of the channel, and may be addressed by either the operator (via the belt), the automation system, or other over-riding sources such as group masters, presets or master clearing devices.

The fabulous Fader covers a 112dB range of operation which is incremented in 1/4dB steps over the first 48dB, then 1dB steps over the remaining 64dB of its range. It is, of course, noise free in its operation and immune to analog forms of wear. The belt itself is a continuous splice free lamination of mylar, nylon and polyurethane, and, though its wear cycle is estimated at 10 to 15 years, is quickly and inexpensively replaced. Such replacement, or cleaning, would be indicated in the event of catastrophies such as cigarette burns and such.

The faders themselves are arranged on a conventional one-per-track basis, allowing the operator instant and instinctive access to all system levels.

The singular button/LED at the top of

concluded overleaf . . .

each fader, however, serves as access to centrally controlled modes associated with each channel.

The level control sub-system, a current ALLISON RESEARCH production item, includes the following centrally controlled parameters, all of which are fully programmable and are individually accessible, one channel at a time, or on an ALL CHANNELS basis:

1. Selection of WRITE or READ/AUTOMATIC UPDATE modes, with respect to levels.

2. A Clear Fader position (Fader off), and a TEST LEVEL position (Gain = Unity ±1/4dB)

3. Solo and Clear Solo positions, as well as READ/WRITE status with respect to solos.

4. Mute and Clear Mute positions, as well as READ/WRITE status with respect to Mutes.

5. Assignment of any channel to one of fifteen group masters, together with READ/WRITE status with respect to group master assignments.

6. Assignment of any channel to one of four input sources, together with READ/WRITE status with respect to input assignments.

7. Separate mute/solo systems for Group Masters, as well as READ/WRITE controls for same.

8. Four addressable RAM presets (expandable to 64), each of which can store all settings of all parameters within the level section, and can be loaded or activated either one channel at a time, or on an all channels basis.

All of the above parameters are controlled with a central matrix of 26 momentary buttons with LED indicators, which is used in conjunction with the singular button/LED located on each channel or group fader.

A unique bi-directional visual communication system allows the operator the required complete visual monitoring of all system parameters.

Since space does not permit this article to continue on much longer, I will briefly describe the partial operation of the system, and leave it up to the reader to piece together the remainder of its operation.

As with the Great Equalizer control, pressing a singular channel button causes the Level Sub-Section central control to indicate all parameters currently associated with the selected channel. If it is desired to change parameters, the desired buttons on the Central Control are operated while the channel button is held down. Reverse communication is also possible, on the following basis: Assume, for instance, that the operator wishes to see which, if any, channels are muted. Pressing the MUTE button on the Central Control causes all channels or groups which are in the muted state to indicate that fact, via their LEDs associated with their singular buttons. The same philosophy applies to all parameters associated with the system.

A further visual communication system exists in that the linear LED array associated with each fader does, indeed, indicate the actual gain of the channel, and includes the effect of mutes, solos and group masters. If a channel is solo-ed, for instance, all other channels will indicate an off condition with their LED arrays. Similarly, if a group master is moved up and down, all channels assigned to that group will indicate this up and down motion on their LED arrays. This, of course, is a direct and instinctively correct visualization of what the effects of the controls actually are.

It can easily be seen that an attempt to configure this degree of programming versatility, on a conventional basis, would result in the addition of some 35, or so, controls and indicators to each module — an addition which would render the system incapable of meeting the requirements which we have listed previously. This is to say nothing of the drastically increased costs and gross operating complexities which would inherently result.

CONCLUSION

Programmable audio systems can offer the user a powerful tool in the execution of his creative work. The real advantage of such a system lies in the operator's ability to shape the control of his precious audio to degrees which heretofore were impossible because of physical limitations.

In configuring complex systems, we, the manufacturers, have the responsibility to fulfill this promise on a basis which does not trade all of our gains for detriments, such as unmanageable size, inoperable controls, over-complexity in the human interface, unnecessary compromises in the system's capabilities, and above all, unbearable increases in the cost of the system.

We, at Allison Research, believe that this can come about only by approaching the problem with a logical and flexible pattern of thought, which is unencumbered by the shackles of conformity to past methods.

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It speaks for itself

HARRISON 3232 MASTER RECORDING CONSOLE
TYPICAL PERFORMANCE CHARACTERISTICS

Measurements quoted are typical of a 24 input console and are the average of 10 input modules measured in serial #001. In no case were any individual distortion figures 50% more. All noise measurements were within 1 db and all crosstalk was within 2 db of the average.

Unless otherwise stated, all measurements were taken with VCA set for unity gain and mike trim adjusted so that a -50 dbm input at 1 KHZ produced a +4 dbm output signal. Inputs not driven were terminated with a 150 ohm resistor. All measurements were taken with all VCA's, transformers, etc. in place.

Mike Preamp Equivalent Input Noise
(20 KHZ Bandwidth) -127.5 dbm

Output Signal to Noise (20 KHZ Bandwidth)
(-50dbm Input Set For +4 Out) 77.5db

IM Distortion of Mike Preamp with +2dbm Input
(equivalent to Sine Wave + 4)
Set to Produce +24 Out .008%

IM Distortion Total Channel
(-50dbm in, +4dbm output .05%)
(-30dbm in, +24dbm output .15%)

Crosstalk with all inputs assigned to their respective outputs;
inputs on either side of measured output driven
(10KHZ -82db) (1KHZ -85db) (100HZ - 85db)

Crosstalk with adjacent input modules assigned to each other
(10KHZ -78db) (1KHZ -83db) (100HZ -84)

Frequency Response (20HZ to 20KHZ)
Ref 1KHZ (-0 -1db)

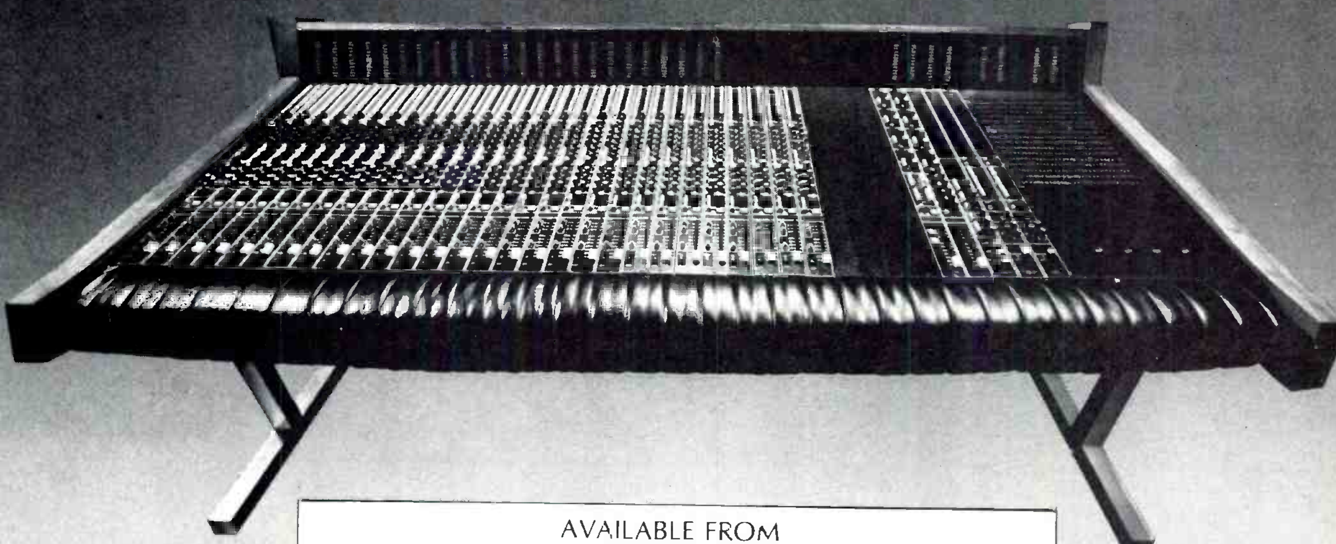
Head Room (output set below threshold of clipping or triangulation

10K Load	(20HZ +28.5dbm)	
	(1KHZ +28.5dbm)	
	(20KHZ +28dbm)	
600 Ohm Load	(20HZ +25.5dbm)	
	(1 KHZ +25.5dbm)	
	(20KH +25dbm)	

Source Resistance of Line Outputs
(all frequencies 20HZ to 20KHZ) 60 ohm

Level Change at Output
(600 ohm Term to 10K Term) .85db

Specifications
subject to change without notice.



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