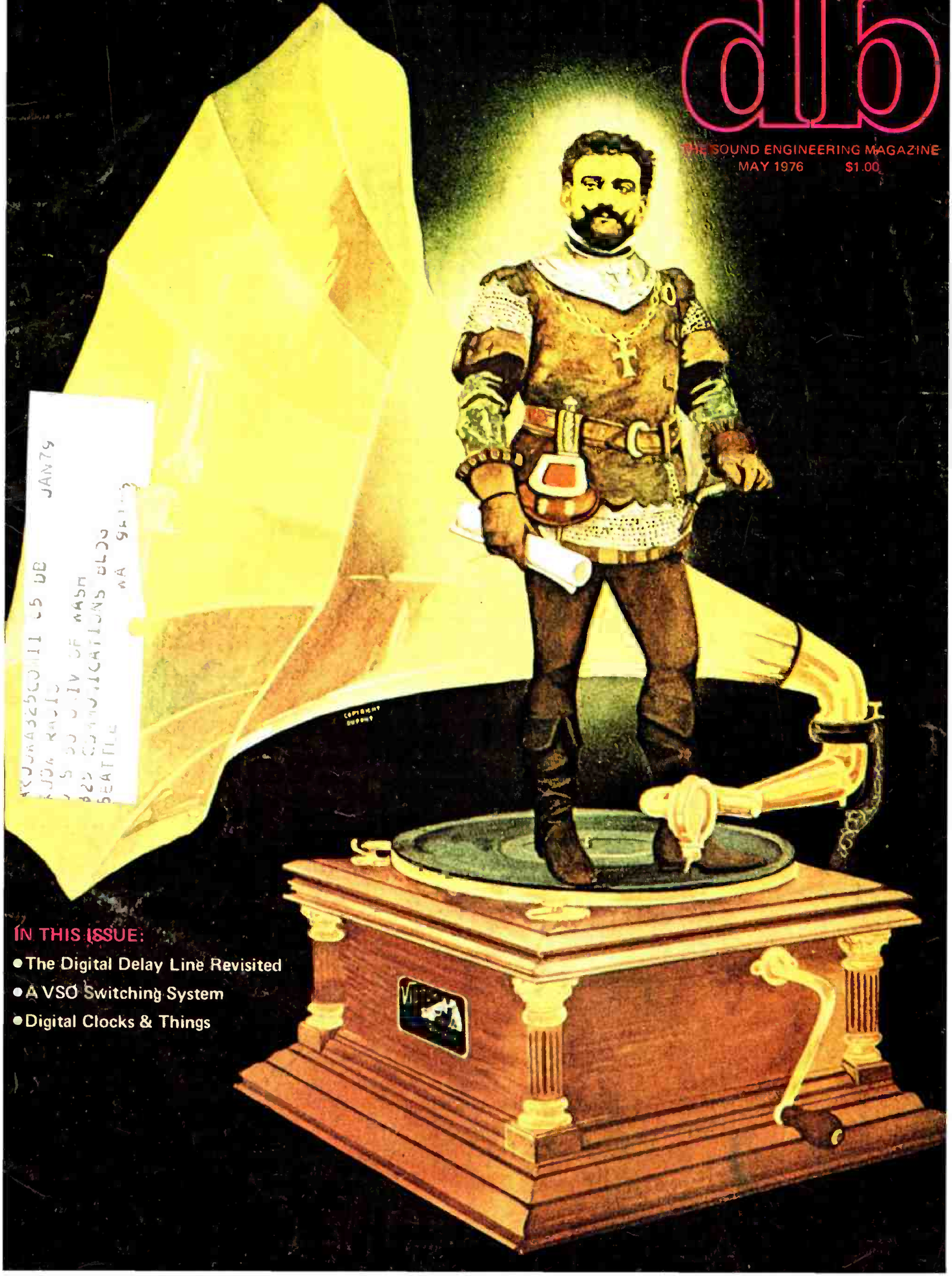
ISSN 0011-7145

db

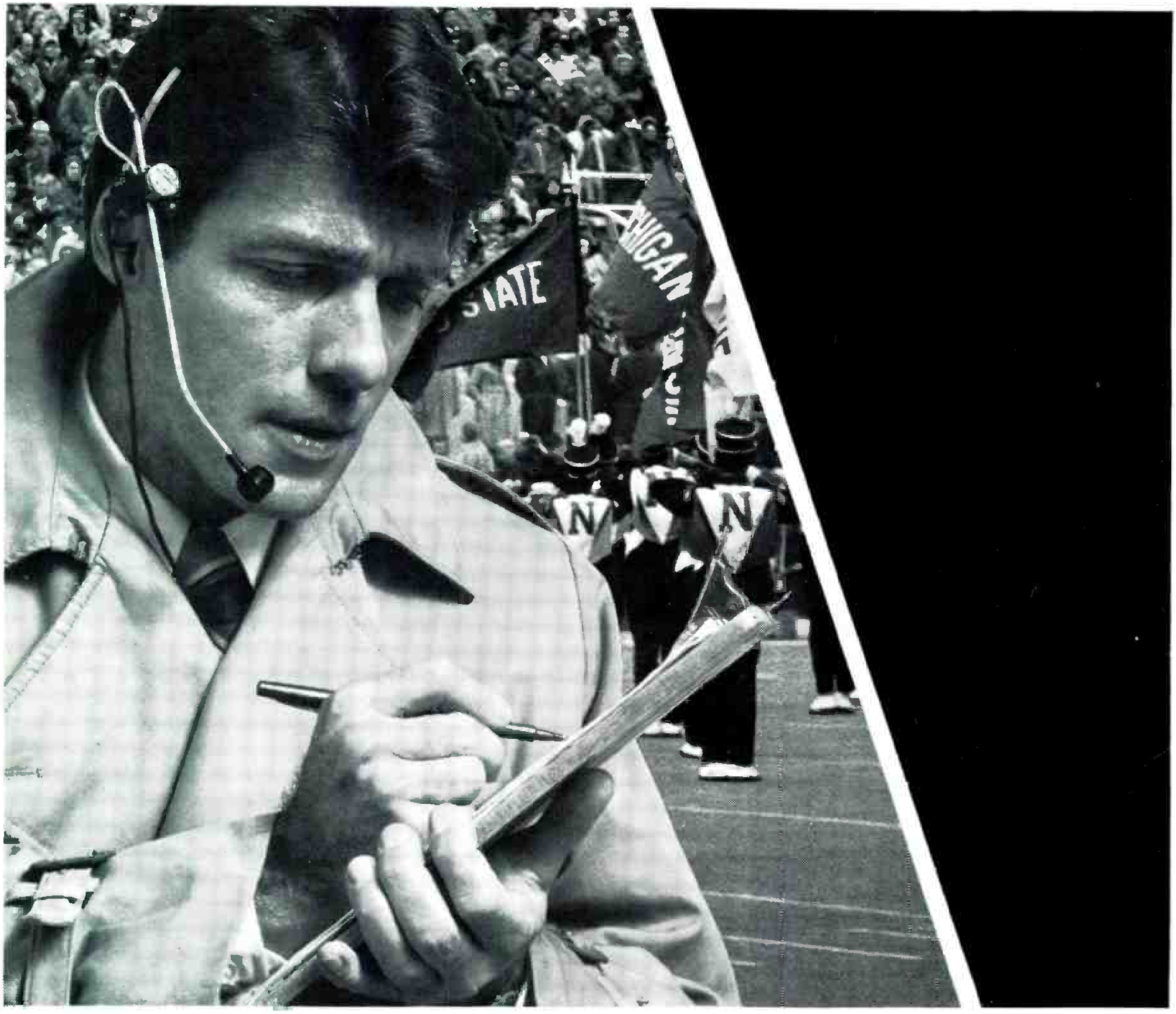
THE SOUND ENGINEERING MAGAZINE

MAY 1976 $1.00

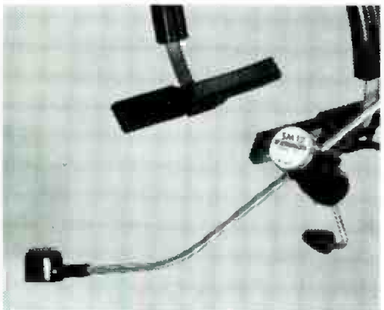


IN THIS ISSUE:

- The Digital Delay Line Revisited
- A VSO Switching System
- Digital Clocks & Things



Next best thing
to a sound proof booth.



Shure's new headset microphones are coming through loud and clear. With their unique miniature dynamic element placed right at the end of the boom, Shure's broadcast team eliminates the harsh "telephone" sound and standing waves generated by hollow-tube microphones. The SM10 microphone and the SM12 microphone/receiver have a unidirectional pickup pattern that rejects unwanted background noise, too. In fact, this is the first practical headset microphone that offers a high quality frequency response, effective noise rejection, unobstructed vision design, *and* unobtrusive size.

Shure Brothers Inc.
222 Hartrey Ave., Evanston, IL 60204
In Canada: A. C. Simmonds & Sons Limited

SHURE®

Manufacturers of high fidelity components, microphones, sound systems and related circuitry.

Circle 10 on Reader Service Card

Rock and roll is in it's third decade and there are mountains of blown diaphragms and discarded speaker systems as evidence of the difficulties loudspeaker manufacturers have had in meeting the challenge. The SP1 was designed and tooled by a new loudspeaker company dedicated to solving the basic difficulties of high level sound reinforcement in order to meet that challenge.

Our two-way is a compact, powerful, reliable, high fidelity loudspeaker with dispersion and power response so uniform that the "sound" of the system is stable in different environments. The SP1's multi-flare radial horn is the most significant advance in the control of high frequency dispersion since the invention of the radial horn half a century ago. Undoubtedly the design will become the industry standard.

The real marvel of the SP1 system, however, is the Model 22 Compression Driver. We have been producing it since August 1975 demonstrating that it is possible to combine adequate high end response (13 KHz), efficiency (30% midband), high power handling (40 watts pink noise 8 hours continous), reliability (6 forms of on-line analysis plus listening), and good sound in a compression driver. Until "22" a high performance 2-way like the SP1 was not possible.

The low-end of the system is provided by a 15" horn loaded cone speaker covering the range of 60 - 500 Hz. The extensive Q.C. system devised for the driver has been expanded to allow the same scrutiny of the SP1.

Some strong statements have been made here, but we know we can deliver. The demand is so great for a system with the SP1's performance that our dealers have ordered over 1,200 units (as of Dec. 1,'75) based on word of mouth from the few people who have heard samples from pilot production.

The SP1.

AN ALTERNATIVE TO THE ESTABLISHED WAY

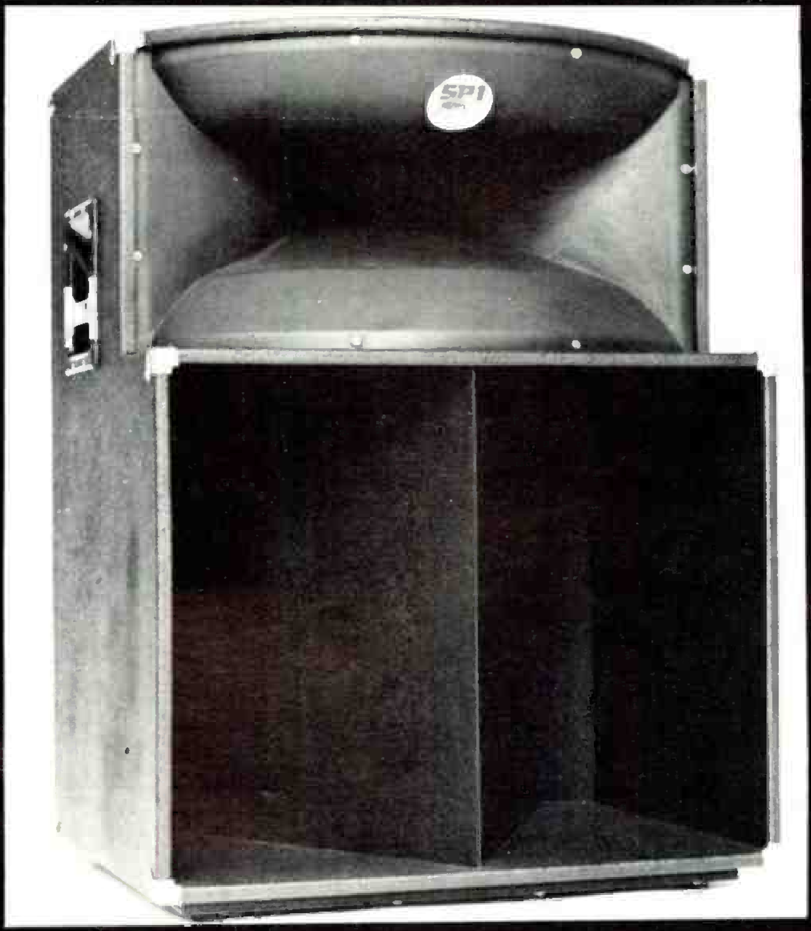
$499.50* Soon at your Peavey Electronics Dealer.

PEAVEY

Peavey Electronics, Corp.
PO Box 2898
Meridian, Ms 39301

*suggested retail

SP1



Circle 16 on Reader Service Card

STL Offers The Most

Complete Selection

Of Test Tapes Available Anywhere

If you are looking for precision test tapes, look no further. STL can serve all your needs with 150 mil cassette tapes, tapes in ¼", ½", 1" and 2" sizes, flutter tapes and magnetic test films in Super-8 mm, 16 mm or 35 mm. STL gives you the most accurate reference possible in the widest range of formats. Order from us and find out where your system really is.

Write for a free brochure and the dealer in your area.

Distributed exclusively by Taber Manufacturing & Engineering Co.

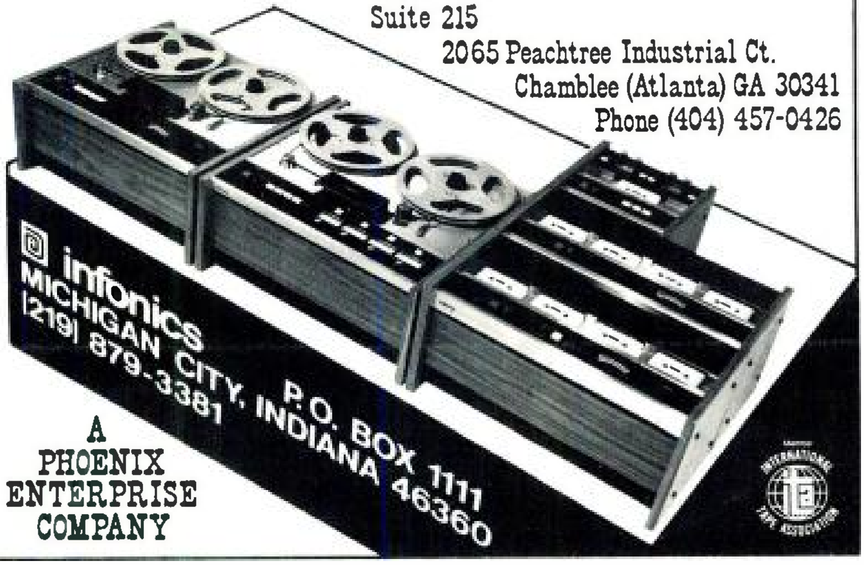
STL STANDARD TAPE LABORATORY, Inc.
2081 Edison Avenue
San Leandro, CA 94577
(415) 635-3805

Circle 14 on Reader Service Card

COMMERCIAL DUPLICATING SYSTEMS

INSTALLATION AND PREVENTIVE MAINTENANCE TRAINING INCLUDED IN PRICING

Southeast Regional Service Center
TECH SYSTEMS
Suite 215
2065 Peachtree Industrial Ct.
Chamblee (Atlanta) GA 30341
Phone (404) 457-0426



A PHOENIX ENTERPRISE COMPANY

Circle 15 on Reader Service Card

db letters

THE EDITOR:

I am interested in obtaining schematics of electronic effects for guitar—such as octave box or phase shifter. I would appreciate any available information.

MICHAEL FATH
Rt. 2. Box 198
Sterling, Va. 22170

THE EDITOR:

I would like to take exception to a portion of Pat Finnegan's otherwise excellent article on f.m. stereo separation in the March, 1976 issue.

Two channels with different response curves, even if out of phase, will not, in fact, affect channel separation in a properly operating f.m. stereo transmission and receiving system. Once the two channels enter the matrix in a stereo generator, you must stop talking about two channels. Only after the f.m. signal has been dematrixed in the receiver can you start talking two channels again. If there is a problem in the bandpass of the stereo generator, transmitter, or receiver, separation may indeed be affected. But remember, we are not talking about two channels; rather, a composite signal containing L + R and L − R.

Let us imagine two channels with non-identical response curves. One has a flat response from 50 Hz to 15 kHz The other has no response at all (i.e. it is shorted out). In the transmission system L + R will equal L − R and the receiver will decode just one channel. Separation is good.

Changes in channel phase relationships from the original will indeed affect what the monophonic (L + R) listener hears and will indeed alter what the stereo listener hears, perhaps as apparent poor separation, but actual measured separation of the system should not be affected.

I believe Mr. Finnegan is confusing the L and R signals in the audio lines with the L + R and L − R signals in the transmission system.

DAVID E. DOUGHTY,
Chief Engineer
WTLB
Utica, N.Y.

Mr. Finnegan replies:

There can be a big difference between *measured separation* and the separation (or lack of it) that we hear on program material which has passed over our station. As broadcasters, we must also be concerned for this program separation. The left and right AUDIO channels ahead of the stereo generator can have considerable effect on the separation of this program material even though our measured separation of the system is good. Our sine wave method of measurement which uses only one channel at a time cannot adequately measure the effect of the audio channels on program separation. About the only real method we have available is a critical listening test on a good receiver.

Good separation at the output of the stereo generator requires that the generator be adjusted and balanced properly. This is done with sine wave (usually 400 Hz) at a fixed amplitude to one and then the other inputs of the generator. The inputs are balanced, the SUM and DIFFERENCE signals balanced and separation controls adjusted for best separation. Under these test conditions, the response curve of either channel will have little effect, for as Dave says, shorting out one channel is immaterial to the measurements. The stations left and right audio channels are really extensions of the generator's left and right inputs and should be included in the balancing and adjustment process of the setup procedures.

During programming, however, a different situation exists. Now both left and right audio channels are very much active and at the same time. Phase and amplitude discrepancies of these channels will affect and change the original signal as it appears at the matrix and poor separation can result. Or we may look at this in a different way: since our generator was balanced against fixed input signals, then when program signal voltages appear on the audio lines in those areas of the bandpass where the discrepancies occur, the stereo generator is now actually unbalanced as far as these signal voltages are concerned and the separation can suffer. This poor separation of program signals can occur in spite of the fact that we have measured good separation of the system with sine wave signals. ■

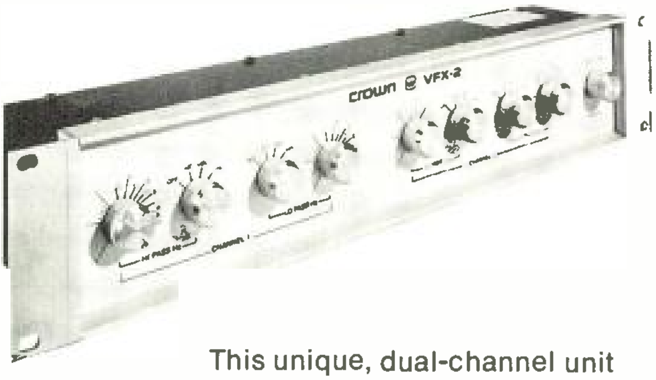
Copies of db

Copies of all issues of **db—The Sound Engineering Magazine** starting with the November 1967 issue are now available on 35 mm. microfilm. For further information or to place your order please write directly to:

University Microfilm, Inc.
300 North Zeeb Road
Ann Arbor, Michigan 48106

Are you sure what the crossover point for your next installation should be?

If not... you might think about including a Crown VFX-2 in your tool kit.



This unique, dual-channel unit has continuously variable filters. With it you can "fine-tune" the crossover point in any sound reinforcement system.

As a temporary test rig, the VFX-2 installs quickly. You can diagnose crossover problems in existing systems, no matter how old or new, and prescribe a solution.

For permanent installation, you'll find that the VFX-2 costs *less* than many fixed filters, and provides other advantages. For one, a 15dB gain that eliminates the need for input transformers. An 18dB per octave rolloff that's sharp by any standard. Crossover points can easily be changed to suit different performances. The VFX-2 also works as a bandpass filter, or for tri-amping a mono system.

Hum and noise 113dB below rated output (IHF), IM distortion less than 0.01%, 19 inch rack mount.

Try a VFX-2 on your next installation. Be sure.

When listening becomes an art, crown
Box 1000, Elkhart IN 46514

Circle 17 on Reader Service Card

FREE LITERATURE

CONSUMER FRAUD BROADCASTS
A circular describes 2½ minute broadcasts under the overall topic "Have You Been Taken Lately?" available to broadcasters. Mfr: National Broadcasters Group.
Circle 94 on R. S. Card.

CITIZEN'S BAND CHANNELS
A pocket-sized pamphlet lists most of the popular Citizen Band channels used on major U.S. highways. Mfr: Siltronix.
Circle 96 on R.S. Card.

INVERTER SCRS
Two new 250 amp rms fast-switching inverter scrs are described in this data sheet. Mfr: International Rectifier.
Circle 97 on R.S. Card.

ELECTRONIC EYELETS
28 questions regarding electronic eyelets are answered in this information sheet. Mfr: International Eyelets, Inc.
Circle 98 on R.S. Card.

TELEPHONE COUPLING TRANSFORMERS
New telephone coupling transformers designed for interconnect to the nationwide telephone network under the FCC Part 68 Registration Program are described in this brochure. Mfr: Microtran Co.
Circle 99 on R.S. Card.

ENGINEERING SEMINARS
A 6-page brochure gives details of three-day nationwide sound engineering training seminars. Mfr: Synergetic Audio Concepts.
Circle 80 on R.S. Card.

EXTENSION SPEAKERS
Application and technical information regarding six models of environment-resistant loudspeakers is contained in this data sheet. Mfr: Atlas Sound.
Circle 81 on R.S. Card.

RAM PROGRAMMER/CONTROLLERS
An extensive booklet details specifications of digital timers. Mfr: ESE
Circle 82 on R.S. Card.

HORNS, SPEAKERS, DRIVERS
Horns, speakers, drivers, and sound columns with complete specifications are described in a six-page condensed catalog. Mfr: University Sound.
Circle 83 on R.S. Card.

WIRE AND CABLE
Specs, illustrations and applications for wire and cable, insulation, cord-sets, and electronic accessory products are covered in a 48-page catalog. Mfr: Associated Graphic Arts.
Circle 84 on R.S. Card.

CALIBRATING DEVICES
This technical paper reviews the various methods for calibrating accelerometers, microphones, and hydrophones. Mfr: B. & K. Instruments, Inc.
Circle 85 on R.S. Card.

INSULATED WIRE AND CABLE
A data sheet includes description, qualifications, performance characteristics, photographs and specifications of Maser wire and cable insulated products. Mfr: Addington Laboratories, Inc.
Circle 95 on R.S. Card.

PHASE RESPONSE
A six-page engineering application reports on "Phase Response Characteristics of a Butterworth Filter." The report contains illustrations and graphs for calculating gain and phase shift through a high-pass, low-pass, bandpass or band-reject filter. Mfr: Krohn-Hite.
Circle 87 on R.S. Card.

The Sensual Equalizer.

Whether on record or in live performance, today's most commercially successful music is more visceral, immediate, and sensual than ever before. This impact has been achieved through advances in the musician's art, and through a quantum jump in the control available in audio processing.

The Orban/Parasound Parametric Equalizer, Model 621, has received outstanding acceptance since its introduction because it combines economy ($369/channel) with extraordinary control. Each of its four non-interacting bands permits continuous, stepless adjustment of bandwidth, equalization, and center frequency. Each band can be tuned over a 20:1 frequency range with no change in curve shape (unlike some competitors), and peak gain remains constant as the bandwidth is varied. The unique "constant-Q" equalization characteristic is more musical than the usual reciprocal curves, and lets the equalizer create infinite-depth dips to remove hum, whistles and ring modes—making it ideal for cinema and sound reinforcement as well as recording studio and broadcast applications. Other outstandingly useful features include a front-panel gain control and a peak-stretching overload lamp which indicates clipping anywhere in the equalizer circuitry.

While our spec sheet (available from the address below) gives the details in cold black-and-white, it cannot describe the sensual interaction between man and machine which occurs when the frustrating limitations of conventional equalizers are finally overcome, and the user is given the power to create sound that feels really right. Our ability to deliver this power at an affordable price is the true reason for the O/P Parametric's success. But don't take our word for it—discover the Sensual Equalizer for yourself, soon.

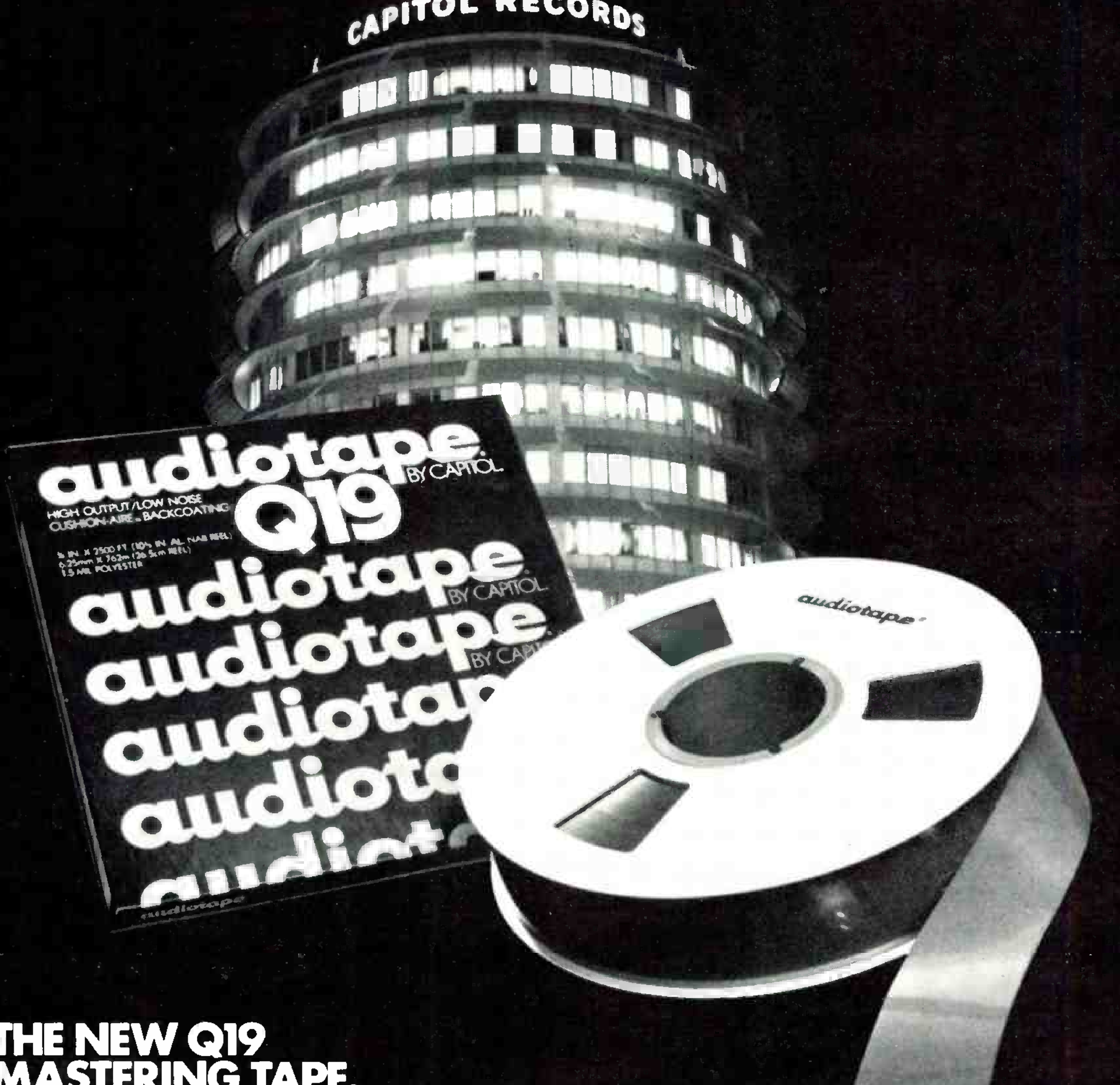
For further information, contact

orban/parasound
680 Beach St.
San Francisco Ca. 94109
Or contact your local
Orban/Parasound distributor

Write for free new paper: "How to Choose Equalizers for Professional Recording Applications"

Circle 18 on Reader Service Card

Introducing a new formula for successful recordings.



THE NEW Q19 MASTERING TAPE.

At any demanding recording session there is no room for any error. That's why the NEW Q19 Mastering Tape is used at Capitol Records' studios. We are the only studio tape manufacturer in the United States that also operates a major studio. Therefore, we know what is required in a quality mastering tape, and we have designed and manufactured FORMULA 19 from a user's viewpoint.

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

These features are just a few of the many reasons why studios are using FORMULA 19. It is the most sophisticated mastering tape we've ever developed. Try it . . . and discover the new formula for success.

AUDIOTAPE® A PROFESSIONAL PRODUCT MANUFACTURED BY THE SAME PEOPLE WHO MAKE AUDIODISCS,® AUDIOFILM,™ AUDIOPAK® BROADCAST CARTRIDGES, AND THE LEARNING TAPE™ BY CAPITOL®

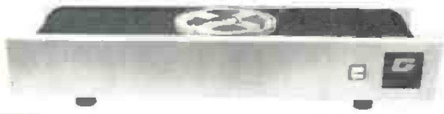
audiotape® BY CAPITOL.

Capitol™ **CAPITOL MAGNETIC PRODUCTS** A DIVISION OF CAPITOL RECORDS, INC. 1750 NORTH VINE STREET, LOS ANGELES, CALIFORNIA 90028

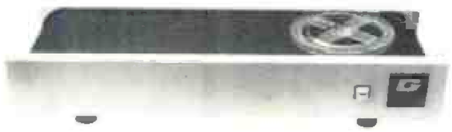
Circle 19 on Reader Service Card

Erase faster

Erase cleaner



Erase easier



Garner Model 70 cuts man-hours spent erasing audio and video tapes. Simple, safe continuous belt operation gives you "hands-off" professional erasures in only four seconds. Handles up to 7" reels, cartridges, and cassettes. Acclaimed by major users, yet priced low enough for the smallest studio or station to afford.

GARNER INDUSTRIES
4200 N. 48th St.
Lincoln, NE 68504
402-464-5911

Circle 20 on Reader Service Card

CALENDAR

MAY

25-27 **B&K Seminar:** Human Acoustics. Contact: B&K Instruments, 5111 W. 164th St., Cleveland, Ohio 44142. (216) 267-4800.

27 **Mobile Radio Market Seminar.** New York City. Contact: Frost & Sullivan, 106 Fulton St., New York, N.Y. 10038. (212) 233-1080.

25-27 **Synergetic Audio Concepts Training Session, Chicago, Ill.** Contact: Don Davis, P.O. Box 1134, Tustin, Ca. 92680. (714) 838-2288.

28-31 **Sound and Vision '76.** Birmingham, England.

JUNE

7-11 **B & K Seminar: Industrial Noise Control.** Contact: B&K Instruments, 5111 W. 164th St., Cleveland, Ohio 44142. (216) 267-4800.

8-10 **Synergetic Audio Concepts Training Session.** Columbus, Ohio. Contact: Don Davis, P.O. Box 1134, Tustin, Ca. 92680. (714) 838-2288.

8-11 **Communications '76.** Brighton, England. Contact: British Information Services, 845 Third Ave., New York, N.Y. 10022. (212) 752-8400.

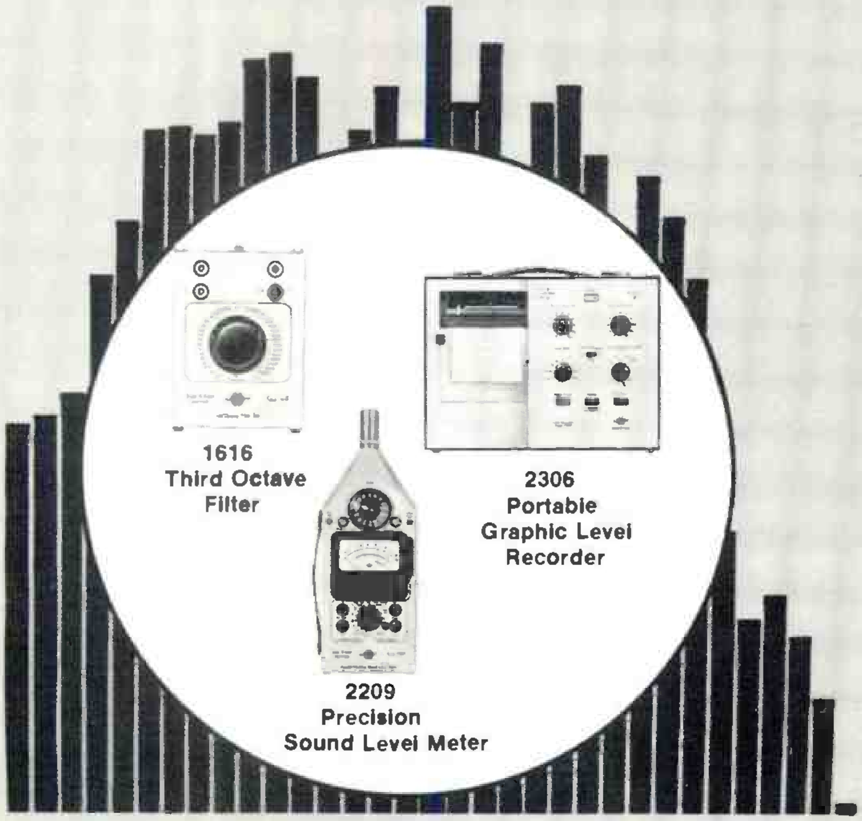
8-11 **Information Retrieval Exhibition, London, England.** Contact: British Information Services.

8-11 **International Audio-Visual Aids, London, England.** Contact: British Information Services.

21-July 9 **Audio Recording Technology Workshop.** Brigham Young University. Contact: Special Courses & Conferences, Brigham Young University, Audio Recording Technology Course, 242 HRCB, Provo, Utah 84602.

Copies of all issues of **db—The Sound Engineering Magazine** starting with the November 1967 issue are now available on 35 mm. microfilm. For further information or to place your order please write directly to: **University Microfilm, Inc.**
300 North Zeeb Road
Ann Arbor, Michigan 48106

All 1/3 Octave Levels Are Not Recreated Equal!!!



That is why professionals equalize recording studios, auditoria and special masking installations. But the job of equalization must not be left to your ears alone for they are neither calibrated nor qualified for sensing frequency-amplitude unbalance. Therefore, precision acoustic instrumentation must be used.

Equalizing procedures require portable instruments which can be powered internally. In addition, the equipment must be ruggedly constructed to withstand the abuse of field environments. Who has the equipment with a proven track record in the portability field? B & K has and we're waiting to prove it to you with our 2209 Precision Sound Level Meter, 1616 Third Octave Filter Set, and 2306 Level Recorder for hard copy proof of equalizing. Call or write today for complete details, at 5111 West 164th Street, Cleveland, OH 44142 (216) 267-4800.

B&K Instruments, Inc.
Bruel & Kjaer Precision Instruments

Circle 21 on Reader Service Card

db theory&practice

NORMAN H. CROWHURST

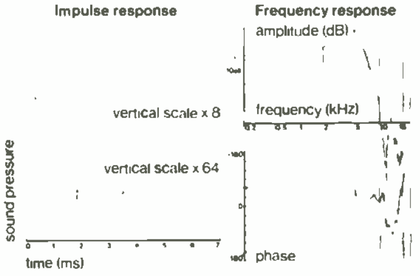


Figure 1. Cumulative decay spectra.

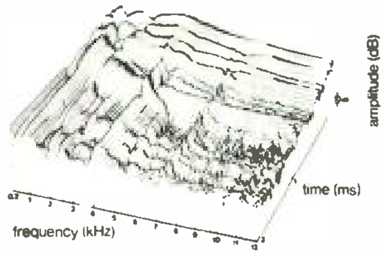


Figure 2. 110 mm. moving coil bass/ mid-range unit in a 7 liter closed box.

● In last month's column, we talked about loudspeaker testing and briefly outlined the problems of taking a meaningful frequency response, indicative of how the unit sounds. The traditional method analyzes the performance of the unit in terms of amplitude and phase, measured against frequency, as the independent variable.

There are two things a little unrealistic about this, both related to the fact that a sine waveform is used. First, in everyday life, a sine wave is not representative of most of the sounds we hear. Secondly, by using a sine wave, we commit ourselves to a purity of that waveform, any departure from which represents distortion.

Now, it is true that any departure from the pure sine wave does introduce components that audibly alter the sound. And, unless the spurious frequencies have a precise harmonic relationship to the test frequency, a quite small departure from the original sine waveform makes an audible difference.

The main problem with using a sine wave for testing loudspeakers in their natural acoustic environment (and nobody listens to program in an anechoic room, does he?) is that a pure sine wave must also have duration, for the response to it to be measurable.

This is one of the problems that those who have tried to present a frequency response as a repetitive display will have encountered. If you hold frequencies down at the lower end, say, 20 hertz, for long enough to enable the system to produce a reading at that frequency, you will take up considerable time taking these readings. True, the higher frequencies can be read more quickly. But the time necessary to read the low frequency response makes presentation in this form a slow business.

That is not all. When you hold each frequency long enough, or change the frequency slowly enough, to get good readings, you will also be sustaining the sine wave radiated long enough to build up standing waves, which make the result spurious for a different reason.

Some practical program material may have notes that last long enough to allow standing waves to form. But the way we hear the program is more apt to resolve the sound differently from that. What we hear is the initial impact of the waves, while any build-up of standing waves tends to be at least partially disregarded, as part of the reverberation that is characteristic of the listening room.

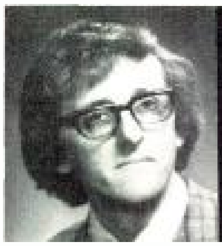
TIME DOMAIN

While it is true that our hearing faculty does seem to analyze the sounds we hear on a frequency basis, it also seems that the response is quite different from, say a wave analyzer. Our hearing also analyzes what we hear in the time domain, so it can more readily separate direct sound waves from reverberant build-up.

The human hearing faculty is a highly sophisticated system! Because it relies much more heavily on time-domain information than any electronic wave analyzer so far built, many have sought to find a way to use time domain testing of loudspeakers.

As any mathematician can tell you, a waveform can be analyzed in one of two ways. Either it can be resolved into a number of frequency components, of varying amplitude and phase, or else it can be analyzed in terms of just varying amplitude against time. This can then, perhaps, be reduced to a succession of time constant growth and decay curves.

Number 99 in a series of discussions by Electro-Voice engineers.



HORNS OF PLENTY

D. (DON) B. KEELE, JR.
Senior Engineer
Loudspeakers

A few years ago, if you were to ask most sound practitioners questions about horn design, each answer would probably be much like the next. A set of assumptions about the value of hyperbolic and exponential horns had taken full root. And products available to the field reflected this unanimity of opinion. Certain shapes and sizes were "best" and differences in horn design were almost exclusively related to materials and minor variations in loading plug and throat characteristics.

Recently, however, intensive restudy of basic horn shape options has led to some new conclusions at E-V. The study aided in the design of new horns (patent pending) with performance differences that can provide meaningful improvements in sound quality if properly utilized.

One study concentrated on determining optimum horn mouth size. Popular belief suggested that "the larger the better" but our studies revealed this to be untrue. When horn mouth size is optimized, internal reflections are significantly reduced compared to both larger and smaller horns. The result is higher acoustic efficiency, especially near and even below the cutoff frequency.

Horn flares also underwent intensive study, with some fascinating results. It was found that the polar lobes typical of most exponential and multicell horns could be virtually eliminated with a multi-flare conical section horn. The conical flare rate is doubled near the mouth to eliminate mid-range beaming typical of exponential horns. The conical flares are combined with an exponential flare near the throat to maintain good response down to horn cutoff. In this new class of horns, constant response throughout the radiation angle, plus uniform beam width at all frequencies is achieved without the complex structure and loading plug problems required by multicell types.

With this new class of horns, plus appropriate drivers, the sound industry enjoys the use of more precise tools for the control of radiated sound. The products assist in more predictable system design, lessen installation problems, and meet the highest performance standards. And the concept has been proven in such recent installations as Pontiac Stadium, Disney World, and Yankee Stadium among others.

This continual challenge of basic assumptions has long been a tradition at E-V. From the unique CDP® dual-horn speaker—still valid decades later—to the current technology of the Sentry® speaker, our products reflect a willingness to upset favorite notions when they stand in the way of industry progress.

Electro-Voice® INC.
a gulton company

Dept. 563BD, 686 Cecil Street
Buchanan, Michigan 49107

Circle 22 on Reader Service Card

Any particular waveform, however complex, will have a unique representation in each form. Thus it would theoretically be possible to translate one to the other, either way. But as we have said before in this column, while a video waveform is vitally dependent on its shape—its amplitude variation in the time domain—an audio waveform has always been believed not to matter, as far as its shape is concerned, so much as its precise frequency content. Are we now saying something else?

No, not exactly. In the sense that shape was unimportant to the sound of the wave, provided the frequency content is unchanged, that is still true. The main trouble is, we tend to think in "either/or" terms. If phase doesn't matter, then why bother with it? Phase, in the order that can occur along the propagation line of a wave, cannot matter, because the waveshape will vary at every point, measured spacially along the wave. At no two points, will the wave have the same shape.

FREQUENCY CONTENT

But it will have the same frequency content—or should have—instant by instant. If a plucked tone, for instance, starts very harmonically rich, and settles into a tone with dominance of fundamental and lower harmonics, this time varying relationship between the frequency content will also be unchanged along the propagation line of the wave, although the waveshape will not be the same at any two points.

The point to realize is that the waveshape at every point must be one possible variation of the same time-varying frequency content. Thus although, in one sense, phase will vary, and in varying, does not change the characteristic of the sound, in another sense, the content of the wave, as regards frequencies present, and in what amplitude, does not vary, at any instant, along the wave.

To use a simple example, much simpler than you would ever encounter in an actual program, suppose a sound consists of fundamental and second harmonic, both of which start suddenly, at the same instant. Where the wave originates, let us say, both waves start up from the zero line at the same instant.

The fundamental will travel at the speed of sound, as will the harmonic. In quarter of the fundamental's period, the harmonic will have moved through half its period. So the phase will now be different. But at all points along the path of the outgoing wave, the two will start at the same instant in time, regardless of the momentary phase relationship.

After the initial wavefront moves out, two kinds of thing can happen. The source of sound may change its composition, so that the two component frequencies do not decay at the same rate. Or, standing waves can build up, due to the acoustic environment of the room.

If you place a single pressure microphone at some point in the room, it is not easy to separate these effects. You pick up a composite wave that has varying amplitudes of the component frequencies due to both causes. But your hearing faculty can tell the difference. It can virtually ignore the variations caused by the build-up of reverberant energy, standing waves, while being quite conscious of the

We in the West welcome

Challenge

Sparta was asked, "Can you make a console which does more, in less space, better, than you've ever built before?"

Our response was the Western double-barrel above — the 3000-Series consoles!

Both the Models 3310 Mono and 3410 Stereo Consoles are all-new, with state-of-the-art DC switching, interchangeable input cards, 4-1/2" illuminated VU meters, larger control knobs on precision step attenuators, and more flexibility than any console except the Sparta Centurions!

They do more . . . in less space . . . better . . . less expensively than any other broadcast console!

SPARTA
Division of Cetec Corporation
5851 Florin-Perkins Road, Sacramento, Ca. 95828 (916) 383-5353

Circle 40 on Reader Service Card

savEnergy

changing composition of the direct sound.

It seems fairly obvious that our judgment of loudspeaker performance, as opposed to its measured response, is based on the apparent realism with which it can convey to us the content of the original program.

What we have been saying about modification of the content of the wave with time and space, could apply equally well to sound from an original performance, say an orchestra or a small group, or to reproduction of such sound by a loudspeaker. Our impression of realism depends on how well the second replicates the first, *as judged by our hearing*. That is the catch.

FUNCTION OF THE LOUDSPEAKER

The loudspeaker can do some of the same things to sound that the other two causes can. In short then, it can modify either the apparent character of the original sound source, or the effect of reverberation or, more likely, both. And virtually none of this capability is taken into account when you take the frequency response of a loudspeaker in an anechoic room. If its average output of energy, by the time it has reached a steady state at each frequency measured, is the same at all frequencies measured, it looks like a good, "flat" loudspeaker.

On actual material, our ears may tell a quite different story. This has been realized for a long time. The first attempt to do something about it was square wave testing. Next came tone burst testing. Both seemed to have some validity for testing amplifiers, but became almost meaningless, when a transducer link, such as a loudspeaker, was included in the test.

DIGITAL TESTING

Now another form of testing has come into the picture and initial work with it shows great promise. This is the application of digital techniques. We show here, courtesy of KEF Electronics, of Maidstone, England, a comparison of three forms of test on the same unit.

At top left is a plot of decay spectra to an impulse signal. The response is successfully amplified, to get more detail of the "tail." At top right are amplitude and phase responses of the same unit. Then below is a 3-dimensional plot of cumulative decay spectra. You will see that this gives much more information than any of the other forms of presentation.

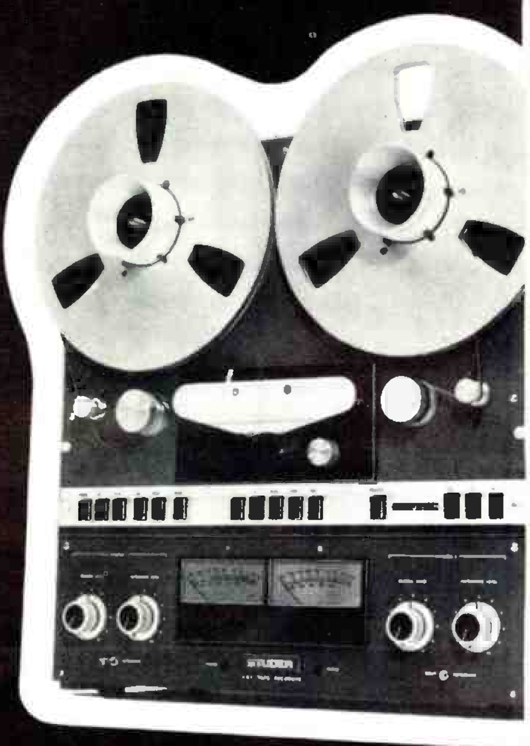
The important thing is that it gives information on both a frequency and time base, frequency being shown from left to right, and time from back to front, the response being shown upward. It gets closer to telling us something about how it will sound, because it uses a computer, with Fourier transforms, to generate the information from data obtained from some 500 impulses, at intervals long enough to allow the response to die down about 60 dB.

Anyone conversant with this kind of measurement will know that measuring response over an amplitude range of 60 dB is going to run into noise problems. The average anechoic room is not good enough. A better place is outdoors, preferably at the top of a high tower, and in a locality well away from airports and other sources of noise.

Then, of course, the prevailing source of noise will be the birds singing! Whoever would call that noise? But, seriously, averaging over some 500 impulses will reduce incidence of bird sounds at any particular point on the decay curve, by at least 54 dB, which means that quite meaningful results can be obtained, in spite of the birds! ■

A67... the NEW STUDER

'portable professional'



Now from STUDER comes a versatile and compact studio mastering recorder for under $3,000.

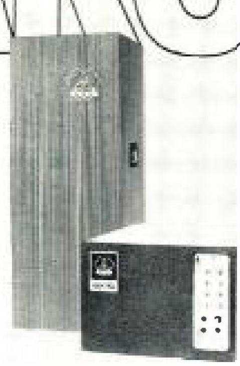
The A67 Tape Transport contains many features found on the world famous STUDER A80 which provides for optimum tape handling ability in all modes of operation.

The plug-in amplifiers feature contemporary design and contain individual record, bias, and reproduce adjustments for all three speeds.

For further information, contact Willi STUDER America Inc., 1819 Broadway, Nashville, Tennessee 37203. Phone 615-329-9576. Telex 55-4453. *In Canada*, STUDER REVOX Canada Ltd., phone 416-423-2831. Telex 06-23310.

Circle 24 on Reader Service Card

AKG



Reverberation Units with AKG-exclusive Torsional Transmission Line Principle for super natural performance.

• Superior performance in the studio: AKG BX-20E Studio Reverberation Unit

• Superior performance wherever you need it: AKG BX-10E Portable Reverberation Unit

The BX-20E Studio Reverberation Unit incorporates AKG's Torsional Transmission Line Principle for superior, *natural* reverberation that exceeds the critical needs of studio technique. It is entirely free from coloration, flutter echoes and similar disturbances.

The BX-10E is the first *truly* portable reverb unit designed to provide the quality and operating features required in studio applications. Utilizing the AKG TTL Principle, it incorporates many design features of the larger BX-20E Unit.

Both BX-20E and BX-10E allow adjustable decay time of each channel independently. Moreover, TTL is the only artificial reverb device—including live chambers—which does not contain any of the dry input signal at its output. Decay time is adjusted silently through motional feedback, allowing dynamic adjustment, even while recording.

And there's more. Like the suspension system that makes both units impervious to acoustic feedback and mechanical vibrations. Get the full facts from your AKG dealer!

At selected dealers everywhere. Or write.

REVERBERATION UNITS
HEADPHONES
CARTRIDGES
MICROPHONES

AKG ACOUSTICS
PHILIPS AUDIO VIDEO SYSTEMS CORP.
Audio Division, 91 McKee Drive
Mahwah, New Jersey 07430 (201) 529-5900

Circle 25 on Reader Service Card

db the sync track

JOHN M. WORAM

• For those who were fortunate enough to miss the October 1975 *Sync Track*, I offered some ill-chosen words about some of the less-than-glorious guitar amplifiers that sometimes show up in the recording studio. When the amp is *really* gross, the engineer may suggest — perhaps in desperation — "What about using an acoustic guitar?" That snappy bit of snide humor prompted the following letter from Dennis Melton of MDM Communications in Wilmington, Delaware.

I am writing to comment on your discussion, What About Guitar Amps? *Your first deduction "What about using an acoustic guitar?" really threw me. You must realize that a musician usually chooses the type of instrument carefully and for artistic reasons, whether it sounds dirty or not. Very few musicians wish to change the sound they've worked hard to get, on the day of recording. I agree that grounding problems should be investigated first, but let's not alter the musician's sound by asking him to use a direct box or an acoustic guitar. My suggestion would be to use a noise gate, and the engineer's talents with the fader during the mix. Also, in many cases, the guitar and other instruments on the tape will mask the hum.*

I feel that musicians using electronic equipment should be educated to the ways of getting a clean sound, for they can certainly distort it, beginning with a clean sound. The reverse rarely works.

I would appreciate any feedback on my comments.

I've really got mixed emotions about what to say. After almost every other sentence, I want to remark, "You're right, but . . ." For instance, about that musician who chooses his instrument carefully and for artistic reasons. I think there's a double negative tucked away in there somewhere. If he chose it thusly, it wouldn't sound dirty. Or at least not all the time. With all the special effects controls at rest, the instrument should sound clean, clean, clean. If it doesn't, the artist should buy another axe, or get the one he owns fixed. Then, and only then, he should come into the studio and—with all the fuzz, wah-wah, tremolo and reverb he can carry—create his "sound." But he should begin with a decent instrument, one that hums, buzzes and whistles *only* on command. The perpetually dirty instrument has no more excuse for existence than the dirty console or tape recorder, and its user has no right to call himself an artist.

WHEN AN AMP GOES SOUR

Of course, this is the real world, and sometimes an amplifier goes sour without warning. It can happen to a musician or to an engineer. The pro makes the best of the situation by whatever means are available, and practical. This might mean using a direct box, putting in a noise gate, or gain riding. But in this context, all of these devices are crutches, used in an emergency situation. When the engineer is forced to use them to salvage an otherwise intolerable sound, there'll be neither time nor equipment left for creative recording.

"Very few musicians wish to change the sound they've worked hard to get, on the day of recording." That's true enough, *if* the musician has only one sound in his repertoire. However, the true artist uses his ears to evaluate his sound in context with everyone else's, and is ready (and able) to change it when that becomes necessary. This may be for an artistic reason or because his amplifier goes beserk. In the latter case, I think the pro would opt for an acoustic guitar rather than hope that the guitar and other instruments on the tape will mask the hum. As for getting the best sound possible under normal circumstances, Mr. Melton sums it all up by saying that musicians should be educated to the ways of getting a clean sound. For that matter, so should engineers.

AND SO TO LITERATURE

Speaking of education, I've got another book to review (How's *that* for a smooth transition?)

HOW TO BE A RECORDING ENGINEER, by Phil York. Attainment Research Books, $6.95.

If that title doesn't raise your suspicions, you're much too gullible. For surely, it must take more than $6.95 to become a recording engineer. My mind wanders to other titles in an imaginary series of "How to" books:

Be a Brain Surgeon	$13.50
Be a Crooked President	$ 5.98
Be a Millionaire	$27.50
Be a Nuclear Physicist	$ 9.75

But you're a trusting soul, so you order the book. Surprise! It's a pamphlet of 28 pages, stapled into a blue paper cover. The cover itself counts for four of those 28 pages.

Inside, the information ranges from misleading to interesting to outrageous, with a little comic relief here and there. For instance,

I can tell you with certainty that its not necessary to spend years "getting into" the business . . . I've personally tutored intelligent candidates from scratch—less than a year later they were working full time turning out records and jingles. It could be done in a couple of months, but who has the time or facilities for that?

Long ago, before Bob Newhart married Suzanne Pleshette and became a psychologist, he used to do a monologue about putting an infinite number of monkeys in front of an infinite number of typewriters. The idea was that, due to the law of averages, sooner or later one of them would turn out some great literature. Sure enough, one day one of the monkeys wrote, "To be or not to be, that is the gezuornigplatx"

That was the end of the skit. But now that Newhart is finished with that chimp, let's give him a contract to write a monthly column for **db Magazine.** What are the odds that he'll ever turn out anything worth reading again?

Of course, people are smarter than monkeys (sometimes), so let's now take an infinite number of eager beginners and put them in front on an infinite number of consoles. We'll also buy each one of them a lottery ticket. Sure enough, one of them will eventually turn out a hit record, and one will win a lot of money. But no one will hail the lottery winner as a financial genius. Neither was that monkey a great writer. And how would you rate that eager beginner who happened to be in the control room the day a hit record was born? If you want to think of him as a great engineer, go right ahead.

If you want to become a recording engineer, Mr. York advises, *"If you are starting from near scratch and have almost no experience with audio equipment, the very first thing to do is to buy a very good electronic or audio dictionary."* Also, it is handy to have played an instrument and done some wiring and soldering.

I suppose he's right. There are many who have gotten launched in just such a way. But I think those who are seeking guidance in entering such a relatively small and highly competitive field as recording shouldn't be misled into thinking that all they need is a book, a soldering iron, and some music lessons.

There is a five page section entitled *How I Entered*, in which York describes his personal experiences since he first visited a small studio in 1959. It's very interesting, and the author's good humor shows through, but it's got nothing to do with how to be a recording engineer.

Next come some interviews with a few recording engineers. Here's a little sample:

Q. *Did you go to any of the trade schools that are available for this?*

A. *No, I was lucky enough to get the experience without going to any schools. That's what the schools try to give you isn't it?*

Engineer S. F. describes his natural gifts; *I can hear things nobody else can . . . I can pick out something and figure out how they did it in the studio . . . I know every EQ setting on the console just by listening to the record.*

No comment.

In the section, *Setting up for a Session*, Mr. York offers this bit of advice, *There is a good reason to store your tape tails out . . . take my word for it. Besides, if you don't, many people will think you don't know what you're doing.*

In the section on schools, he says, *In case you want to gain some skills before attacking the job problem, you may want to check the following schools. The first one has branches all over the U.S.A.*

Institute of Audio Research
64 University Place
New York, N. Y. 10003

Well, the address is right, but everything else isn't. Skill is the ability to do something well. You con't get that in school; not at the Institute of Audio Research nor at any other school. You *might* get an education, but you'll have to develop the skills yourself, after graduation. Also, the school with branches all over the U.S.A. is the Recording Institute of America, not the I. of A.R.

For an informative book on what microphones to use where, I recommend "Microphones: Design and Application."

Wrong again! Burroughs' microphone book doesn't have a word to say about what microphone to use where. In fact, he took very great pains to avoid any mention of specific microphones, since that aspect of microphone usage is entirely up to the taste of the user.

And finally, the correct address for Billboard Publications' *International Directory of Recording Studios* is 1515 Broadway, New York, N.Y. 10036. (It costs $10.00)

Ironically, there is some good advice interspersed in all of the above nonsense. In fact, I suspect that Phil York is indeed well-intentioned, but that Attainment Research needs a good copy editor. Although, at $6.95, it makes one wonder! ■

The Stax SRX-III is quite simply the finest sound reproduction system you can buy. Period.

Much better than any speaker system at any price. Literally the standard for "other" headphone manufacturers.

Compare the facts you can measure: flat frequency response from 25 - 30,000 Hz (± 1.5 dB); distortion is essentially unmeasurable.

Now compare the facts that really matter. The SRX-III is entirely hand assembled and evaluated by the family company who created the first electrostatic headphone. A company dedicated to research — to ultimate sound. So every SRX is the product of detailed effort. Even the low mass diaphrams are individually paired — both by electronic testing and by listening.

Listen to an SRX system with program material you "know". You will be in the front row — the same room — the recording studio. You will hear perfection. The truest, clearest, most transparent reproduction ever possible. A reality now at audio dealers nationwide.

STAX

Backed by **American Audioport, Inc.**
909 University
Columbia, Mo. 65201

Circle 26 on Reader Service Card

DON'T MIX THE GOOD SOUND WITH ANYTHING BUT THE BEST....................

THE
MODEL 6440 STEREO AUDIO CONTROL MIXER

Micro-Trak's 6440 Professional Audio Control/Mixer; The mixer to choose as the standard for your Disco installations. With its basic two turntable, two microphone, and three auxiliary inputs coupled to four mixing channels, it makes the ideal answer for the discotheque, theater, or broadcast production requirement.

Totally professional from design...to application of the highest quality components...to its heritage from a company that's been providing broadcasters with the finest audio equipment for over thirty years, the 6440 has it all.....

Smooth pots...FET audio switching Panel laid out for the Pro...Metered output levels adequate to drive almost any power amp...Inbuilt cue amplifier and speaker...noise so low its virtually not there...Built to take all you can give it and still come up swinging . . . .

See your dealer for specifications or write directly to our marketing department.

MICRO-TRAK CORPORATION

620 RACE, HOLYOKE, MASSACHUSETTS USA

Circle 27 on Reader Service Card

db sound with images

MARTIN DICKSTEIN

• If you expected to read something in this column this month about video tape editing as indicated in last month's "Things to Come" . . . April Fool. (Well, almost!) When work was first begun on this column, after a short hiatus last month, the material was going to include information on video tape editing and video projection with some details on a few of the pieces of equipment that have become available in the recent past. The subject became voluminous in material and will be included in a near future column. Meanwhile, so many other things have been happening, also worthy of your awareness, that it might be better to catch up on a thing or two. Since it's spring, how about video fun for the home?

For instance, you might already be familiar with the latest from Sony—the Betamax system for home t.v. recording.

Betamax was launched some time ago when Sony decided that the U-Matic system with which they had all but inundated industry, education, and training facilities would not reach into homes. That market seemed to need something different. The video disc was a hot item and just about ready for marketing, but Sony's idea was to stay with tape.

Two other giants had recently tried to introduce a video tape system for the home—Sears, with Cartravision, and RCA with MagTape. You may recall that the Sears system was a complete package with the deck and t.v. set in one cabinet. This meant that the owner of a color t.v. set had to buy another one in order to have the deck. At the time, the electronic system in the tape deck was not compatible with standard t.v. sets and required its own special unit to play video material. Whether or not for this reason, there was non-acceptance by the public, and Sears decided not to continue with the unit. This all came about shortly after Sony started pushing its ¾-in.-U machines which could stand alone and feed into almost any standard color set.

After its first venture with SelectaVision, a film type of material, RCA began work on the MagTape system which would utilize magnetic tape, the up-and-coming medium, made popular with the realization that it was necessary for the public to be able to record as well as play back previously recorded material. After much development and investment, RCA decided not to continue with this approach, and to return to a playback-only type of home device. They felt that the tape unit pricing would probably have to be higher than anticipated, and went to work on their video disc instead.

Sony, with its know-how in development and marketing, decided that this was the time to get its product into the home. The system they developed used, until now, a total package with the tape recorder/player and t.v. set in one cabinet. The impact of advertising was to impress on potential owners that they no longer had to miss any program that was on the air at the same time as another favorite show, or that went on when the viewer would not be home to watch. By incorporating two tuners, the owner could watch one program while the other was being recorded for future viewing, or, by using the unit's timer, a not-to-be-missed telecast could be recorded while no one was home to watch. (Of course, this also provided the person who owned a recorder the opportunity to watch a program and record it at the same time just in case a repeat performance might be desired.)

BETAMAX

As with the ¾-in.-U cassette system, Sony was again innovative with its Betamax. In the full console model, the unit was provided with a camera input, a microphone input for high impedance, and two cassettes. (When the unit was first demonstrated, there was no camera input.) The unit also is furnished with an earphone, an antenna splitter, two 75 ohm coax cables, and a 300 ohm to 75 ohm matching transformer. The system has a video s/n ratio of better than 40 dB and an audio s/n ratio better than 43 dB. The audio frequency response is given as 50 Hz to 10 kHz. Resolution horizontally is more than 280 lines monochrome and more than 240 lines in color. The t.v. set is Sony's 19 in. Trinitron. Cost is a bit over $2,000.

Don't let the fact that a cassette is used fool you into thinking that this means that the system is similar to the U-matic ¾-in. system. The tape in the Betamax is ½ in. wide. This makes the cassette smaller, which is better for handling and storing. However, the previous ½-in systems conformed with existing systems using the same width tape. EIA-J standards, to which the previous ½-in system conformed, runs the tape at 7½ in/sec. Betamax uses a speed of 4.0 cm/sec., equivalent to about 1.6 in/sec. At present, the maximum recording time in a Betamax cassette is 60 minutes. (Incidentally, the model number for the Betamax is LV-1901.)

The complete Betamax console was first put on the market in New York at the end of last year, then was distributed elsewhere. At that time, since the system was being touted as a two-program device (view one, record the other), there was no mention of a free-standing unit or already-recorded cassettes. Things have changed since then. Early this year, there emerged a possibility of a separate player machine, and there was talk of a pre-recorded tape. By the time you read this, the player unit may be on the market and plans may be far along on getting movies and other programs into production. By working out deals with other manufacturers, Sony may get the Betamax to be the "standard" for the home as it did for the U-system for industry and education. The unit will also, no doubt, get attention from those users of video as well, for applications where the double-program idea can be beneficial.

As a semi-final note (there will be more detail in the near future), a camera is also being offered as an accessory to the Betamax system—model AVC 1420. It is a black-and-white unit, made to go with the Betamax. There is also the possibility that in the very near future there will be a cassette available which will be longer than the present 60 minutes in length—perhaps up to 2 hours.

T.V. GAMES

In the past year, the public was given an opportunity, or two, or three, to use the t.v. set for something other than to watch programming that was not to their liking. A device could be purchased that could be hooked into the antenna terminals that would provide the owner with a game he (or she) might enjoy better. Similar to the stand-alone type found in game areas of amusement parks, these devices offered several choices, each usually requiring the competition between two players, or at least two hands.

Among the games were ping-pong, tennis, and auto racing. Then there were others with planes or sharks. Prices for the devices ranged around $100. At the end of the year, color games were introduced. Most major department stores offer one type or another, and the number of manufacturers is growing in leaps and bounds, or is it pings-and-pongs?

This year, there will be several new game developments. It is expected that the simple games will remain but will fade in novelty and importance. The next generation will introduce computer action. This offers opportunities for more sophisticated games like chess, word games, number or math games, and the ultimate situation, for the present, anyway, an opportunity to program the computer to play games of the owner's own choosing and even with changing rules. Possibilities of this diversion seem limitless. There's even the chance that the consumer will be able to communicate with a large computer by phone line to get desired information, perhaps for projects the customer can do himself from instructions on a readout screen.

MICROPROCESSOR

The new word in the home game field is *microprocessor*. Through the use of the multi-function chip, the more usual games of hockey, football, handball, and squash will be joined by the solitaire feature, as it is called—a provision built in for the user to play against the machine itself. The device will analyze moves, counter against all steps taken by the just-human player, and in full color, too, and the player can even set the level of expertise at which the machine will play. In games where a certain amount of speed is required, like handball, the device will include a variable speed control. (Well, the human has to be able to win somehow, doesn't he?) Depending on the sophistication of the game and the device, the pricing will be from somewhere above $100 upwards to close to $1,000. (Some of the devices will also include the sound of the ball being hit, on-screen score keeping, and even a remote control unit.)

Pricing will go down on the less complex games as a result of improved manufacturing practice, development of less expensive material, and the entry into the field of toy manufacturers, who always seem to know how to introduce new "toys" for less than they cost when they were called games.

So, this year, you have a number of choices, for home amusement—you can watch two programs that are on at the same time (watch one now, one later) play a pre-recorded tape or make your own programs, or use your old t.v. set for the amusement of your own selection and programming. Think of all the fun you've just been introduced to. With several t.v. sets, several different games, a few of the Betamax units—drive you crazy, couldn't it? No fooling! ■

Rauland

PURE PRO

audio design

The new standard in Professional Sound offers you everything:

Spectrum-Master Equalization

Most complete line available: ⅓-Octave Equalizer/Test Set: two additional ⅓ and 1-Octave Equalizers; a unique Tunable Notch Filter; a versatile Equalization Test Set.

Spectrum-Master In-Wall Amplifiers

There is nothing to equal these professional units, each with built-in 1-Octave Equalizer and exclusive Dynamic Range Extender. Available in 35-watt, 60-watt, 100-watt outputs.

Spectrum Master Amplifiers

Imcomparable DX and TAX Solid-state amplifiers, designed for optimum continuous-duty performance. Available in a broad selection from 70 to 250 watts RMS to meet any professional audio requirement.

Spectrum-Master Mixer-Amplifiers

Superior 4400 Series with less than 1.5% THD and program equalization provisions. Flexible, advanced design; professional in every sense—for the most demanding applications.

Spectrum-Master Input Equipment

Have optimum mixing performance with maximum flexibility in the 4900 Series. Ideal for broadcast and recording use, theatres, auditoriums, and churches. Distinguished for ultra-low distortion and wide-range.

Rauland *quality's other name in Sound and Communications*

WRITE FOR TECHNICAL BULLETINS

RAULAND-BORG CORPORATION

3535 W. Addison St., Dept. N., Chicago, Ill. 60618

Circle 29 on Reader Service Card

Optimizing the Remote Pickup Audio

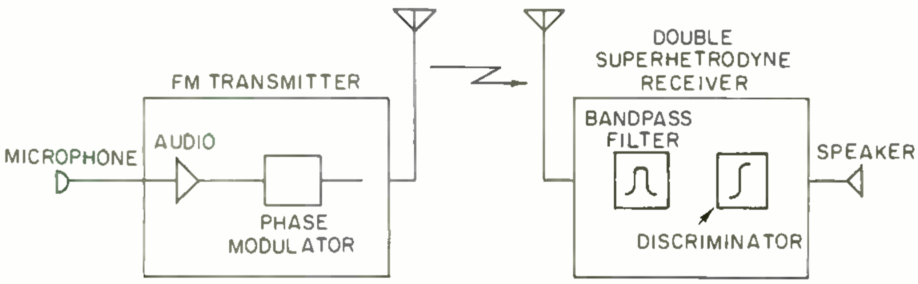


Figure 1. The basic system.

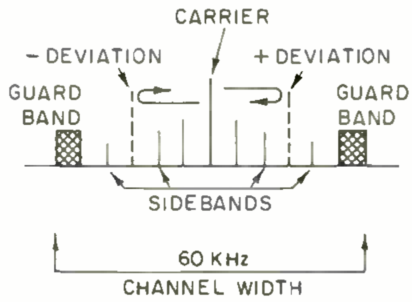


Figure 2. A typical remote pickup channel.

• Many broadcast stations use small portable radio transmitters for on-the-spot news reports and similar broadcasts. The type of unit used in most cases, is the small two-way radio system designed for voice communications. The audio specifications for these small systems may seem rather limited as compared to our large broadcast facilities, but they do provide a good voice channel, and this is quite adequate for the purpose. There are many natural as well as regulatory limitations which can either directly or indirectly affect the audio which can be recovered from these systems. In this column, we will discuss a few of the important areas which can directly effect the system's audio.

BASIC SYSTEM

The portable transmitter does not broadcast directly to the public, but instead, to a receiver back at the studios, where the audio is demodulated, and then coupled to the station's regular facilities. The system then, is made up of both a transmitter and a receiver. At the remote site, there is a transmitter and an antenna. Audio is picked up by the system's microphone and used to modulate the transmitter. The modulation process is usually achieved by phase modulation so that direct crystal control of the carrier may be maintained. The signal is an f.m. signal.

Transmitters may be licensed in many bands, from short wave to UHF. Very few channels (let alone bands) are allocated specifically for remote pickup use. Generally, the remote pickup shares the band with industrial or safety services. The particular band in use will have its own characteristics that will limit the distance the signal will travel and contribute noise and other factors to the recovered audio.

The transmitting antenna is usually at a low height, so to make up for this, the receiving antenna will be mounted high above the ground and connected to the receiver with a co-axial transmission line. The receiver is usually a double superheterodyne type (2 i.f. frequencies) and will contain a bandpass filter. This filter is intended for rejection of adjacent channel interference, but it will also automatically limit the maximum bandpass of the system. Limiters are used for noise suppression and detection is effected by a discriminator. The recovered audio passes through a de-emphasis network and on to the receiver's local speaker.

There are several areas in the system where the audio can be directly affected. The main ones are the microphone, the modulation process and bandwidth, receiver bandpass filter, demodulation process and then the interface of the audio to the station's facilities.

The system bandwidth is determined by the equipment capabilities and by FCC rules. The permitted bandwidth varies with different bands, and in some cases, with specific channels within a band.

BANDWIDTH

The bandwidth is based on a mathematical formula: $B_w = 2D + 2M \times K$. D = the deviation on one side of the carrier; M = the highest audio modulating frequency; K = a constant that, for this service, is usually 1. Narrowband telephony, for example, is permitted a deviation of 5 kHz, and the highest audio frequency is 3 kHz. Thus: 2 (5 kHz) + 2 (3 kHz) = 16 kHz. Adding the FCC emission designator F3, this becomes 16F3. This means: the bandwidth is 16 kHz, the modulation is f.m., and the operation is telephony (voice). A wideband communications channel is permitted a deviation of 15 kHz. Using this in the formula: 2 (15) + 2 (3) = 36 kHz, or 36F3.

Since the bandwidth is the product of the deviation and the audio modulating frequency, the actual bandwidth may be higher or lower in operation than the licensed value. All the emissions (including all the sidebands created during modulation) must fit within the channel. So, if you are attempt-

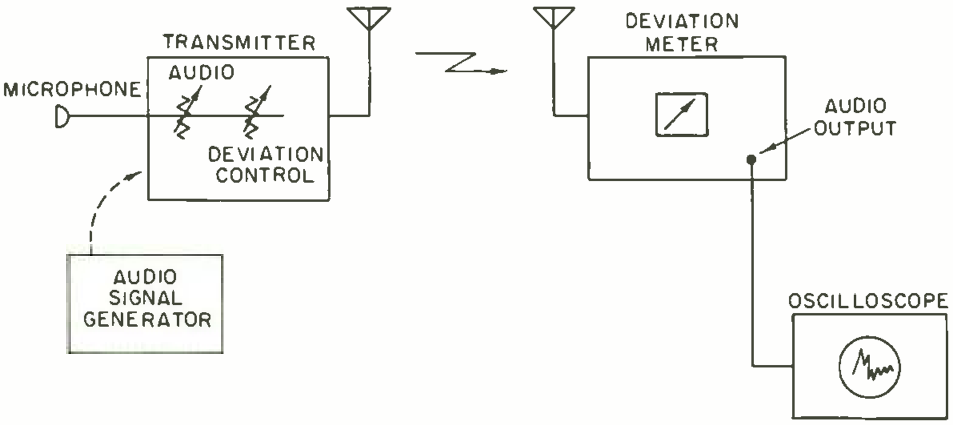


Figure 3. The setup to measure and adjust a transmitter deviation.

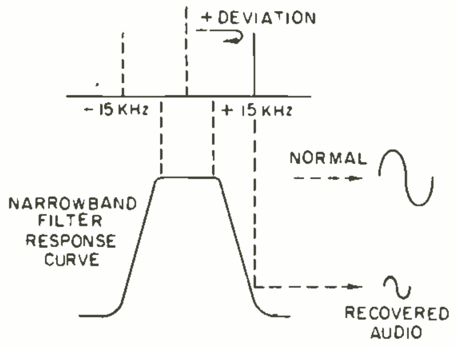


Figure 4. The narrowband filter in the receiver can reduce the output of a wideband system.

ing to extend the audio bandpass to 5 kHz instead of 3 kHz, the actual bandwidth can exceed the licensed channel width. For example, the audio is now 5 kHz and the deviation is adjusted for 15 kHz. Thus, 2 (15) + (2 (5) = 40 kHz. If the licensed bandwidth is 60 kHz, you are okay, but if it is 25 kHz, then the signal is out of channel. It would be necessary to reduce the deviation to 7.5 kHz to accommodate 5 kHz in your channel. Thus: 2 (7.5) + 2 (5) = 25 kHz.

AUDIO INPUT

The microphone has a direct bearing upon the quality of the audio. The usual microphone supplied with these units is designed to produce audio on a speaker which can cut through noise at the receiver location. This type of mic does not provide the best quality for broadcast purposes; select one of the better grade microphones that are available for these systems. This will be either a variable reluctance or dynamic-type microphone and may have its own transistorized preamp inside its case.

Internally, there will be one or two audio stages (in the transmitter) for amplification, and then a speech clipper. This clipper is required to provide a "brute force" limit on transmitter deviation by clipping off any excess audio peaks. This clipper needs proper adjustment, and of course, the audio should be kept below the point where clipping will occur or there will be distortion.

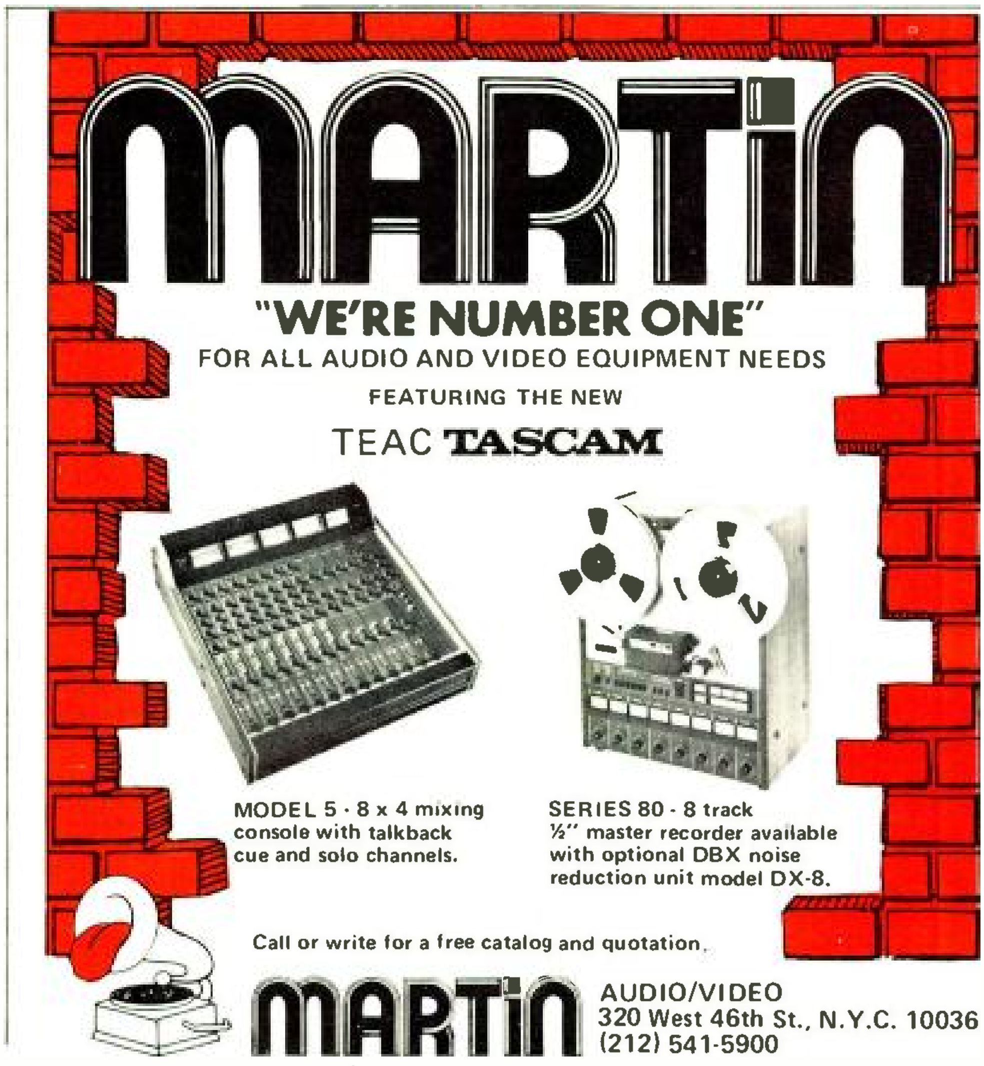
DEVIATION

The deviation of the carrier has a direct effect on the recovered audio and its quality. If the deviation is too low, then the recovered audio will be low and the system signal-to-noise ratio will suffer. Should the deviation be set too high, the system bandwidth can be exceeded and there can be emissions outside the channel. Emissions out of the channel can cause adjacent channel interference to other stations.

The system bandwidth can be exceeded in two ways. In the first case, the entire system has been tuned too narrowly and the bandpass filter in the receiver is for narrowband. This narrow bandpass of the system will have a filtering effect on the signal, so that the reproduced audio can be low, have a poor response curve, or be distorted. It all depends upon the actual conditions at the time. In the second case, the modulator and the modulated stage may be called upon to deviate for more than it has the capability. This can cause non-linearity of the audio signal or outright clipping. Both will produce distortion. The deviation is a very important factor in audio quality, so it should be adjusted carefully and properly.

ADJUSTING DEVIATION

The actual amount of deviation should be measured; using a deviation meter is the best method. This is a test instrument that serves in the same capacity as the station's modulation



Circle 30 on Reader Service Card

"G" LINE
Offers Full-Range Components For
Professional Sound Reinforcement

Preamplifiers-Mixers

G5TA • G12T • G11T
Versatile, expandable, top-notch mixing components, perfect for large areas such as auditoriums and churches, as well as recording, broadcast, and other commercial sound applications. Ultra-low distortion, full-range performance.

Power-Amplifiers
50, 100, 200 RMS WATTS

G52 • G102 • G200
Tops in continuous performance **reliability** for serious professional audio usage.

POWER-AMPLIFIERS
35, 50, 100 RMS WATTS

GT-35 • GT-50 • GT-100
Choose from a **flexible** line of solid state integrated units that will control, mix, and amplify four low impedance microphones and two auxiliary inputs. Satisfies even the most critical demands.

Background Music at Its Best

G-109B
Special **high performance** FM-AM monaural tuners can be used with any amplifier, since volume control is located on front panel.

To Order, Call Toll Free,
800-323-0228
or, write for complete information today.

Precision Electronics, Inc.
9101 King Street • Franklin Park, Illinois 60131
In Canada, Superior Electronics
In Caribbean, E. D. Magnus, Chicago, Ill.
Foreign, Morhan Exports, New York, N.Y.

Circle 31 on Reader Service Card

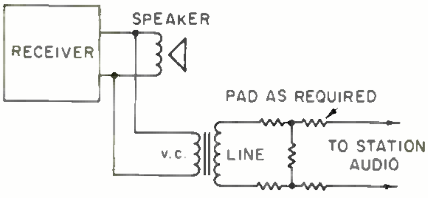


Figure 5. A line-to-voice coil transformer can be used in reverse to isolate and match the receiver to the station's audio system.

monitor. Deviation can also be measured by a communications receiver or a heterodyne frequency meter and the tone modulation of the carrier, although this is a more cumbersome method. Tone modulation is applied to the carrier and the deviation adjusted while listening for the carrier nulls. When the correct null occurs, then deviation is correct. The deviation meter is the better arrangement; an oscilloscope attached to its audio output can be useful.

Make the preliminary adjustment, using 1 kHz tone modulation of the carrier. This will allow getting all the adjustments within the ballpark and the oscilloscope, at the same time, can observe if there is non-linearity or clipping by the modulator. The speech clipper should be adjusted out of the way at this time so that it does not enter the considerations.

Once the preliminary adjustments have been made, connect up the regular microphone that will be used with this transmitter. The signal peaks for voice are 8 to 10 dB higher than sine wave peaks, so the final adjustments should always be made with voice transmissions. If the adjustments are made only with tone, these voice peaks can be driving the unit into severe distortion. Speak into the microphone at normal announcer delivery levels. Adjust the audio gain control and the deviation for the correct amount. It is best also to pull back the audio control because these peaks may be high enough to cause distortion in the audio stages.

When the correct deviation is observed on peaks, then observe the oscilloscope that is viewing the audio after the deviation meter. If any of the peaks are clipped, then back off the deviation or the audio control until the peaks are clean. Those clipped peaks mean that either the audio or the modulator is being overloaded and causing clipping. When the deviation has been set to the maximum undistorted amount, adjust the speech clipper to begin to clip at that deviation

Dub faster

Dub easier

Garner Model 1056 updates your dubbing operation. Five 1200′ professional copies in four minutes. Threads fast. Rewinds in 60 seconds. Single capstan drive and solid state electronics guarantee unvarying high quality. Priced low enough for quick payout. Write for brochure and names of users.

GARNER INDUSTRIES
4200 North 48th St.
Lincoln, NE 68504
Phone 402—464-5911

Circle 34 on Reader Service Card

A COMPACT WITH ALL THE OPTIONS

The ROBINS/FAIRCHILD Model 659A, REVERBERTRON, is a complete dynamic reverberation system designed to enhance Broadcast/Production or Recording Studio sound.
With solid state electronics and high performance electro-mechanical delay lines in two separate enclosures, the REVERBERTRON is compact in size yet loaded with features. Continuous reverb mix controls, VU metering, selectable decay times, 3 band equalization and remote control . . . to name a few. All in 7" of rack space.
REVERBERTRON! Designed to fit your rack and priced to fit your budget.
FOR COMPLETE DETAILS, CALL OR WRITE SAM JONES.

ROBINS FAIRCHILD
A Robins Industries Corporation
75 Austin Boulevard, Commack, N.Y. 11725
(516) 543-5200

Circle 39 on Reader Service Card

level. Leave the settings in these positions.

Another important aspect is microphone technique. Since there are no operating audio gain controls, announcers should practice their mic technique until they can determine the correct distance to hold the mic from their mouths for their normal delivery levels. They should develop a technique that will provide strong, full deviation, that is just below the clipping point.

RECEIVER

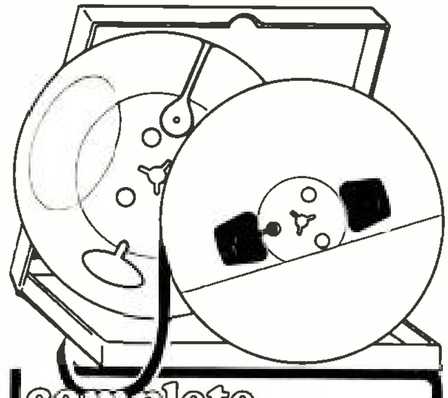
The bandpass filter is intended to reject adjacent channel interference, but it will also effect the system bandpass. If the filter is for a narrowband systems and you are trying to operate wideband, then the filter will remove most of the wideband deviation you so painstakingly coaxed into the transmitter. If you plan to operate wideband, then order the receiver with a wideband filter.

The discriminator response curve should be centered exactly on carrier (which should be right down the center of the entire system bandpass). During receiver alignment, these adjustments will have been made with a signal generator. But the final adjustment of the discriminator should be made with the carrier itself and modulating with voice. Tweak up the adjustments so the reproduced voice sounds good. This is only a touchup adjustment, so don't overdo it.

The industrial unit is designed only to supply audio to its own speaker. For remote pickup use, this audio must be coupled to the regular system audio. Unless this interfacing is done properly, the audio can be deteriorated and hum can be introduced. The best method is through a line-to-voice-coil transformer. The transformer will provide both a match and isolation. Attach the voice coil side to the speaker output of the unit and the 600 ohm side to the system. So the regular speaker can operate at normal volume, add a pad on the line side of the transformer to reduce the signal level to system requirements.

SUMMARY

The small, industrial-type communications transmitter systems can be used for remote pickup use, and the results will be satisfactory for voice broadcasts. But there are many possibilities where the audio can be deteriorated, so the units must be tuned and adjusted properly, especially the deviation, and the interfacing to the station's audio system done with care. ■



complete supplies for
reel to reel audio recording

TAPE AMPEX and Scotch all professional grades on reels or on hubs

EMPTY REELS all sizes, widths and hub types

BOXES for all reels, in various colors

LEADER-, TIMING-& SPLICING TAPES

Top Quality Competitive Pricing
Immediate Shipment
Call or Write for our Catalog

recording supply corp.
1291 RAND RD -DES PLAINES, IL 60016
312/297-0955

Circle 32 on Reader Service Card

You oughta have your head examined.

And your pinchwheel inspected.

And your clutch and brake checked.

In fact, if you depend on your Nagra for your living, a periodic check-up with Jerry Ozment (The Nagra Specialist) will bring every gear and gizmo under his scrutiny.

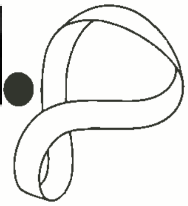
Jerry has lived and breathed Nagras for the past ten years as both a repair technician and motion picture sound man.

His ability to modify and adapt standard Nagras to specific needs for unusual situations is startling.

His repair skills have set a standard in professional circles.

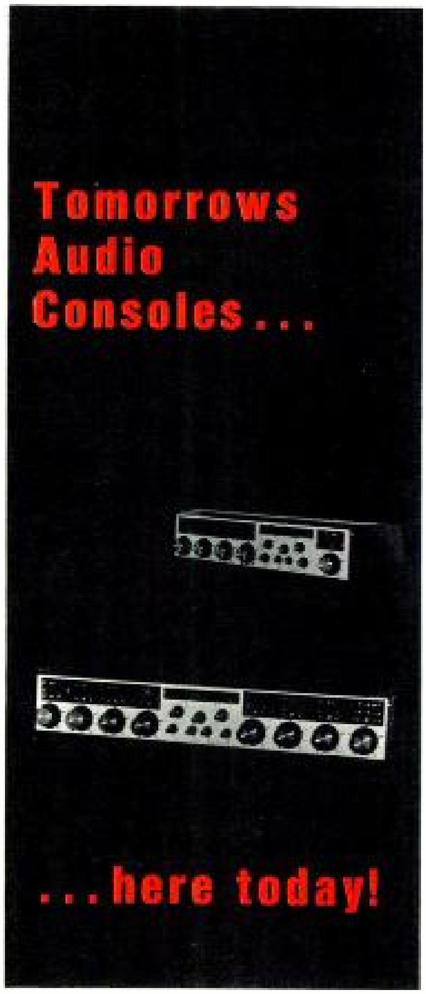
And, of course, all modifications and repairs are fully guaranteed by Mobius Cine, Ltd., the home of the Nagra Specialist.

The Nagra Specialist at Mobius Cine Ltd.



7 East 47th Street, New York, N.Y. 10017 212 758-3770

Circle 33 on Reader Service Card



The biggest advance of audio control in the last 15 years.

Totally DC controlled for noiseless switching and audio mixing. Lighted touch pad switching eliminates mechanical noise and breakdown. Advanced solid state light emitting "VU" meters. Cermet mixers and level controls for years of trouble free operation. Plug in amplifier cards. Full range input gain select from mic thru high level. All inputs and outputs balanced. Distortion — 0/3%; Response — +0, –2 db, 20 Hz - 20 KHz; Noise — –65 db (mic inputs). Flexibility? Complete complement of accessories for input expansion, equalization, remote control, etc.

10 day free trial and 2 year warranty.

Call collect or write today. You'll find it both an exciting and profitable adventure!

Models & Prices

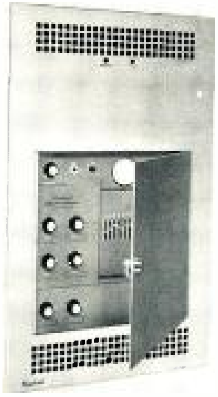
SC-5M Single Channel, mono	$ 605
DC-5M Dual Channel, mono	$ 742
DC-5MS Dual Channel, stereo	$ 979
DC-8M Dual Channel, mono	$1,199
DC-8MS Dual Channel, stereo	$1,760

RAMKO RESEARCH
3516 C LaGrande Blvd.
Sacramento, California 95823
Telephone (916) 392-2100

Circle 42 on Reader Service Card

db new products & services

IN-WALL AMPLIFIERS

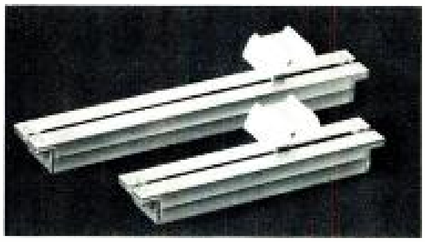


• Spectrum-Master series in-wall amplifiers offer a choice of three units: 35W, 60W, and 100W rms. All units have a built-in one-octave equalizer and a dynamic range extender, which it is claimed will eliminate microphone overloads. Included are four microphone inputs, expandable to 6; microphone precedence; remote volume control facilities; optional volume compressor; computer-grade electrolytics; key-locked side-opening door.
Mfr: Rauland-Borg Corp.
Circle 50 on Reader Service Card

SIGNAL-TO-NOISE SQUELCH

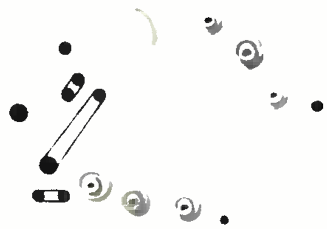
• A miniature signal analysis device which constantly monitors the content of a channel and determines if information or noise is present is the basis of Sound Off self-compensating signal-to-noise squelch. Sound Off operates directly on the audio signal, can be placed anywhere in the audio line and will squelch the signal whenever other information is removed. The unit, which contains its own a.c. power supply, compensates for changes in atmospheric noise and is insensitive to false triggering by impulse or other noise.
Mfr: Kahn Communications, Inc.
Circle 51 on Reader Service Card

SLIM LINE FADERS

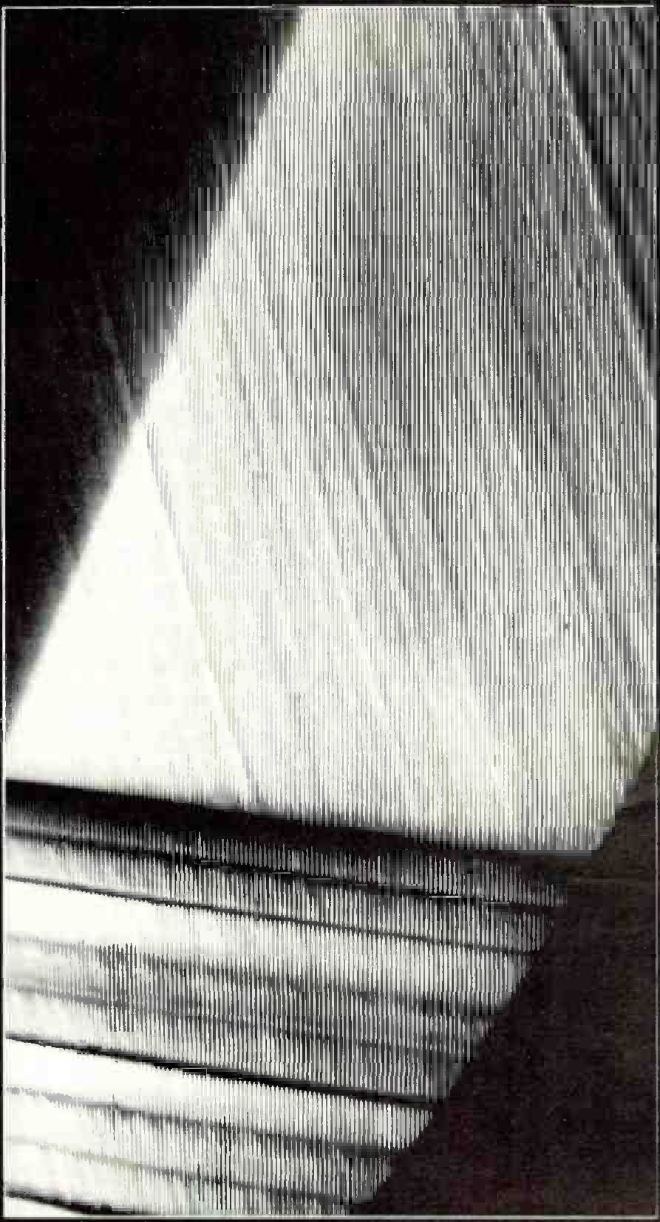


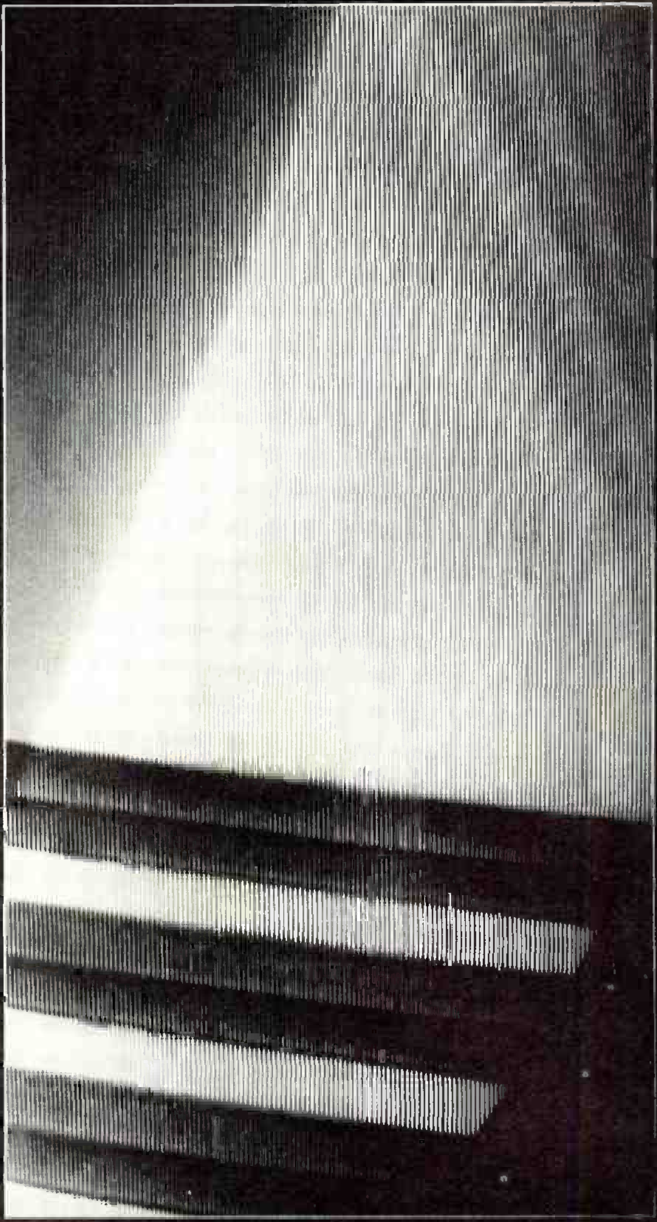
• Twice as many controls can be accommodated in the same space through the use of slim MM-4 and MM-6 conductive plastic faders. Units can be mounted close together on ¾-in. centers with two units taking up 1½ in. of space. The attenuators are available in linear taper, modified audio taper, or logarithmic. The units contain metal moving contacts and extrusions, including aluminum, and gold plate. A touch control knob is optionally available.
Mfr: Waters Mfg. Co.
Circle 52 on Reader Service Card

ELASTOMER BELTS



• Synthetic-woven belts used for motor driven machines, such as cassettes where pulleys are cut with a 90 degree groove, have an elastic quality. Materials vary to suit chemicals present, temperatures, etc. The belts can range from 1 to 45 in. in circumference; widths from 0.01 in. to 2 in. The belts return to their original shape when stretched and are resistant to abrasion, acids, solvents, water absorption, oxidation, ozone, sunlight aging, heat and cold. They can be solid cast, or made from cast foam or from elastic foam, or can be made by injection molding and shaped to the desired size.
Mfr: Butler Precision Belting
Circle 53 on Reader Service Card


Master from Transco Products Corporation


Master from Audiodiscs® by Capitol®

Your eyes are now a cutting stylus.

And with Audiodiscs® they've just cut a perfect master.

The Audiodiscs® master disc by Capitol is magnified 200 times as is the master from Transco Products Corporation. As this close-up reveals, there is a remarkable difference between the two. Even more exciting is the fact that these differences didn't exist 6 months ago.

What caused this remarkable superiority? Since early 1975 Audiodiscs by Capitol have been manufactured in the newest and most advanced disc plant in the world. All aspects connected with the manufacturing process, including cleanliness, have been optimized to produce a superior master disc. With absolutely no compromise. This is apparent in the coating process where the exacting application of lacquer ensures a smoother disc.

At this point, look at the pictures again... they speak louder than words.

For a free sample and brochure, write on your company letterhead, or call Harry Preston at (213) 462-6252.

AUDIODISCS® A PROFESSIONAL PRODUCT MANUFACTURED BY THE SAME PEOPLE THAT MAKE AUDIOTAPE® AUDIOFILM,™ AUDIOPAK® BROADCAST CARTRIDGES, AND THE LEARNING TAPE™ BY CAPITOL.

audiodiscs
BY CAPITOL

Capitol CAPITOL MAGNETIC PRODUCTS A DIVISION OF CAPITOL RECORDS INC 1750 NORTH VINE STREET LOS ANGELES CALIFORNIA 90028

Circle 44 on Reader Service Card

Ampex MM-1200. A

Twenty-four tracks of dynamic audio, on two-inch tape. Twenty-four tracks of drums, cymbals, wailing clarinets, groaning electric bases, pianos, synthesizers, sopranos, castanets and tambourines, all doing their stuff in splendid isolation. That's what you get with the new Ampex MM-1200.

It took years of give-and-take between top recording engineers and Ampex design engineers to achieve the MM-1200. The long pathway from a gadgeteer's dream to a multichannel machine you can take for granted has reached an endpoint.

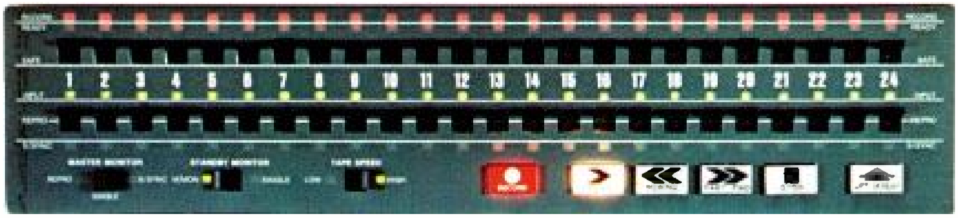
There's a strong technical story to tell about the MM-1200. You'll find many new features that mean better mechanical service, more reliable electronic performance, and significant savings for your studio in production time and effort.

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.

Small points, but important.

The MM-1200 is a rugged machine, with lifetime lubrication on all moving parts. The master ON/OFF switch is protected against accidental operation. Cabinet "bumpers" assure enough wall or corner clearance for adequate ventilation. The VU meters tilt out, and the faces are non-glare.

More than a recording machine.

An Ampex MM-1200 in your studio is a powerful statement of your professional capabilities. Producers know the fidelity you can capture with an MM-1200, and they'll want it for every session. It's an investment that can help you sell more studio time.

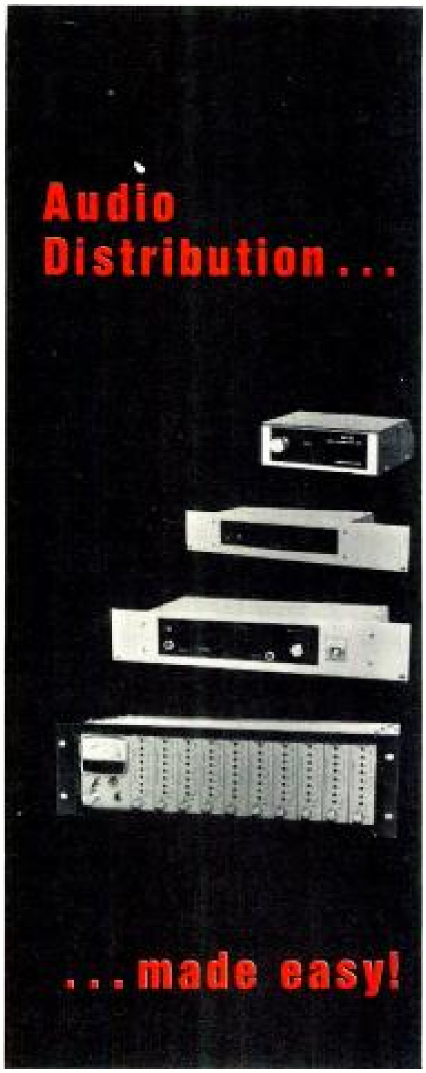
Specifications.

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

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

Circle 36 on Reader Service Card



Six different audio DA's designed to solve all of your distribution problems.

From our table top 1 in/6 out to our powerful 20 in/80 out. Stereo or mono operation, output metering, individual level controls and balanced inputs and outputs are just a few of the many features found in these superb DA's. Performance? Response — 10 Hz - 20 KHz ±0.5 db; Dist. — 0.1%; Output level — +20 dbm max; Signal/Noise — –90 db; Channel separation — 80 db. Quality? All RAMKO products are backed by our 10 day free trial and 2 year warranty. They have to be good to do that.

Call collect or write today!

Models & Prices

DA-6/E 1x6 (table top)	$ 143
DA-6R/E 1x6 (rack)	$ 165
DA-6BR/E 1x6 (rack, indiv. cont.)	$ 179
DA-6RS/E 2x12 (rack)	$ 239
DA-16BR/E 2x16 (rack, meter, etc.)	$ 295
DA-2080 up to 20x80 (rack)	$325 - $1,675

RAMKO RESEARCH
3516 C LaGrande Blvd.
Sacramento, California 98523
Telephone (916) 392-2100

Circle 43 on Reader Service Card

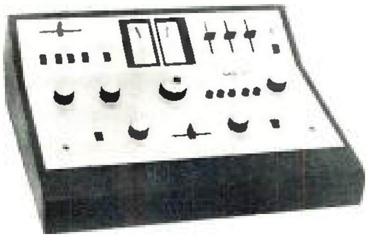
MATCHING AMPLIFIER/ PREAMPLIFIER

• Two tube components, a monophonic power amplifier, model MB-3045 and matching stereo preamplifier, model CL-35 111 are available. The amplifier features a new triode tube, claimed to make possible a high-power, low-distortion triode effect. MB-3045 delivers a minimum of 50 watts continuous power into 4, 8, or 16 ohms. Model CL-35 111 stereophonic preamplifier is rated at 0.6 per cent harmonic distortion at 2.0 volts, 20 to 20,000 Hz. It features six selectable turnover frequencies, high and low noise filters, switchable phono input impedances, and variable input sensitivities.
Mfr: Lux Audio of America.
Price: MB-3045, $445.
CL-35 111, $745.
Circle 54 on Reader Service Card

DELAY & REVERBERATION SYSTEM

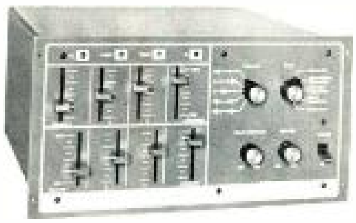
• Totally electronic SD-50 system is built around charge-coupled devices (ccd) or bucket brigade large scale integrated circuits. Incoming signals are pre-emphasized and compressed for noise reduction, then enter the ccd circuits, where a front panel control provides continuously variable delay times from 5 to 50 milliseconds by adjusting the duration time in each bucket. After expansion and de-emphasis, the signals are available to feed any existing rear channel amplifier/speaker system. The SD-50 also provides a continuously reverberation level control to simulate the multiple reflections from walls and ceilings that characterize real concert halls.
Mfr: Sound Concepts, Inc.
Price: $600.00
Circle 55 on Reader Service Card

MIXER/PREAMPLIFIER

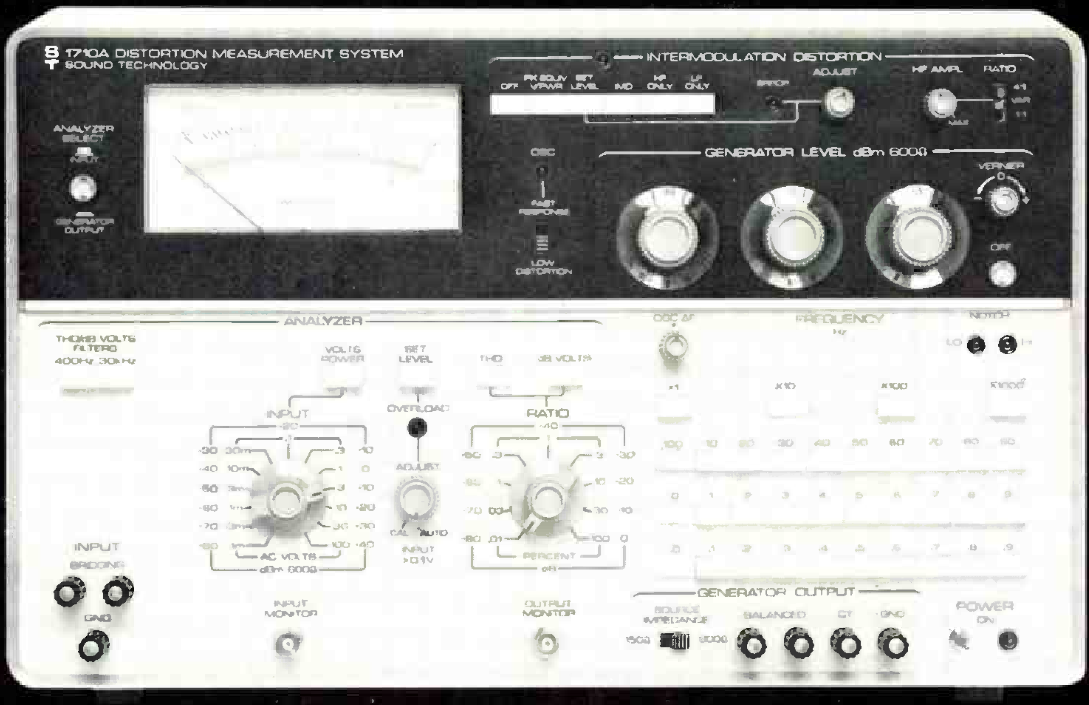


• Primarily designed for discotheque use, model 5880 mixer/preamp is sophisticated enough for a range of other applications. The two primary inputs accept either phono or high-level stereo signals and are connected to a slide control for gradual fading from one program to another. A universal impedance microphone input feeds both stereo channels; an auxiliary stereo input services high-level sound sources. Each input has an individual level control, with a master level control and master stereo balance control. Cueing control permits headphone previews; vu meters show the strength of the program being played and the one being previewed so their levels can be matched. An "RG" peak unlimiter/downward expander, a 3-band equalizer and two sets of monitor jacks are included.
Mfr: GLI, Inc.
Price: $600.00.
Circle 56 on Reader Service Card

LIGHTING CONTROLLER



• Four-channel Sonalite Four controller offers 1,200 watts per channel at 120 volts a.c. Each channel is individually adjustable together with auto switched progression, auto fade progression, sound progression, and sound sync. It has selectable sequence control with five pattern ability. Each channel has its own emphasis control. Led indicator lamps are of a different color for each channel; a master brightness control enables dimmer controls to be set separately to background for all dynamic modes. Level control is automatic. Up to five extender units can be cascaded (2 kW each per channel) together with an optional manual keyboard. The manual over-ride button can be played in any sequence or duration.
Mfr: Meteor Light & Sound Co.
Price: $1,275.
Circle 57 on Reader Service Card



Your new automatic distortion measuring system for balanced measurements

REDUCED OPERATOR ERROR

Here's something you'll like — Sound Tech's new distortion measuring instrument for use in balanced work.

The new 1710A is much more than just a distortion analyzer. It's a system. It contains its own ultra-low-distortion generator tracked with the analyzer. It's a system that greatly simplifies measuring — gives you fast measuring with simple operation that reduces operator error.

For example, push the frequency buttons and you set both generator and analyzer. Push "Distortion" and you have your reading. Automatically. No slow, tedious manual null-searching.

Features in the new 1710A include:

- **a balanced, floating output (600/150 ohms)**
- **a balanced (bridging) input**
- **a high-level +26 dBm signal**
- **+26 to −90 dBm attenuator**
- **distortion measurements to .002%**
- **fast 5-second measuring speed**
- **automatic nulling, optional automatic set level.**
- **both harmonic and optional intermodulation distortion measurements.**

SPECIAL OUTPUT CIRCUIT

In the 1710A you get a transformerless audio generator output that's balanced and floating. No transformer means no transformer distortion. Floating *and* balanced means you can connect to virtually any audio circuit regardless of configuration. And you can set the output from +26 to −90 dBm in 0.1 dB steps.

FAST, SIMPLE MEASURING

Automatic nulling and the automatic set level option (ASL) give you extremely fast measuring and little chance for operator error. You can measure in 5 or 6 seconds. *With ASL you can measure distortion vs. frequency, and distortion vs. voltage or power without resetting level.*

IM OPTION

An additional optional bonus is that the 1710A also measures intermodulation distortion. After you've made a harmonic measurement, just push the "IMD" button. In 3 seconds you'll have the IM reading. With this option you'll be ready for future IM requirements.

CALL/SEND NOW FOR LITERATURE

It's worth while getting the information on this major new distortion measuring system. Call Larry Maguire or Bob Andersen now and get our new product brochure. It's ready and waiting.

SOUND TECHNOLOGY
1400 DELL AVENUE
CAMPBELL, CALIFORNIA 95008
(408) 378-6540

Circle 45 on Reader Service Card

ROBERT E. BERGLAS

Digital Clocks & Things

Electrical impluses work through a series of switching computerized aggregations to display the flashing signals of time. Save money by assembling your own digital timer.

THE DAYS of the circular dialed clock and wristwatch with the pointer pointing somewhere or other—never quite on the nose—are numbered due to recent advances in readout devices and lsi (large scale integration) semiconductors. The era may well nigh be drawing close when all means of telling time will be digital and crystal controlled accuracy will be as taken for granted as it was elusive yesterday.

This discussion will look at some of the various readout devices available as well as some of the "mini-computer" insides that do the actual calculations in digital clocks and wristwatches.

READOUTS

Probably the oldest available readout devices are the *Nixie*™ tubes. Every number from zero to nine in a Nixie has its own cathode wire; the tube requires about 170 Vdc supply voltage, putting something of a strain on the power supply transformer. There are a few more drawbacks to the Nixie tubes, however. First of all, the viewing angle is very limited either side of head-on. Secondly, these tubes deteriorate markedly with age. Finally, since each number has its own cathode, the number being viewed also illuminates some of the other wires.

Available for a few years now, are the RCA *Numitron* tubes. They operate off the standard 5 volt logic supply, are easily plugged into ordinary sockets, and their brightness can be modulated. One of their nicest features—and a feature that few readout devices can accomplish—is that they can be filtered to just about any color the designer wishes. On the negative side, Numitrons require 108mW/segment, so the power supply components and transformers must be conservatively rated.

Probably the most recent types of readouts are the *Sperry Gas Discharge* tubes, now marketed by Beckman. These produce a striking, bright orange (they cannot be filtered) color that is very "clean"—by that I mean the segments are solid and fuse nicely into one another to make up a number. Furthermore, Sperry tubes are relatively large, and require very little current, albeit they require at least 180 volts polarizing supply. The new Heath digital clocks, for example, use Sperry tubes.

LIGHT EMITTING DIODES

When quality, not cost, is the deciding factor, most designers choose light emitting diodes (leds). They are available in either common anode or common cathode formats, and can be had today in either standard ruby red, green, or yellow. Leds have a life expectancy of 90 to 100 years! They require only the standard 5 volt logic supply, although care must be taken to limit the segment current with a dropping resistor so as not to instantly frizzle the led. Since they are now available in three colors, there is little problem about filtering; yet a cleaner, more easily read display will result if one uses a polarizing filter with them.

Because leds require a relatively large amount of current to drive each segment (between 20 and 40 mA), using them in digital wristwatches necessitates that the readout be *off* when not being viewed; the watch is turned on by depressing a button—at once a mild inconvenience and distinctive aspect of the pulsar watches.

LIQUID CRYSTALS

Perhaps the most controversial type of readout is the liquid crystal display. Developed by RCA to complement their cmos semiconductors, its greatest advantage is in the incredible small amount of current it draws from the power supply. It is meant for use primarily in devices using batteries.

Curiously, these displays require ambient light from the "outside world" to make them visible. Obviously, then, they cannot be seen in the dark, and only minimally in a dim environment.

Furthermore, liquid crystals are to an extent heat sensitive, and care must be taken when using them not to expose the device to direct sunlight. If one were walking outdoors or driving a car, one could easily bring these readouts out of their safe region.

Given the present format for liquid crystal displays, they do not read seconds when used in a digital wristwatch. Of course, when used in digital wristwatches, they can be left *on* all the time, but it appears that liquid crystal displays

Robert E. Berglas is associated with radio station WBAI-f.m. in New York City.

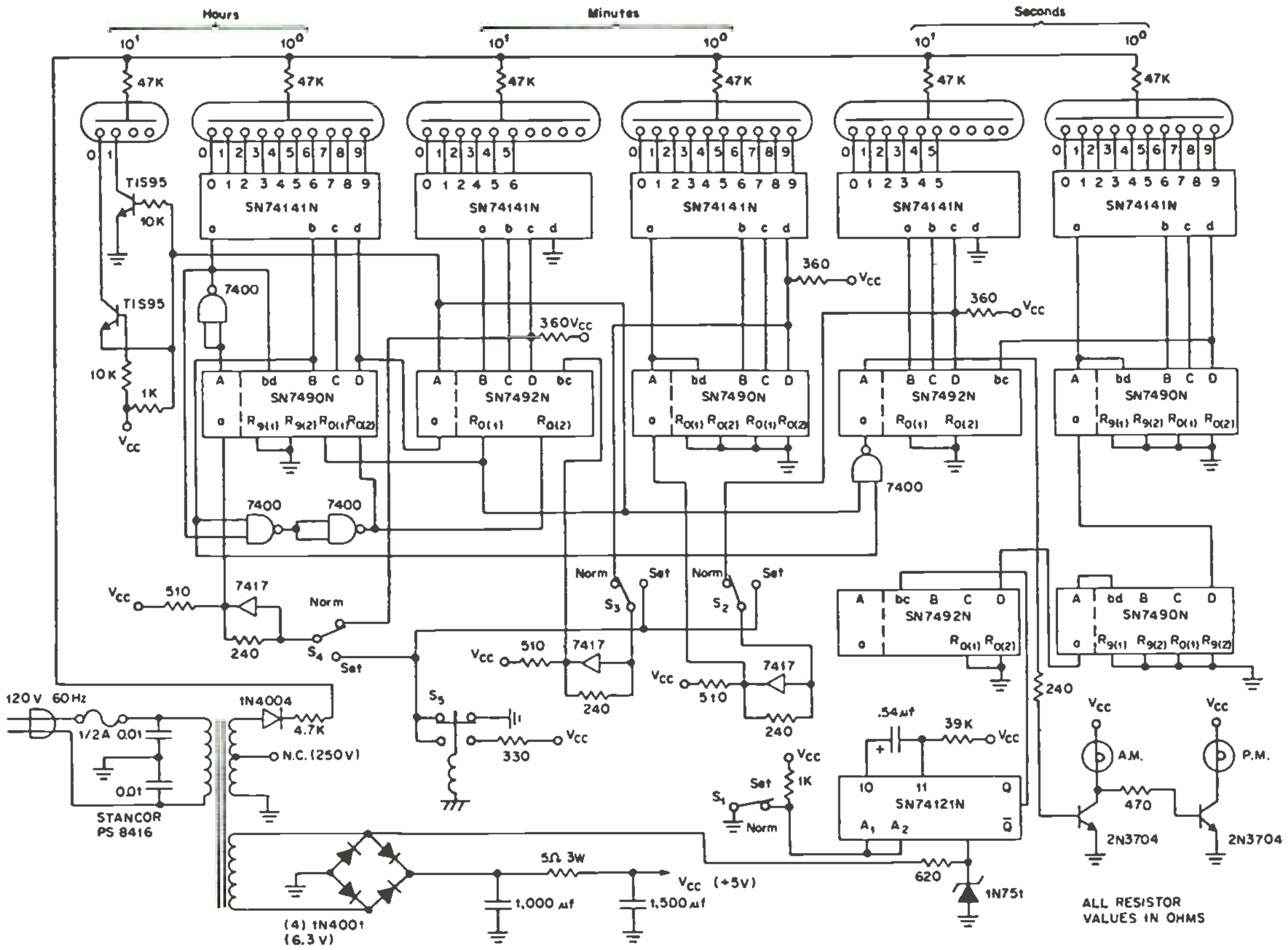


Figure 1. 12-hour clock made up of discrete ttl chips.

have a long way to go before they achieve the popularity RCA expects of them.

THE MINI-COMPUTER INSIDES

Perhaps the best way to introduce you to the way a typical digital clock works is to discuss how a clock made up of discrete ttl chips actually works. FIGURE 1 shows such a clock.

Basically, what we want to do is divide the 60 Hz line frequency to produce one output pulse per minute. The count-by-six and the count-by-ten circuits count and decode/display these pulses in tens, unit minutes, and seconds, respectively, up to fifty-nine minutes. The carry pulse from the count-by-six circuit is produced once an hour and is passed on to the hour counter. The circuit shown uses Texas Instruments chips, interfaced with a gas discharge type of readout (either Nixies or Sperry's).

As mentioned above, the system is synchronized with the 60 Hz power-line frequency. The SN74121 one-shot is triggered at its Schmitt input by 60 Hz pulses from the bridge rectifier. The one-shot pulse width is set for 15 ms (about 90 per cent of the 16.7 ms period of the line frequency) so that power-line noise is inhibited for 90 per cent of the period. An indicator for a.m. and p.m., using small incandescent lamps, is shown. If this indicator is not desired, then the connection of flip-flop (A) to the tens-of-seconds counter and the connection of the *NAND* gate to its input may be deleted.

There are five switches provided for setting the clock. The procedure for setting is: 1) When the desired indication for seconds appears, switch S1 is placed in the *set* position. This stops the clock by inhibiting the one-shot clock-pulse generator. 2) Switch S2 is placed in the set position. Switch S5 (a spring-loaded pushbutton) is then operated by pushing and releasing until the desired number is obtained for the minutes unit digit. The counter transition occurs when the pushbutton is released. Switch S2 is now returned to the *normal* position. If a toggle switch is substituted for the pushbutton S5, the switch should always be left in the position which causes the counter to step (the grounded terminal of the switch) before S2 is returned to the normal position. 3) The procedure of step 2 is repeated for S3 (tens-of-minutes) and then S4 (hours). S4 is also used to set the a.m. and p.m. indicator. Note that if the hours counters are set past 12, the tens-of-minutes counter will reset to zero. 4) With all digits set and S2, S3, and S4 in the normal position, S1 is set to the normal position and the clock begins to run.

The unregulated 5V power supply is adequate where good-quality line regulation exists. Note that the outputs of the SN74141N devices that are connected to the BCD outputs of the SN7492 counters are rearranged for connection to the Nixie tube displays. If 7-segment displays are used, the A-B-C inputs should be used to drive the decoder/driver. An extra flip-flop package will then be necessary.

CRYSTAL-CONTROLLED OSCILLATOR

The line frequency is not the only method of timing the logic. For greatest accuracy, a crystal-controlled oscillator

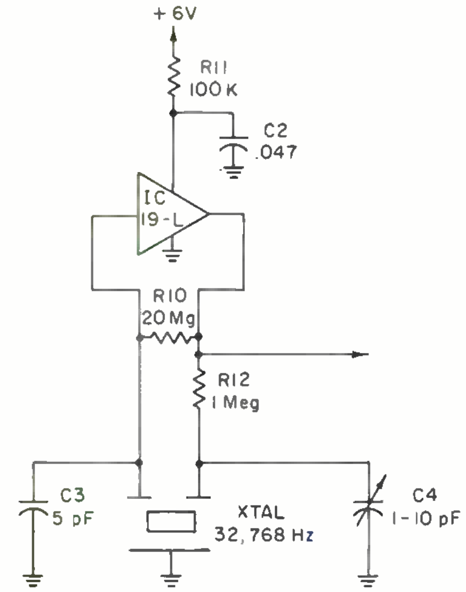


Figure 2. Crystal-controlled oscillator.

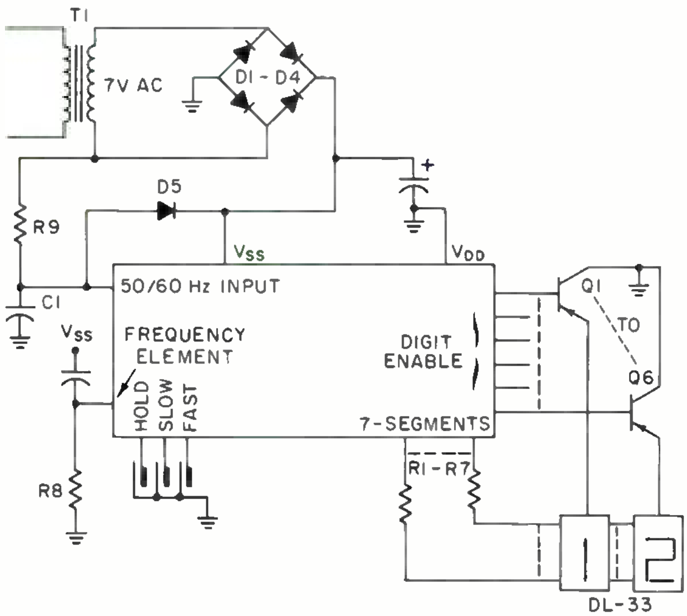


Figure 3. Circuitry of a kit-built clock.

can be used. FIGURE 2 shows such an oscillator, taken from RCA's battery-powered i.c. digital clock. It starts with the crystal oscillator. This integrated circuit section is an mos complementary inverter that provides the gain needed to maintain oscillation. Resistor R10 self biases the oscillator stage by providing d.c. feedback, setting the input and output at approximately half the 6 volt supply.

Although this i.c. circuitry usually draws very low currents when fast rise time signals are processed, in situations where signal transition are slowed, the supply current sharply increases. During a portion of the signal transitions both the p-channel and n-channel complementary devices are on and there is a current path directly between the power supply and ground. The input waveform of the oscillator has slow transitions since the crystal eliminates the high frequency components. One of the things done to keep oscillator current down is to operate it with a 6 volt supply even though we need a higher-voltage supply in other parts of the circuitry.

Let me say a few words about the crystal itself. First, as probably appears obvious by now, the higher the basic frequency of the oscillator, the greater the possible accuracy of the clock or wristwatch. And yet, from what I have heard from industry sources, the availability of properly cut crystals is the single most limiting factor in keeping digital wristwatches from being mass produced and inexpensive enough for the average consumer (the watches go for about $300 now). The inaccuracy of a crystal controlled unit is around five seconds a month, or about a minute a year, while a line-timed clock achieves an accuracy of 0.05 per cent of the 60 Hz waveform. It must be said, however, that a frequency counter is necessary to calibrate a crystal-controlled watch. There is really no other way to do it.

LARGE SCALE INTEGRATION CHIP

Recent advances in large scale integration (lsi), whereby the equivalent of several thousand transistors can be found on one integrated circuit chip, have made the discrete i.c. clock almost obsolete. Today, all that is needed is a single mos lsi chip, a few switching transistors, along with a handful of resistors and capacitors, and there you have it.

Several manufacturers have come out with the necessary chip, but National Semiconductor seems to have the lead in variety and simplicity of operation. Their chips range from a simple four-digit display (MM 5312) to a 40-pin chip with both regular alarm and snooze alarm in either a 12 or 24-hour format (MM 5316). FIGURE 3 shows a schematic of one such clock using the rather small DL-33 led displays. All the digital logic is performed inside the MM 5314. The remaining circuitry supplies d.c. power and responds to the commands from the chip to drive the readouts.

Some caution should be mentioned before anyone attempts to work with these mos i.c.s. A low wattage soldering iron must be used (not above 20 watts), and the tip of the iron must be either isolated from the power supply or grounded securely. The latter is accomplished simply by attaching a heavy duty clip from the heating element to a good ground. The chip, then, must be protected from static charges at all times, and it is a good idea to use sockets and not to insert the chip until the last step.

AVAILABILITY

Below are mentioned some suppliers of digital clocks in kit form. First, is the Heath Company (Benton Harbor, Michigan). They have three types, with one displaying both the date and month as well as the time. Poly Paks (P.O. Box 942R, Lynnfield, Mass.) has a clock with several readout formats and the possibility of using a crystal time base. I would suggest that only the experienced kit builder use this source.

The circuitry shown in FIGURE 3 is identical to a clock marketed by Bill Godbout Electronics (Oakland Airport, Calif.). Another source for this clock is Solid State Time in San Jose, California.

Finally, CEI (P.O. Box 327, Upland, Calif.) puts out several different and interesting clocks. Quite a novelty, is their Model SDC-1, which displays hours and minutes in sequence with only one digit. It shows complete time repeats every four seconds, or 15 times per minute. They also put out a crystal controlled model that uses, I believe, Numitron tubes, as well as a large, 3½-in led display clock that can be hung from a wall. These clocks from CEI can also be bought assembled. And for those who go all out for accuracy, CEI has a Standard-Time Receiver which is tuned to NBS radio station WWV.

In sum, digital clocks and wrist watches are surely the trend of the future. ■

Five monitors. One sound. Five JBL studio monitors.
You could record with any one, play back on any other, and take your pick among the rest for mixing or mastering.
The only differences are acoustic output, size and cost.
No matter what size your studio is, you can cross reference with any other studio using JBL's.
But reading isn't knowing for sure. Come listen to one. Or two. Or five.
JBL Studio Monitors from $303 to $1596.

James B. Lansing Sound, Inc./Professional Division/3249 Casitas Avenue/Los Angeles 90039.

JBL

Circle 23 on Reader Service Card

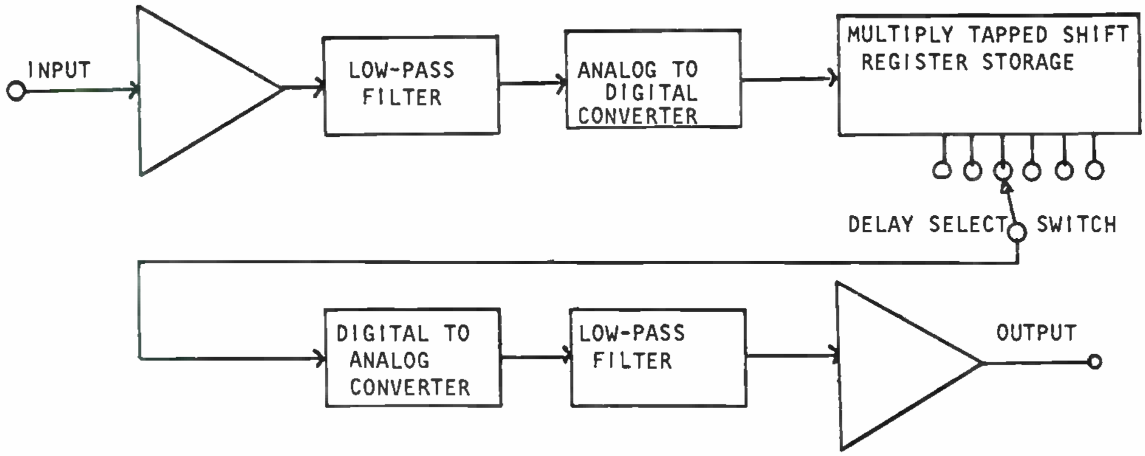


Figure 1. Standard delay line block diagram.

RICHARD FACTOR

The Digital Delay Line Revisited

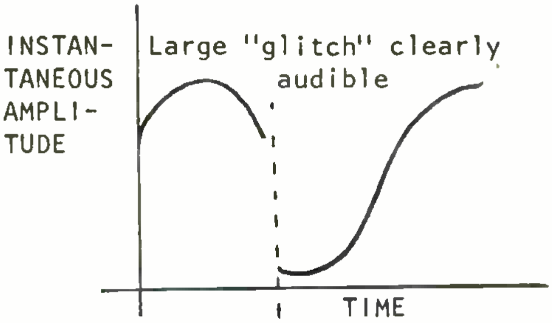
Random Access Memory allows greater versatility, including flanging and pitch changing.

YOU DON'T have to go back too far in the audio business to remember the first digital delay line, complete with such exotic concepts as an analog-to-digital converter, magnetostrictive delay lines, anti-aliasing filters, and such marvelous *arcana* as *bits* and *clocks* (which didn't even tell time).

Well, if you've been awake in recent years, you know a bit about digital technology by now. Your watch is digital, you have a digital calculator (or pocket computer for a few extra bucks), your voice is frequently digitized on the telephone, your tape machine searches digitally, and you may have even learned to count on your fingers. Of course, if you've done any mixing or sound reinforcement work, you've undoubtedly used a digital delay line for the special effects or time synchronization of which it is uniquely capable.

Richard Factor is Vice-President of Eventide Clockworks, Inc., New York City.

Somewhat less likely, you've had occasion to delve into the electronic circuitry by which these units achieve their delay. The ddl is "transparent" to the end user; i.e., an audio signal goes in, and an audio signal comes out somewhat later. Absent curiosity or malfunction, there is no need for the user to know what goes on inside the unit. Ask a non-technical person what is going on inside a ddl, and he will probably make some reference to analog-to-digital converters, shift registers, and digital-to-analog converters. Don't ask any further questions as they will probably lead to mild embarrassment. They needn't, though, because that last sentence is a fair summary of what really does go on in a ddl. After a brief flirtation with magnetostrictive delay lines (very clumsy mechanical delay lines), the industry universally adopted the integrated cir-



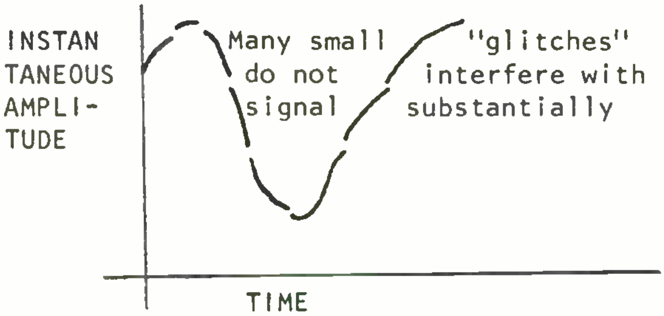


Figure 2. Delay-change glitches.

cuit shift register as its method of storing signals to be delayed.

HOW IT WORKS

Very briefly, the principle of operation of the ddl is this:

1. An audio signal is low-pass filtered to eliminate super-audible signals which could cause beat notes and spurious outputs.
2. The signal is then converted into a digital format at a rapid rate, typically 25 to 50 thousand times per second. Each of these 25k to 50k samples is represented by a digital *word,* which consists of a group of 1's and 0's. The word represents a specific voltage level as present in the original signal.
3. This word is *clocked* into a digital storage medium, such as a semiconductor shift register, in which it is progressively moved towards the output, one storage location per clock or sample period. The delay is thus determined by the clock rate and the number of storage locations available by the rather simple relationship

$$\text{DELAY} = \frac{\text{Number of storage locations}}{\text{Clock or Sampling frequency}}$$

4. After the digital word reaches the output of the shift register, it is reconverted to an analog format and again filtered to remove spurious frequencies, this time primarily associated with the sampling clock.
5. The signal, thus converted and delayed, is conducted to the outside world, ready to begin its career.

The above admitted oversimplification completely ignores the differences between delay lines from different manufacturers, which are primarily related to methods of encoding the analog signals into digital format, and the various control features of the competing units. The method of signal processing determines the dynamic range of the unit; depending upon the application, ranges of from 40 dB to over 90 dB are desirable. It makes little sense to purchase more range than required because the cost is directly proportional to the dynamic range, as both the amount of storage and circuit complexity increase with increasing dynamic range. Likewise, a variety of units are available ranging from those with delay switchable in narrow increments with great facility to those in which delay is completely fixed at purchase. Naturally, you pay for control features, and for delay time.

The third major trade-off is frequency response. Within narrow limits, the frequency response is about one-third the sampling rate. Doubling the sampling rate doubles the rate at which the digital samples pass through the shift registers, and thus cuts the delay time in half. So, one would expect a given delay line to give half as much delay at 50 kHz as at 25 kHz. It is possible to compromise, however, so that a given delay line may have variable or selectable clock rates to allow longer delays when wide frequency response is not necessary, such as in some special effect or sound reinforcement applications.

SHIFT REGISTER

With all these differences, there has been one unifying and limiting factor in ddl design. All units have used the shift register as a storage medium. To see why this is limiting, let's look at the shift register.

The shift register is a *serial* storage device. It comes in various lengths, from 4 bits to several thousand bits. New technology has recently made 16 kilobit registers possible. At first glance, it would seem that shift registers are ideal for delay, because of their very structure. They work by transferring a packet of charge representing a digital 1 or 0 from one internal node to the next. No additional timing circuitry is necessary—the registers accomplish the delay all by themselves.

Furthermore, they may be connected in series to achieve longer delays, and the points at which they are connected can be used as delay taps. If each shift register provides, for example, 10 milliseconds of delay, and 20 are connected in series, 200 milliseconds are available in 10 millisecond steps. As a practical matter, many parallel shift registers are required to handle a full word, and the switching becomes cumbersome after a few taps are necessary. There are many techniques for circumventing this problem which add only minimal complexity to the entire system.

So what's the problem? Well, consider how to vary the time delay. There are two basic choices: one is to switch shift register taps; the other is to vary the clock rate. Switching taps creates discontinuities in the signal. Looking at a signal at point A and point B at, say, 1 millisecond time difference creates a sharp transient at the splicing point (see FIGURE 2). Even if the switching is accomplished electronically, such as with an optical encoder and digital multiplexers, this transient is unavoidable, except when the signal level is zero, or the points to be joined coincidentally have the same amplitude. If the delay is to be changed rapidly, this becomes a serious detriment, as many of these splices add noise to the signal.

Varying the clock rate overcomes this problem, but creates another one, the inability to vary the delay by more than a certain percentage. The upper limit of variation is governed by the allowable decrease in frequency response; the lower limit by the capability of the A to D converter and the timing circuitry. A typical variation of 50 per cent is insufficient to change from short to long delays, and since all outputs vary by the same percentage, it is impos-

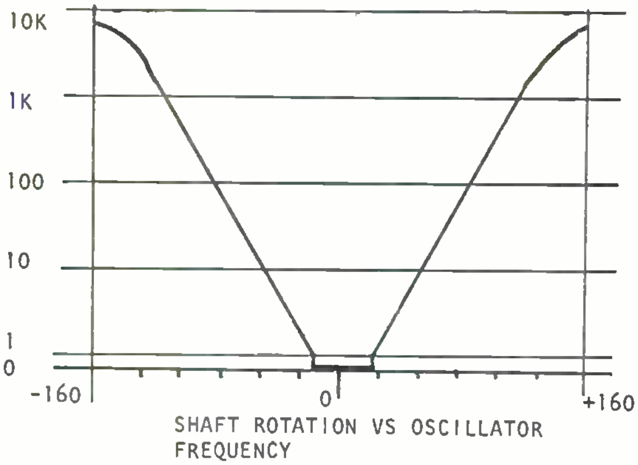


Figure 3. Delay-change control operation.

sible to vary one output with respect to the other. The basis of the flanging effect is the sliding of one signal past another in time, and so this lack of capability is a great disadvantage.

One solution to the above problems would be to have shift registers with taps at every sample. So doing would enable one to switch from tap to tap rapidly while encountering only insignificant splicing noise (FIGURE 2). This is because, with normal program material, the amplitude difference between successive samples is very small. Switching between samples in this manner produces at worst a low amplitude tone at the switching rate which is effectively masked by the signal. The only time you run into trouble with this method is varying the delay of high frequency deterministic signals, which is very unlikely in normal circumstances.

Unfortunately, tapping shift registers at every bit would require a maze of wiring large enough to fill an ordinary city dump, and the equipment would end up there after one repair was attempted. (It is possible to use combinations of shift registers to achieve an arbitrary delay time, but this is not the same and will not work, just in case anybody is tempted to try. The reason it won't work is that in switching individual shift registers around, it becomes necessary to wait for them to fill up, which can take an arbitrary length of time, much greater than one sample, before the output is usable.)

RANDOM ACCESS MEMORIES

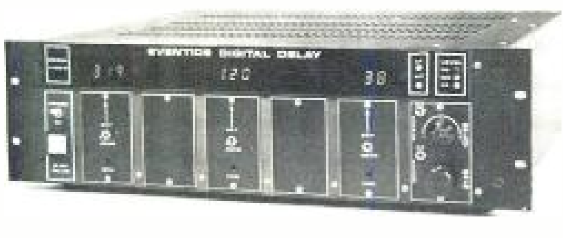
Another solution to the above problem is to use random access memories. This solution does work. A random access memory is a semiconductor chip (or assemblage thereof) which can store individual chunks of data, and deliver them up upon command. It differs from the shift register in that any of the stored data is immediately available regardless of when it was entered into the memory. If the shift register is compared to a pipeline, the ram may be compared to a book, in which each page is numbered and accessed without reading its neighbors. In digital terminology, the "page numbers" are *addresses*. An individual ram integrated circuit may have from 16 addresses, as used in either very old or very fast chips, up to 4,096 addresses as used in many of the newest computers. Assuming each address location holds 1 bit, it is obvious that one "4k ram" is the equivalent in storage capacity of four industry standard 1k shift registers. Of course, nothing exists in a vacuum, and the 4k ram must be economically viable, available, and reliable. Fortunately, this is the case, as the price of the 4k ram, which was astronomical until mid-1975, is now in the range where it is reasonably comparable to the price of the shift registers which it replaces. Anticipating the price reductions, we began the design of a digital delay line using random address storage instead of shift registers. Some of the details of the design, and the possibilities of rams are described below.

GETTING DELAY FROM A MEMORY

To begin with, getting delay from a memory is a bit more involved than getting delay from a shift register. To continue with the book *vs.* pipeline analogy, if you put something into a pipeline, you need only wait for it to come out. The delay is equal to the length of the pipeline. If you wanted to get delay from a book, you would have to read the data a certain number of pages after the beginning. Since this is a dynamic process, the point at which the data is written must also vary. (If it did not, data would be overwritten at the same location. Look at an old piece of carbon paper and try to imagine what that would *sound* like!) Therefore, the way to get delay from a memory is to employ a *pointer* or *base address* register. Every time a new sample is writttn into the memory, the register is decremented (decreased by 1). This register then points to the location of the most recently written data. Since the register is decremented once each sample period, adding 1 to the register contents points to a sample that is delayed by a sample period. If the sampling rate is 50 kHz, then, each sample is 20 microseconds delayed from the previous one. If data is being written at address #150, then data being read at address #200 is delayed by 1 millisecond. Fortunately, the actual numerical address is irrelevant, as all data storage locations are identical. Thus, an output can be obtained at an arbitrary delay simply by generating a number equal to the number of samples difference between input and output. The "housekeeping" is taken care of by a single register and arithmetic unit located on the circuit board which contains the memory.

MULTIPLE OUTPUTS

Most delay lines, especially those used in recording studios, have two or more outputs. Multiple outputs are especially desirable in those applications involving choral effects and reverberation. In order to accommodate multiple outputs, it is necessary to organize the system as a *bus*, which means simply that several different signals can share the same physical connection. Doing this requires "3-state" logic, which differs from ordinary 2-state digital circuits in that not only can a 1 or a 0 be output, but the output can also be turned "off," in which case it assumes a high impedance. If multiple outputs are connected together, but only one is in a low impedance state, then the bus assumes the state of the low impedance output.



Model 1745M digital delay line.

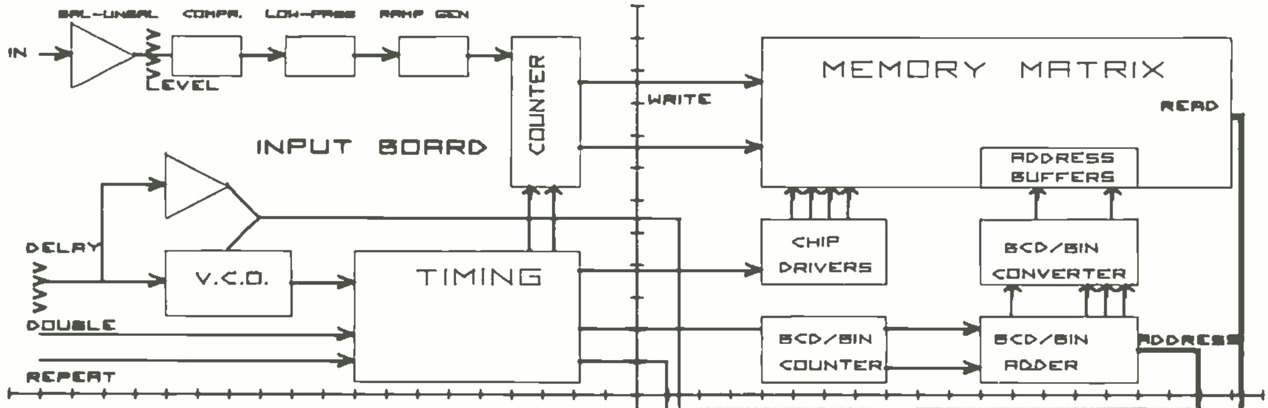
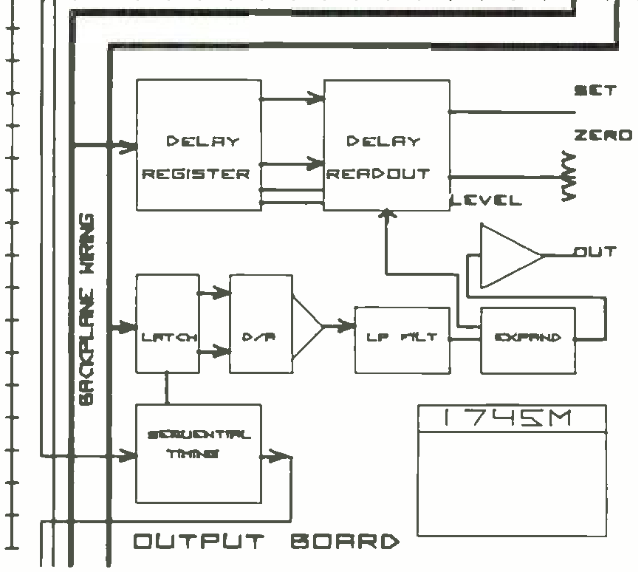



Figure 4. Diagram of the operation of the model 1745 M digital delay line.

By using modular output cards, and activating them sequentially, one can access several different addresses during any given sampling interval. The number of addresses which can be accessed is determined by the *access time* of the memory chip used and the sampling interval, which, as stated above, is 20 microseconds. Most common 4k rams have access times on the order of 0.2-0.4 microseconds, which should be sufficient for 50 to 100 outputs. However, computing the proper address also requires time, and common ttl circuits, which are very fast, consume quite a bit of power. As one of the objectives of the design was to reduce power consumption, we used all cmos circuitry, whose power consumption decreases almost to zero as its speed decreases. Even so, there is plenty of time to service up to seven outputs, and module positions are provided for up to five, leaving margin for extra functions.

Another problem arises in that rams are universally binary devices, and human beings are generally decimal. Some provision must be made to allow setting the delay (and reading it out) in human-decipherable units. Since each sample is 20 microseconds, one could simply convert the binary number representing the delay to decimal form and multiply by 0.020 to give milliseconds. Unfortunately, long binary-to-decimal conversions are slow or require much hardware. A far simpler solution involves using a small (0 to 999) decimal-to-binary converter, and doing all the output module addressing in decimal form.

Each 4k ram has binary addresses frm 0 through 4,095. Then memory locations 0 through 999 (binary) are addressed for the first 1,000 samples of delay. If 1,001 samples of delay (20.02 milliseconds) are required, the next address used would be binary 1,024. In effect, for convenience, 96 addresses out of every memory are ignored and wasted. These addresses are still there, by the way, if needed, but it is a lot cheaper to waste them than to perform the conversions otherwise required.

Using this system and sampling rate, a 16k memory, requiring 40 memory chips gives 319.98 milliseconds of delay in 20 microsecond steps, as compared to our previous system requiring 108 shift registers for 199 millisecond steps. A small additional advantage is that no extra shift registers are required for additional outputs. Assuming comparable reliability between the rams and shift registers, the memory design should be over twice as reliable. Furthermore, the industry has settled on two or three basic 4k ram designs, and all are multiply-sourced. We opted for the 22 pin, non-multiplexed address configuration for simplicity, and because space saving was not an urgent criterion. We have evaluated several manufacturers' chips and found all but one acceptable. This, hopefully, will make delivery times independent of Silicon Valley idiosyncrasies.

OPTICAL ENCODER

One final unusual design feature: our 1745A delay line utilized an optical encoder (see January, 1974 **db** Magazine) for delay switching. This eliminated the need for coarse and fine delay controls and permitted fully incremental switching in 1 millisecond steps. The control had 20 lines, and produced 20 pulses per revolution, and could be spun to traverse the whole control range in one spin. To do the same with a unit which varies in 20 microsecond steps would require an encoder with 1000 lines. Although such devices are producible, and are used in precision mechanical systems, it was deemed impractical to use such a unit for reasons of cost and ruggedness.

Instead, we designed an oscillator of wide frequency range with a small deadband in the center of its control range, and with parabolic control taper. The oscillator frequency range is from 1 Hz to 5 kHz, which allows the full range of delay to be spanned automatically over several seconds to several hours. It should be noted that although the delay varies in 20 microsecond steps, the readout is only to the nearest millisecond. If one needs to know the precise delay, the information is available on data lines which may be connected to readout drivers. Other features of the 1745A were duplicated in relatively uninteresting ways. The delay double feature which allows doubling the delay at the expense of frequency response was implemented by inhibiting alternate base address decrementing pulses. The repeat feature was implemented by inhibiting memory write pulses, thus preventing overwriting old data and so saving it.

Superficially, then, we have a new delay line with a bit more delay and a whole bunch of familiar features employed in a wholly new way. As stated earlier, the delay line is, and should be, transparent to the user in that, regardless of the implementation, you put audio in and get audio out. What does all this mean to the user?

SIGNIFICANCE OF RAMS

Nobody would waste time with rams if they didn't mean something; of that you may be sure. They mean that true flanging can be accomplished with a single digital delay line. The fine delay steps allow sweeping a variable output past a fixed output in tiny increments, thus giving an apparent continuous variation in tonality, as opposed to the step variations possible with other systems. Unlike analog delay flangers, the flanging may be performed after any fixed delay. Even "pre-flanging" is possible with a single delay line, by taking the flanging effect from the first two outputs and the dry signal from a third at greater delay. Of course, as with all digital systems, there is no degradation of signal-to-noise ratio as the delay is increased.

In addition, continuous doppler shift or pitch change may be introduced into any signal simply by increasing or decreasing the delay in a continuous fashion (facilitated by the oscillator control). A rather long period of change is available before an artifact is introduced by the delay recycling from 319 milliseconds back to zero, and this can be minimized in certain ways involving interleaving more than one output.

A wholly new effect, *tunneling*, can be achieved by connecting the audio output back to the audio input, in a manner similar to classic "tape reverb." As the delay varies, an effect similar to flanging, but much more intense, gradually changes to rapid reverberation to variable length echo. All the while, the pitch of previously processed material is increasing or decreasing, depending upon the direction of delay change. The effect is audibly similar to standing in the center of several whirling program sources as they spiral towards you approaching the speed of sound. This effect is so new that you probably haven't heard it yet. You will.

Using rams means that an auxiliary module which computes addresses in accordance with an appropriate program can be used to obtain an arbitrary pitch change ratio, and maintain that ratio. *Any* ratio can be obtained. By proper address manipulation, it is possible to read audio out backwards! Think of that, special effects fans! And of course, pitch change can be turned into tempo change in conjunction with a tape player.

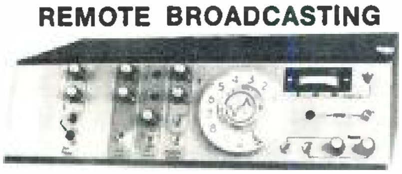
It also means that precise comb filters may be implemented with much finer control of delay. Since the system is crystal-controlled, the delay time will not drift more than a few parts per million, and so a null can be set for some frequency and that frequency and its harmonics will disappear.

This process makes it possible for largely experimental applications, such as narrow bandwidth speech scrambling, to be reasonably implemented. For instance, you can construct a card that will read out addresses 0 to 1,000 normally, then jump to 5,000 to 6,000, then read out 2,000 to 1,000 backwards, etc., until the whole capacity is used up! Performing the inverse operation at a remote end will descramble the signal. The whole process, unlike digital encoding, does not significantly increase the bandwidth of the output, and thus it may be put on ordinary communications channels, such as the telephone. This may also be an aid to communications, by using the memory delay line in conjunction with several filters, and setting each filter band at a different time delay. This is known as time-diversity transmission and can also be implemented with non-memory delay lines.

Another possibility is that experimental applications requiring digitized audio available at various times can be implemented with a convenient test bed. Computer time need not be taken up implementing A to D routines and housekeeping. Samples may be withdrawn at any computed delay and used as the experimenter desires.

In addition to the above advantages, there is another capability built into the system as a consequence of the memory bus architecture. If one can hang several outputs on a given bus, why not several inputs as well? Why not indeed! In fact one can, and so we did. The delay register inputs, normally set sequentially by the oscillator, can be set in parallel by placing the desired number on the address bus and strobing the output card which it is desired to set in the "dead time" between the last readout and a new sample. Provision has been made for a remote control module which enables precision setting of any or all outputs by digital control. Data may be furnished either from a remote console or from an automated mixdown system. Thus, another component can be operated in conjunction with automation, leaving the engineer and producer even freer to *listen*. ■

REMOTE BROADCASTING



REMOTE SITE CONSOLE

- Use standard telephone line
- Direct distance dialing
- Set up and check out in minutes
- High and low end frequency compensation with automatic level control option
- Lost line auto hang up option Auto pick up on redial option
- Up to 6 microphone inputs with level control option, 4 headphone outputs with level control
- Output matched for standard telephone line or loop with level control, VU meter, built-in telephone line coupler and output for PA amplifier
- AC/DC with battery test meter
- Built-in telephone dial
- Options in addition to those noted include: carrying case, microphones, headsets, test tone generator, aux inputs, phono cartridge input and three pin connectors

STATION END CONSOLE

PULSE DYNAMICS MANUFACTURING CORP.
Box 355, Depot St., Colchester, Ill. 62326
(309) 776-4111

Circle 28 on Reader Service Card

JOHN WORAM

A Visit With Uncle Horsely

The quiet will drive you crazy.

Who? Well, it seems there's this Doctor Adams—an electron-microscopist by specialty. Being a doctor is okay, but playing bass is a lot more rewarding. Perhaps not in $$, but who wants to look at electrons all day anyway?

Now Doc Adams' uncle John Horsley had a lot of property up in the lake country north of Montreal. Some time ago, Uncle Horsley went on to his reward, and there were all these houses on the property, and nobody was living in them, and one of them would make a great little recording studio, and who wants to be a doctor anyway?

But you can't call a place "Doctor Adams Recording Studio;" it sounds too clinical. "Adams Sound" doesn't make it either. So, *Uncle Horsley's* it is, giving recognition where due.

Uncle Horsley's is not one of your super slick infinitely baffled multi-track sound emporiums (emporia?). As a matter of fact, its a fairly small room, about right for a combo, providing they leave their retinue of roadies, groupies, accountants, and assorted hangers-on back at the airport in Montreal. The control room is even smaller, so the persistent session crasher can't possibly squeeze in. There's just room for the bare necessities, like an MCI 16-tracker, some Ampex AG 440's, dbx, a 19x16 console, and a few bottles of wine. With the possible exception of that last item, everything else was provided and installed by Chromacord of Montreal.

There are two windows in the studio. One looks out on the woods behind the house, the other on the woods in front of the house. If you want to gaze into the control room while playing, you're going to find it tough. There's a big brick fireplace in the way. Uncle H's man at the knobs can keep an eye on your microphone via cctv, and you can watch the birds outside if you don't know how to read music.

Most visitors linger a while at Uncle Horsley's. In fact, some three days slipped by unnoticed while this epic was being "researched." The time was divided almost equally between meals, walks in the woods, and long naps, plus a brief look at the studio. For others who might find this sort of tempo exhausting, some of the nearby houses are being spruced up for living in while attending to recording chores.

For a change of pace, Uncle Horsley's seaplane (actually, it belongs to Doc Adams) stands ready to take you lake hopping between sessions. Or, if you have the time, there's a canoe and a fishing rod available.

Uncle Horsley's is mostly about music—jazz or jingles. If you need Marshall amps for earphones, the quiet up here will drive you crazy. Besides, you'll scare the birds. ■

Doc Adams' seaplane.

Small, but good.

The control room, with the 19-in, 16-out console, MCI 16-track, dbx, and Dolby units at hand.

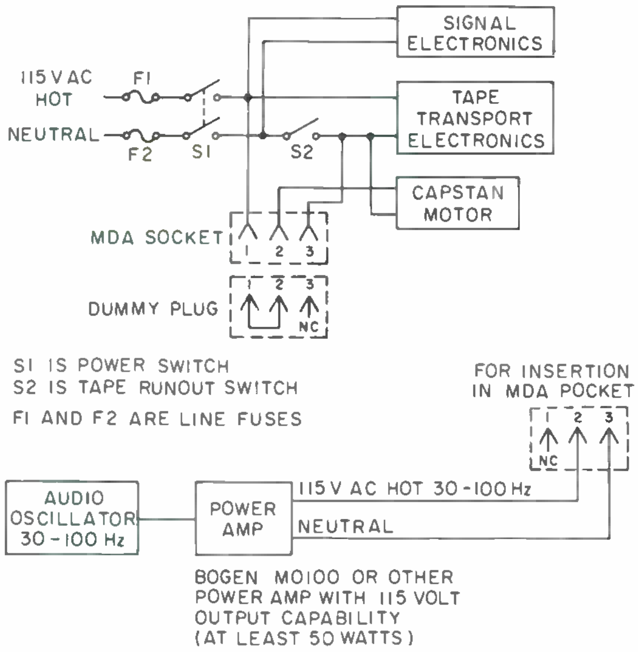


Figure 1. Removing the dummy plug from the mda socket.

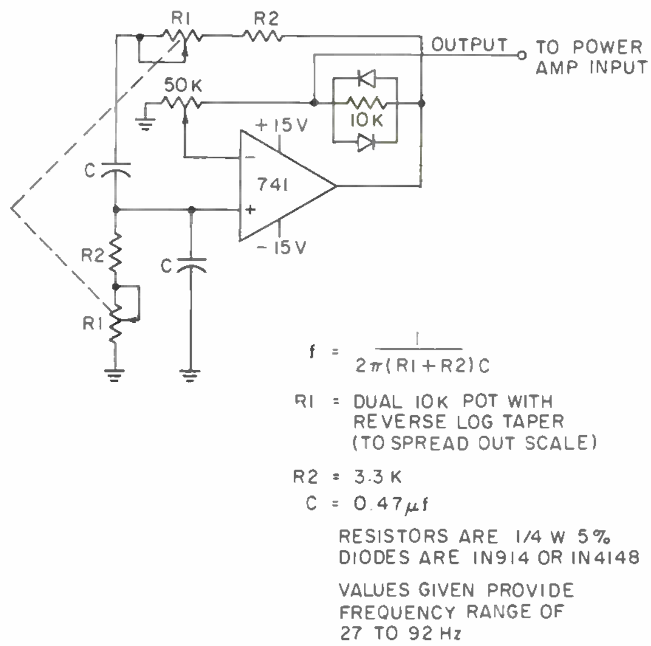


Figure 2. Design for an inexpensive oscillator.

ROBERT E. RUNSTEIN

A VSO Switichng System

Using electronics already present, the complications are taken out of connecting a variable speed oscillator.

Connecting a vso (variable speed oscillator) is usually a bothersome task. It requires the connection of a variable frequency oscillator to a power amp capable of delivering 115 volts to its load, the use of a voltmeter to measure the voltage reaching the load while the oscillator output level is adjusted, and fumbling underneath a tape transport to remove the dummy plug from the mda (motor drive amplifier) socket so that a connector carrying the variable frequency power from the vso can be inserted (Figure 1).

The complexity of the situation arises from two causes. First, the output voltage from the power amp must be

Robert E. Runstein is the author of the book, Modern Recording Techniques.

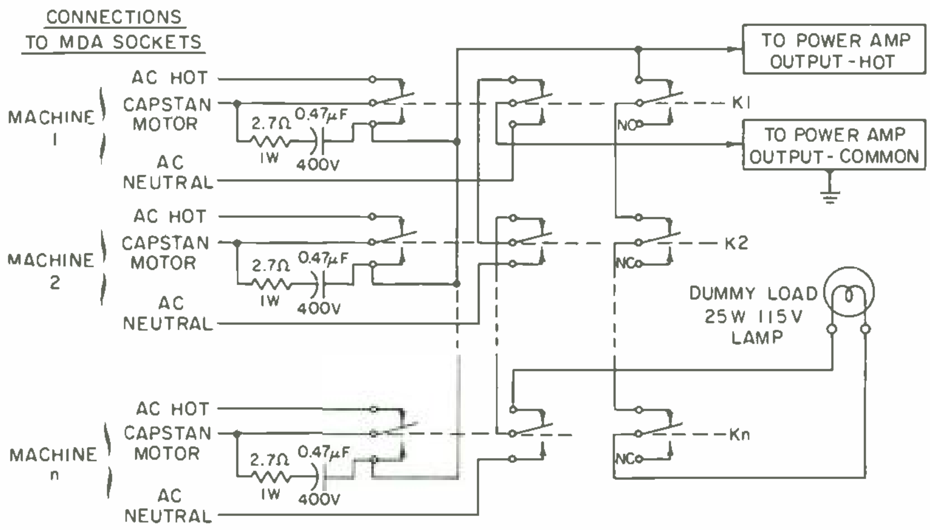


Figure 3. A switching system that automatically connects a capstan motor into a vso.

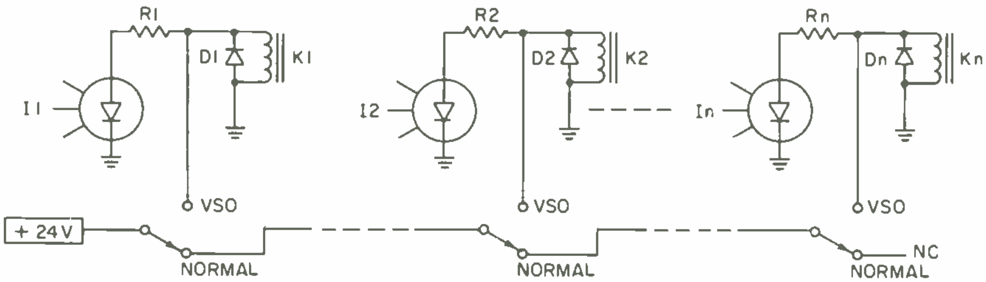


Figure 4. Using this circuit, the capstan motor will run whenever vso operation is selected, regardless of mode.

D1, D2, . . . Dn = IN4001
K1, K2, . . . Kn = POTTER & BRUMSFIELD KHP 17D11 OR EQUIV. (ONLY 3 POLES NEEDED PER MACHINE)
I1, I2, . . . In = LEDs
R1, R2, . . . Rn = SERIES RESISTOR TO CONTROL LED CURRENT

monitored so that the oscillator output can be set to provide the proper voltage to the motor. While synchronous motors will operate at synchronous speed (their speed will depend on the *frequency* of the applied voltage) over a fairly wide range of voltages, too low a voltage may cause the motor to stall, while too high a voltage may overheat the motor windings and may also exceed the power amp output capability. That causes distortion of the waveform, which can result in speed irregularities. Secondly, it is desirable to be able to vary the speed of any one of several machines at the flick of a switch without having to go to the expense of obtaining a separate power amp to drive each one.

The answer to the first of these problems is to allocate one oscillator to the vso system exclusively, and preset its output level so that the power amp delivers the proper voltage to the capstan motor. The oscillator need only be capable of limited frequency operation; most capstan motors will stall much below 25 Hz (less than ½ normal speed) and will reach maximum speed somewhere between 90 and 100 Hz (more than 1½ times normal speed). For smooth operation the oscillator should be capable of covering these frequencies without switching ranges, and with the high and low limits at opposite ends of the dial for good resolution. FIGURE 2 is a design for an inexpensive oscillator which appeared in *Operational Amplifiers Design and Application*.[1]

The frequency of oscillation is determined by the formula $f = ½\pi (R_1 + R_2)C$. Resistors R_2 limit the oscillator's highest frequency to that which produces the highest tape speed—determined experimentally for the machines in use. If the oscillator frequency is permitted to exceed that frequency, slippage of the capstan motor's rotor can actually cause tape speed to slow down a bit. R_1 varies the frequency and should have a reverse log taper to spread the values out, but a linear taper will work adequately (settings above 60 Hz will be compressed into approximately the last 90 degrees of the control's rotation). R_3 is adjusted for minimum distortion, which occurs near maximum output before clipping. Distortion is on the order of 2 to 3 per cent, which is sufficiently low for this application.

SWITCHING SYSTEM

The answer to the second problem is the switching system to be described (FIGURE 3). The principle is simple. The vso is left on, driving a dummy load (a 25 watt standard light bulb) until variable speed operation is de-

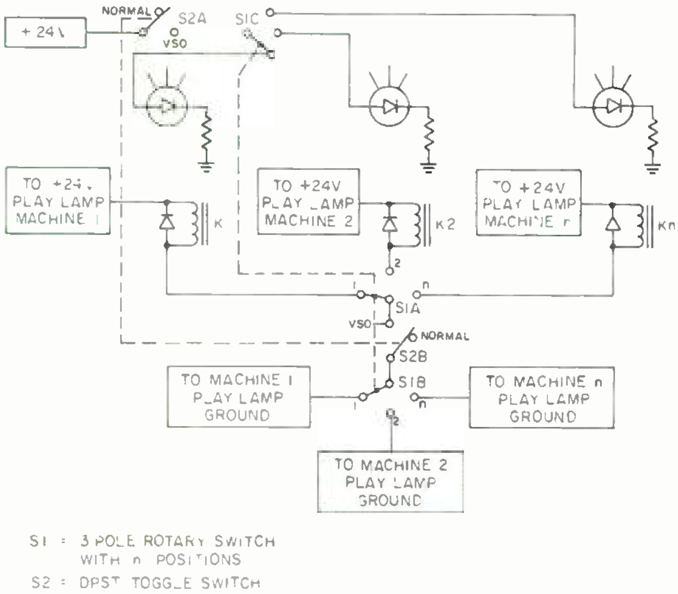


Figure 5. Substitute circuit, used if the tape machine stops capstan rotation when the machine is not in the play or record mode.

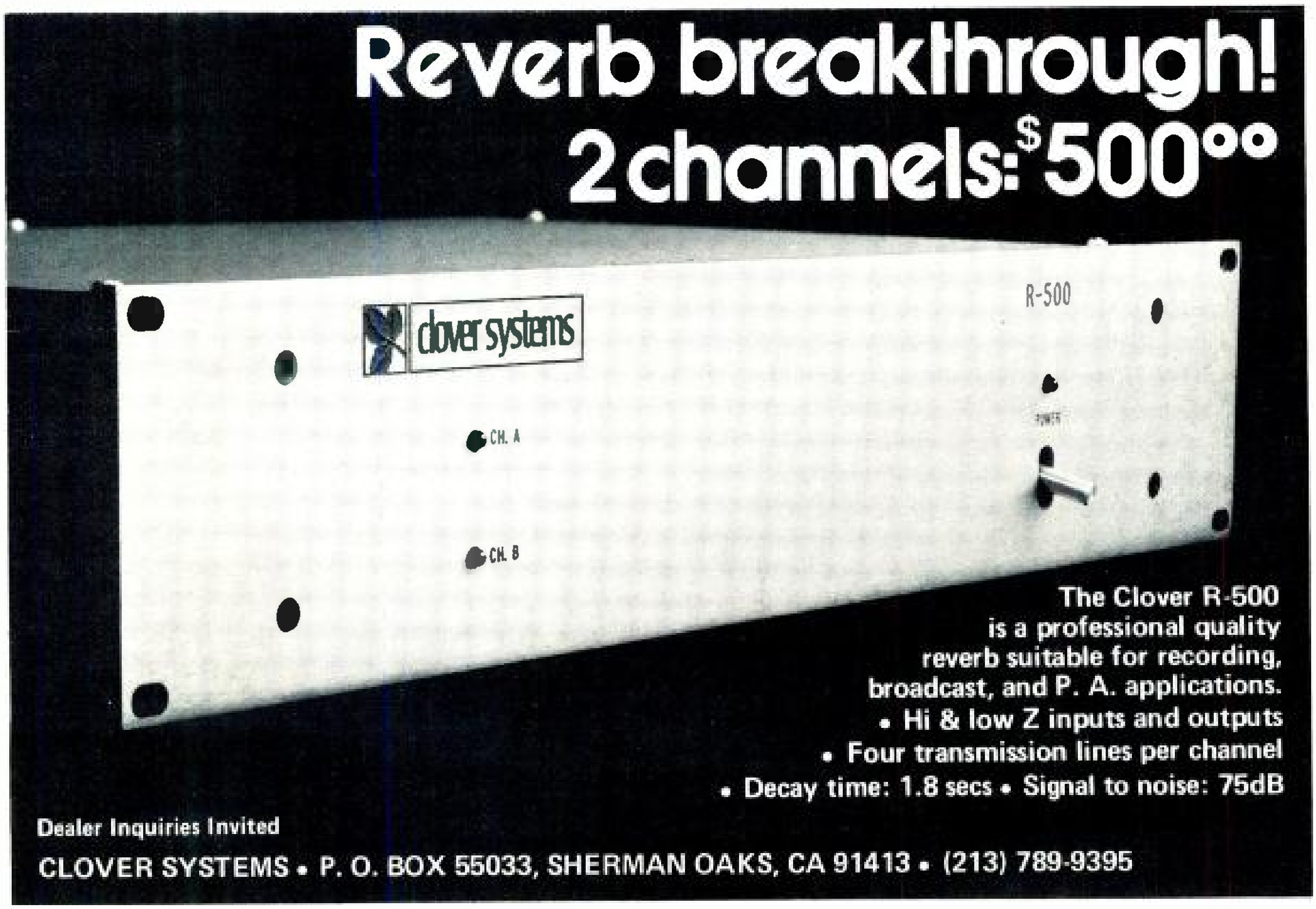
sired. At that point, a relay disconnects the power line from the capstan motor of the desired machine and substitutes the output of the vso. As many machines as desired may be wired into the system; the selection switch circuitry permits only one machine to be connected to the vso at a time. The circuit illustrated in Figure 4 results in the capstan motor running whenever vso operation is selected, regardless of what mode of operation the tape machine is in.

The circuit illustrated in Figure 5 can be substituted if the tape machine is of the type in which capstan rotation is stopped when the machine is not in the play or record mode. Construction is straightforward, with the only complications being precautions in the wiring of the 115 volt vso circuitry (use at least #18 wire) and the avoidance of ground loops.

The vso wiring is complicated by the fact that most tape machine manufacturers send only the high side of the a.c. line to the capstan motor through a jumper in the mda socket's dummy plug, leaving the low or neutral side of the line permanently connected. Thus, connection of the vso to the mda socket means that one side of the vso's output is connected to the a.c. power line neutral through the tape machine selected. Most power amp manufacturers connect the common side of the power amp's output transformer to the amp's chassis and therefore to the power line ground (if the unit is furnished with a three wire a.c. cord) or to one side of the power line through a capacitor (if a two-wire a.c. cord is used).

NECESSARY PRECAUTIONS

Given these conditions, the following precautions must be taken. The side of the output transformer connected to the power amp's chassis must be connected to the neutral side of the wiring to the capstan motor. If the power



Circle 41 on Reader Service Card

amp does not have a three-wire line cord which would assure that the common side of its output transformer is at ground (and therefore close to neutral) potential, the a.c. plug should be rotated so that the side of the line connected to the chassis through the capacitor is connected to the neutral side of the power line. The chassis of the power amp should then be grounded to the a.c. power ground. For additional protection, the power amp's two-prong a.c. plug should be replaced with a polarized three-prong plug which also provides a convenient point for the ground connection to be made.

Since it is common studio practice for tape machine a.c. power grounds to be left floating via the use of three- to two-prong a.c. adapters to prevent ground loops, you must be sure that the two-prong end of the adapter is inserted into the wall socket in the proper direction. If it is not, the connection labeled *neutral* on the mda socket will really be the hot side, and when the vso is connected to this machine, the hot side of the a.c. line will be connected to the a.c. neutral through the tape machine's power switch and fuse, and through the power amp's chassis, causing the tape machine fuse to blow. Rather than risking this, the tape machine's power cord ground wire should be disconnected inside the machine so that the polarized three-prong a.c. plug can be inserted directly into the wall socket, eliminating the possibility of blown fuses.

The neutrals of the tape machine a.c. lines must not be connected together by the vso switching system, for this may cause ground loops. The relays must therefore switch the neutral as well as the hot side of the vso power to the capstan motor selected, as shown in FIGURE 3.

NO CONNECTION BETWEEN CONSOLE GROUND AND POWER AMP

Since the power amp chassis is connected to the a.c. power ground, it is necessary that there be no connection between console ground and the power amp chassis, to prevent ground loops. If the oscillator used has its own power supply (rather than operating from the console's d.c. supply) and its chassis is not electrically connected to the console via rack mounting or to a.c. power ground via its power cord, no ground loops will occur. When an oscillator powered from the console or electrically connected to the console in some manner is used, an isolation transformer will be needed to isolate console ground from the vso power amp chassis. In the event either the oscillator output or the power amp input is already transformer-coupled and floating, that is sufficient if the cable to the power amp is shielded with the shield connected to the power amp chassis at that end and cut off at the oscillator end (telescoping shield).

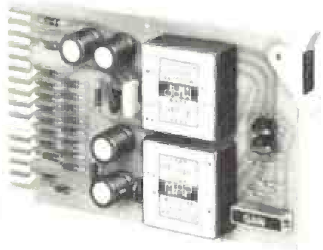
Only one other item needs to be mentioned to prevent some head scratching when the system is connected. Some tape machine manufacturers insert the tape runout switch in the neutral side of the a.c. line and wire the neutral connection on the mda socket to the transport electronics side of the runout switch (see FIGURE 1). As a result, when vso operation is selected, the power amp's neutral connection bypasses the tape runout switch, preventing automatic shutoff at the end of the reel. This is a small drawback when considering the many advantages of an easily implemented vso system. ■

REFERENCES

Tobey, Graeme, & Huelsman. *Operational Amplifiers: Design and Application.* McGraw-Hill Book Co., New York, N.Y. 1971. pp. 383-385.

What's new from MAP?

DA's...



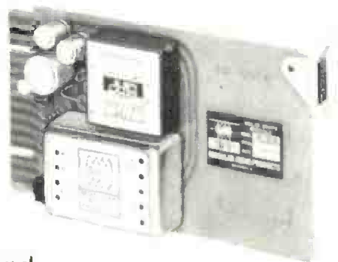
Model 4820 Distribution Amplifier

- Bridging (6K ohms), Balanced, Transformerless (Differential) Input Configuration
- 8 Balanced, Transformerless Outputs (Precision Resistor Network)
- Continuously Adjustable Gain, Up to +10 dB
- Low Noise (Output) −90 dBm
- Low Distortion (Typ. 0.1%)
- High Output Level, +20 dBm per channel
- 80 dB of isolation between Outputs & Output to Input
- Reverse Polarity & Overload Protected
- Miniaturized, Plug-in P.C. Card Construction
- Utilizes MAP 1731A Audio Operational Amplifiers

...and VCA's

(Voltage Controlled Amplifiers)

Model 5100 VCA Module



- 130 dB Control Range (−30 dB Gain, −100 dB Attenuation)
- Low Noise, −98 dBm (Output) at Unity Gain
- Low Distortion, 0.05% Typical
- Broad Frequency Response, 10 Hz to 45 kHz, ±0.25 dB
- Accurate Control Linearity & Tracking, 20 dB/Volt
- Virtual Ground Summing Input

Model 4100-VCA Card

- Self-contained, Requires No External Circuitry
- High Input level, +27 dBm
- High Output Level, +20 dB into 100 ohm load
- Provides Reference Voltage Supply for External Fader
- Utilizes MAP Model 5100 VCA in Conjunction with MAP 1731A Audio Op-Amp
- Optional Built-in Output Transformer (Model 4200)

Other MAP Pro Audio Components

- Audio Op-Amps
- Console Modules
- Equalizers
- Compressor Limiter
- Precision Oscillator
- PC Card Amplifiers
- Audio Transformers
- Power Supplies and Accessories

SEE THEM IN L.A. AT THE AES • Booth 33

Shopping for a Console?

Ask about our Recording & Broadcast Console Modules . . . and Consoles.
For prompt assistance call or write Rick Belmont.

MAP MODULAR AUDIO PRODUCTS, Inc.
A UNIT OF MODULAR DEVICES, INC.
1385 Lakeland Ave., Airport International Plaza
Bohemia, New York 11716 516-567-9620

Circle 37 on Reader Service Card

db classified

Closing date is the fifteenth of the second month preceding the date of issue.
Send copies to: Classified Ad Dept.
db THE SOUND ENGINEERING MAGAZINE
1120 Old Country Road, Plainview, New York 11803

Rates are 50¢ a word for commercial advertisements.
Employment offered or wanted ads are accepted at 25¢ per word.
Frequency discounts: 3 times, 10%; 6 times, 20%; 12 times, 33%.

FOR SALE

MCI input modules, $550.00 each. Tested and Guaranteed. **Paul. (312) 225-2110.**

AVAILABLE SERVICES. Milam Audio Co. specializes in every phase of professional studio wiring, from complete systems to individual pre-wired parts and components. Available from stock: patch bays, custom mic panels, multi-paired cabling and harnesses, etc. **Milam Audio Co., 1504 N. 8th St., Pekin, Ill. 61554. (309) 346-3161.**

CUSTOM CROSSOVER NETWORKS to your specifications; a few or production quantities. Power capacities to thousands of watts; inductors and capacitors available separately; specify your needs for rapid quotation. Also, PIEZO ELECTRIC TWEETERS—send for data sheet and price schedules. **TSR ENGINEERING, 5146 W. Imperial, Los Angeles, Ca. 90045. (213) 776-6057.**

PRO AUDIO EQUIPMENT & SERVICES

Custom touring sound, 2-, 4- and 8-track studios, disco systems. Representing Akai, AKG, Altec, Beyer, BGW, Cetec, Cerwin-Vega, Community Light & Sound, dbx, Dynaco, Dokorder, E-V, Gauss, Lamb, Langevin, 3M, Martex PM, Maxell, Meteor, Russound, Revox, Sennheiser, Shure, Sony, Soundcraftsman, Sound Workshop, Spectra Sonics, Switchcraft, TDK, TAPCO, TEAC, Technics, Thorens, and more. Offering these professional services: custom cabinet design, room equalization, loudspeaker testing, custom crossover design, electronics modification, and custom road cases. Call or write for quotes, or drop us a line for our latest catalogue. **K & L Sound, 75 N. Beacon St., Watertown, Mass. 02172. (617) 787-4073. (Att: Ken Berger)**

AGFA-GEVAERT. Professional PEM Mastertape, the international studio standard available from **TECHNIARTS, 8555 Fenton St., Silver Spring, Md. 20910. (301) 585-1118.**

NAB ALUMINUM FLANGES. We manufacture 8", 10½", & 14". For pricing, write or call **Records Reserve Corp., 56 Harvester Ave., Batavia, N.Y. 14020. (716) 343-2600.**

DON'T MISS THE "DISCO WAGON"! Excellent complete line of discotheque equipment from West Germany is available to fulfill your needs. Request your information package today. Write to: **D.T.S., Dept. DISCO, P.O. Box 16049, Seattle, Wa. 98116.** Reserve your territory in time!

FOR SALE: MELLOTRON 300-D; split keyboard; both keyboards may be played simultaneously. 12 selections on each keyboard. Total of 24 available sounds. $5,500. **E. Delaney, P.O. Box E-2, Altmamont, N.Y. 12009. (518) 861-5389.**

DUPLICATORS, blank cassettes, recorders, boxes, labels, cassette albums and supplies; lowest prices, top quality. Write for free brochure, "50 Tips for Better Duplication." **Stanford International, Box 546, San Carlos, Ca. 94070.**

MEASURE REVERB TIME IN REAL TIME—instantly! New, easy-to-use RT-60 delivers precise, instant real time digital readout. Eliminates chart recorder analysis. Only $460. Write: **Communications Co., Inc., 3490 Noell St., San Diego, Ca. 92110.**

NAGY SHEAR-TYPE TAPE SPLICERS

FOR CASSETTE ¼ & ½ IN. TAPES
- **HAND-CRAFTED**
- **FIELD PROVEN**
- **FAST, ACCURATE**
- **SELF-SHARPENING**

NRPD Box 289 McLean, Va. 22101

REPAIR

SCHOEPS (TELEFUNKEN) VACUUM TUBE CONDENSER MICROPHONES REPAIRED.
Original factory parts & factory calibration of capsules.
Models CM51, 61, 66; M201, 221, MK24, 26 etc.
ALBERT B. GRUNDY
64 University Pl., N.Y., N.Y. 10003
(212) 929-8364

SCHOEPS SCHOEPS

TAPCO MIXERS, graphic equalizers and reverbs at low, low cost. Specs and prices. Write **Daugherty Audio, 7313 Inzer, Springfield, Va. 22151.**

INFONICS DUPLICATORS! For a bunch of reasons, you can't afford not to consider Infonics Duplicators — especially since factory installation and training are included in the list price. **INFONICS DUPLICATORS, (219) 879-3381.**

FOR SALE: STUDER MODEL A80 4-track ½" tape recorder. Three years old, but with very low mileage on it. Only six months' use. The four electronics need general checkup but otherwise in excellent condition. Spare parts included. $5,000. Call **Tony Ochoa (809) 764-4440. Write Ochoa Recording Studios, Inc., GPO Box 3002, San Juan, P.R. 00936.**

1800 WATT ISOLATION TRANSFORMERS. $150, F.O.B. **Pragmatech Sound, 70 Sheldrake Pl., New Rochelle, N.Y. 10804. (914) 633-8556.**

TASCAM WARRANTY SERVICE STATION. Mixing consoles, $1,350; ½" recorders, $1,950; 8-track machines, $2,950. All shipped prepaid & insured, including free alignment + equalization + bias + calibration + life test. **Music & Sound Ltd., 11½ Old York Rd., Willow Grove, Pa. 19090. (215) 659-9251.**
Note Special Prices

FOR SALE: 3M M-56 8-track 1-in. w/remote, modified and rebuilt. Call **Westlake Audio. (213) 655-0303.**

CLASS D SWITCHING AMPS; B.B.C. reference monitors; pre-equalized J.B.L./Altec transducers; Nakamichi mastering cassettes; I.M.F. transmission lines; Ampex/Scully/Crown/Revox A-700 recorders/tapes; Micmix / Orban / Multi-Track reverbs; Eventide flangers / omnipressors; Parasound stereo synthesizers / parametrics; Lexicon digital delays; dbx/Burwen N.R. companders; Little Dipper hum/buzz notch filters; Cooper Time Cube echo send/doubler; moving coil Denon/Ortofon; B&O/Rabco straight line arms; Beyer condensers/ribbons; U.R.E.I. comp/limiters; White equalizers/filters;1,000s more. **Music & Sound Ltd., 11½ Old York Rd., Willow Grove, Pa. 19090. (215) 659-9251.**
Enclosure Designs Included—FREE

BODE FREQUENCY SHIFTERS
SINCE 1963

Featuring the universal model 735, described in March issue of db, and other special models, plus the line of Polyfusion synthesizer modules and equipment. For details, contact:

Harald Bode
Bode Sound Company
1344 Abington Place
N. Tonawanda, N.Y. 14120
(716) 692-1670

MODERN RECORDING TECHNIQUES by Robert E. Runstein. The only book covering all aspects of multi-track pop music recording from microphones through disc cutting. For engineers, producers, and musicians. $9.95 prepaid. **Robert E. Runstein, 44 Dinsmore Ave. Apt. 610, Framingham, Mass. 01701.**

AMPEX 300, 352, 400, 450 USERS—for greater S/N ratio, replace first playback stage 12SJ7 with our plug-in transistor preamp. For specifications, write **VIF International, Box 1555, Mountain View, Ca. 94042. (408) 739-9740.**

NEUMANN STEREO CUTTING SYSTEM: 2 LV-60 amps and 2 GV-2A feedback amps; one WV-2A feedback/monitor amp; one SI-A circuit breaker; one Ortofon 631 h.f. limiter. $3,000. **Paul (312) 225-2110.**

CERWIN-VEGA pro, full range speaker systems; bass horn, including the 2k rms Earthquake double-D horn; midrange horns; high frequency horns; musical instrument speakers; stage monitors; disco systems; power amps; equalizer and mixers with lifetime speaker warranty. A New England exclusive at **K & L Sound Service, 75 N. Beacon St., Watertown, Mass. 02172. (617) 787-4073. (Att: Ken Berger)**

FOR SALE, 16-track professional recording studio in Aspen, Colorado. MCI, Dolby, 3M, plus musical instruments. In business two years with excellent opportunity for future growth. **E. D. Thorne, Box 1498, Aspen, Colorado, 81611. (303) 925-5530.**

MCI JH416 CONSOLE

16 bus 24-output, included director's desk & additional cabinetry w/@62" of rack space. Included are twelve 440 modules & space for 12 more. Priced for quick sale. For more info, call Marty at **Colorado Nashville, Inc., (303) 473-1272.**

RADFORD

Order Radford direct from England! Immediate dispatch by air of HD250 stereo amplifier, ZD22 zero distortion preamp, Low Distortion Oscillator ser. 3, Distortion Measuring Set ser. 3, speakers and crossovers. Send for free catalogues, speaker construction plans, etc.

WILMSLOW AUDIO
Dept. Export DB, Swan Works, Bank Square, Wilmslow, Cheshire, England

MCI . . . DOLBY. Two great names! Two great products! For authorized factory representation in the progressive Midwest, contact: **Jerry Milam, Milam Audio Co., 1504 N. 8th St., Pekin, Ill. 61554. (309) 346-3161.**

AMPEX SCULLY TASCAM, all major professional audio lines. Top dollar trade-ins. 15 minutes George Washington Bridge. **Professional Audio Video Corporation, 342 Main St., Paterson, N.J. 07505. (201) 523-3333.**

WHATEVER YOUR EQUIPMENT NEEDS —new or used—check us first. We specialize in broadcast equipment. Send **$1.00** for our complete listings. **Broadcast Equipment & Supply Co., Box 3141, Bristol, Tenn. 37620.**

STUDIO SOUND—Europe's leading professional magazine. Back issues available from June '73 through June '75. $1 each, postpaid. **3P Recording, P.O. Box 99569, San Francisco, Ca. 94109.**

ORTOFON
DYNAMIC MOTIONAL FEEDBACK mono disc cutting systems. Complete with drive, feedback, and feedback-playback monitor amplifiers and cutterhead. All systems guaranteed. Spare cutterheads available for exchange/repair. **Albert B. Grundy, 64 University Place, New York, N.Y. 10003. (212) 929-8364.**

CROWN INTERNATIONAL. Complete repair, overhaul, and rebuilding service for current and early model Crown tape recorders and amplifiers. New and used machines bought and sold. **TECHNIARTS, 8555 Fenton St., Silver Spring, Md. 20910. (301) 585-1118.**

ONE STOP
FOR ALL YOUR PROFESSIONAL
AUDIO REQUIREMENTS
BOTTOM LINE ORIENTED
F. T. C. BREWER CO.
P.O. Box 8057, Pensacola, Fla. 32505

When precision quality is important!
AEG-TELEFUNKEN
TUBES
HARRISON ELECTRONICS
20 Smith St. Farmingdale, N.Y. 11735
(Dealers invited)

TEST RECORD for equalizing stereo systems. Helps you sell equalizers and installation services. Pink noise in ⅓ octave bands, type QR-2011-1 @ $18. Use with precision sound meter. **B&K Instruments, Inc., 5111 W. 164th St., Cleveland, Ohio 44142.**

Corners, handles, audio hardware. Tapco mixes and EQ's, Shure SM mics. Professional Audio gear at Big Savings. Catalog 50¢. **Headtronix, Box 31012, Dallas 8, Texas 75231**

Neumann recording console, 18 input, $14,000 (originally $35,000). Scully 8-track with remote control, can be expanded to 12 tracks, $7,000. Neumann lathe with Westrex 2-B mono system plus accessories, reasonable. Pentagon cassette duplicator, reel/cassette, cassette/cassette, $1,000. Ampex AG-500 stereo, $1,250. Ampex PR 10 stereo, $600. **Paul. (312) 225-2110.**

AMPEX 440B 4-track, low usage, $4,000/offer. **Jeff (315) 463-8631.**

CUSTOMIZED TUNED ROCK P.A.'s

Expandable high intensity touring/permanent sound systems; including narrow band (**5Hz!**) feedback and ring mode suppression, detailed regenerative response environmental equalization (**±1 dB at your ears**), room design,/measurement / treatment / our engineered enclosures plus 18 dB crossovers, ∠15% articulation loss of consonants; **1,000s** of customized professional products including: fiberglass horns, consoles, comp/rms/peak limiters, continuously variable electronic crossovers, digital/acoustic/analog delays, omnipressors, flangers, reverb, echo, doubling/tripling, p.a. noise reduction, piezo transducers, frequency shifters, notch filters **from** . . . J.B.L./Altec pro, Tascam, U.R.E.I., Eventide, Gately, Beyer, Crown, Community, Mom's Audio, McIntosh, Allen & Heath, Cetec, Multi-Track, Orban, White, etc. All shipped prepaid + insured. **Music & Sound, Ltd., 11½ Old York Rd., Willow Grove, Pa. 19090. (215) 659-9251.**

Anechoic Chamber 0.9a Working Floor
Inventors/Engineers

(continued)

FOR SALE

A FEW competitively priced used Revox A77 decks available. Completely reconditioned by Revox, virtually indistinguishable from new and have the standard Revox 90-day warranty for rebuilt machines. Satisfaction guaranteed. Example, A77 with Dolby, $675, plus shipping. Write requirements to **ESSI, Box 854, Hicksville, N.Y. 11802. (516) 921-2620.**

16-TRACK AMPEX MM-1000, excellent condition. Converted to 15-30 w/input switching on s/s and auto. Dolby switching. $13,500. Call **D. Frey, A&R Recording, (212) 582-1070.**

REELS AND BOXES 5" and 7" large and small hubs; heavy duty white boxes. **W-M Sales, 1118 Dula Circle, Duncanville, Texas 75116. (214) 296-2773.**

AMPEX TAPE. Ampex Audio Studio Mastering Tapes, 631-641, 406-407, and "Grand Master" in stock for immediate shipment, ¼", ½", 1", and 2", factory fresh. Best prices, BAC charge cards accepted. **TECHNIARTS, 8555 Fenton St., Silver Spring, Md. 20910. (301) 585-1118.**

THE LIBRARY . . . Sound effects recorded in STEREO using Dolby™ throughout. Over 350 effects on ten discs, $100.00. Write, **The Library, P.O. Box 18145, Denver, Colorado 80218.**

ARP SYNTHESIZERS! Strings, $1,385; 2600, $2,260; Axxe, $730; Prosoloist, $875; Odyssey, $1,165. **Dickstein Distributing, 1120 Quincy, Scranton, Pa. 18510.**

FOR SALE: 20 x 16 QUAD recording console. Monitor mix, 2 cues, 4 echo, solo, mute, 2 joy sticks, patch bay and producer's desk. Expandable. $18,000—will finance. **Call Tom (213) 783-9235 or Bill (213) 462-8515.**

FREE CATALOG & AUDIO APPLICATIONS

OPAMP LABS INC.
1033 N. SYCAMORE AVE
LOS ANGELES, CALIF. 90038
(213) 934-3566

CONSOLES
KITS & WIRED
AMPLIFIERS
MIC., EQ, ACN,
LINE, TAPE, DISC,
POWER
OSCILLATORS
AUDIO
TAPE BIAS
POWER SUPPLIES

WE WILL BETTER anyone's price on new Recordex high speed cassette duplicators. Your written request can save you a bundle. Also get our large-user cassette deal. **Tape and Production Equipment Company, 2080 Peachtree Industrial Court, Atlanta, Ga. 30341. Phone (404) 458-TAPE.**

SMALL 4-16 TRACK STUDIOS. Detailed technical assistance + acoustical consultation, from our engineering division to our clients—either here or via phone, & included FREE. **Music & Sound Ltd., 11½ Old York Rd., Willow Grove, Pa. 19090. (215) 659-9251.**

THE ONLY ONE

FOR SALE: LARGE MOOG with two keyboard controllers, one ribbon controller and custom cabinet, $4,950. **Woodland Sound Studios, Nashville, Tenn. (615) 227-5027.**

CASSETTE LABELS, 1,000 on typewriter roll, $10.98 ppd. Other formats and custom labels. **TARZAC, 638 Muskogee Ave., Norfolk, Va. 23509.**

MCI 24 x 24 console with extras. Also MCI 16-track tape recorder. Will sacrifice. **Sound Room, 300 S. 69th St., Upper Darby, Pa. 19082. (215) 734-1117.**

3/M 16-track tape recorder, M-56, $14,500. Perfect. **Paul. (312) 225-2110.**

WIRELESS MICROPHONES. Professional hand held and lavalier wireless microphones, new and used. **EDCOR, 3030 Red Hill Ave., Costa Mesa, Ca. 92626. (714) 556-2740.**

MOVING?

Keep **db** coming without interruption!

Send in your new address *promptly*.

Enclose your old **db** mailing label, too.

Write to:

Eloise Beach, *Circ. Mgr.*
db Magazine
1120 Old Country Rd.
Plainview, N.Y. 11803

PATCH CORDS, new 3 ft. Switchcraft tip-ring-sleeve (PJ-051R plugs). Commercial overstock makes these available at $5.25, which is below dealer cost. No minimum, dealers welcome, satisfaction guaranteed. Shipped prepaid or C.O.D. NOTE: NEW BOX NUMBER. (Post Office error sent letters back, please write again.) **Kapes Audio Supply, P.O. Box 5045, River Station, Rochester, N.Y. 14627.**

CREATIVE CASSETTE & CARTRIDGE LABELS. Custom designed; small and large runs; cassette, cartridge duplication. **Omega Audio, 25520 Graham, Detroit, Mich. 48239.**

CASSETTE WINDERS: Ramko Research ACL-25's; excellent condition; $256 both. **TARZAC, 638 Muskogee Ave., Norfolk, Va. 23509.**

MAXELL TAPE. All widths. Write NOW and SAVE! **N.A.B. Audio, Box 7, Ottawa, Ill. 61350.**

DISCO MIXERS. Write for free bulletin. **Berkshire Audio Products, P.O. Box 35, Great Neck, N.Y. 11021.**

$2 MILLION USED RECORDING EQUIPMENT. Send $1.00 for list, refundable, to **The Equipment Locator, P.O. Box 99569, San Francisco, Ca. 94109.**
94109.

3M SERIES 79 1" 8-track headstacks, mint condition, $1,400 or best offer. **(614) 663-2544.**

DECOURSEY ACTIVE ELECTRONIC CROSSOVERS. Model 110 dividing network; complete with regulated power supply, for bi-amp, tri-amp, or quad-amp. Custom assembled to meet your specifications. Monaural, stereo, or with derived third channel. Plug-in Butterworth (maximally flat) filters; 6, 12 or 18 dB per octave at any desired frequency. OPTIONS: Summer for single woofer systems, VLF hi-pass filters for elimination of subsonic noise, derived third channel. FOR OEM OR HOME ASSEMBLERS: Model 500 or 600 dual filters. Regulated power supplies. Write for new brochure. **DeCoursey Engineering Laboratory, 11828 Jefferson Blvd., Culver City, Ca. 90230. (213) 397-9668.**

Two can ride cheaper than one.

Ad Council

A Public Service of This Magazine & The Advertising Council

DYNACO RACK MOUNTS for all Dynaco preamps, tuners, integrated amps. $24.95 postpaid in U.S., $22.50 in lots of three. **Audio by Zimet, 1038 Northern Blvd., Roslyn, N.Y. 11576. (516) 621-0138.**

UREI SONIPULSE w/AKG condenser microphone. $1,000, also Allison Kepexes, API 525 Limn/Compressors, JBL 2390 lenses in good condition. **Bursting Sun Sound Services. (413) 863-4356.**

ELECTRO-VOICE SENTRY PRODUCTS. In stock: Sentry IV-A, Sentry III, and Sentry II-A monitor loudspeaker systems for professional monitoring and sound reinforcement. Immediate air freight shipment to any N. American destination. **National Sound Corp., Ft. Lauderdale, Florida. (305) 462-6862.**

FREE CATALOG of studio kits, consoles, p.a., discrete opamps. **QCA, Box 1127, Burbank, Ca. 91507.**

CENTRAL MUSICIANS' SUPPLY. Instruments and sound equipment for the professional. All major brands; sound reinforcement systems custom designed; home studio recording equipment. See our pricelist before you buy. **DJ's Music Limited, 1401 Blanchan, La Grange Park, Ill. 60525. (312) 354-5666.**

WANTED

WANTED: ALTEC #633 microphone. **Dept. 51, db Magazine, 1120 Old Country Rd., Plainview, N.Y. 11803.**

WANTED: SONY ECM-22, ECM-377, C-57, C-37A, C-37P, ECM-64P, ECM-65P, C-22, C-500 microphones and console cabinets for TEAC 7030 SL or 7030 GSL recorders and remote controls. **Blue Diamond Co., Box 102C, Chubbic Rd., R.D #1, Canonsburg, Pa., 15317. (412) 746-2540.**

EMPLOYMENT

WANTED: AUDIO ENGINEER/TECHNICIAN/JACK-OF-ALL-TRADES. Resumes only. **Bradley Recording Co., Inc., 531 N. Howard St., Baltimore, Md. 21201.**

SPIRIT-FILLED recording engineer/technician for 16-track rural recording studio. Hard work, long hours. Must dig rock/country. Contemporary Christian ministry. **(614) 663-2544.**

CHIEF ENGINEER
AUDIO PRODUCTS

We are seeking an audio equipment engineer who is an aggressive and innovative designer—one who has designed consoles or components for consoles used in recording and broadcasting applications.
If you are this creative and product-oriented individual, looking for a rewarding career, we would like to talk to you.
Please send your accomplishments and salary history to:

Modular Audio Products
1385 Lakeland Ave.
Bohemia, N.Y. 11716

RECORD COMPANY established 1953 seeks freelance recording engineer associates in all major cities, colleges, etc. Write for very interesting offer. **Educo Records, Box 3006, Ventura, Ca. 93003.**

EUROPEAN ORGANIZATION requires General Manager for U.S. outlet (New York area). Outstanding opportunity for competent sales engineer to participate in establishing a new corporation for the distribution of recording equipment. Good understanding of tape equipment and of the recording industry essential. Send resume in confidence to **Dept. 52, db Magazine, 1120 Old Country Rd., Plainview, N.Y. 11803.**

MUSIC FACULTY OPENING. The UCSD Department of Music announces the following academic opening for 1976-77: A recording specialist to teach undergraduate and graduate courses in recording techniques, possibly graduate course every other year in tuning and temperament, and supervise the departmental archiving and tape duplication. Applicants should have a strong musical background; BA or MA in music desirable. The UCSD Department of Music is committed to an active, experimental program of contemporary music. All resumes, appropriate scores, and other documentation should be sent to: **Search Committee; Music Department (BO26), UCSD, La Jolla, Ca. 92093.** The University of California, San Diego, is an Affirmative Action/Equal Opportunity Employer. Women and minorities are encouraged to apply.

EXPERIENCED MUSIC MIXER

For major N.Y.C. studio, expanding staff. Send resume to **Box 11, db Magazine, 1120 Old Country Rd., Plainview, N.Y. 11803.**

Sabor presents
the
MK-668C
wow and flutter meter

The MEGURO MK-668C capable of measuring DIN, IEC & ANSI at 3.15 kHz and JIS, NAB & CCIR at 3.0 kHz is truly a world-wide universally applicable Wow & Flutter Meter. It features selectable calibration to permit reading of peak, average or effective values of W/F and tape speed error is indicated on the built in digital frequency meter.

W/F range is 0.003% to 10% at inputs above 30mVrms and 0.01% to 10% with inputs from 0.5mV to 30mVrms

Price is $975

12597 Crenshaw Blvd
Hawthorne CA 90250
(213) 644-8689

Sabor corporation

Circle 38 on Reader Service Card

EXPLORE SOUND!
...and meet our researcher Dennis Colin.
Dennis has been exploring sound for more than twenty years. He plays gigs with "Family Affair", and his 1970 paper published by the Audio Engineering Society of America is considered classic in the field of electronic music.

Dennis' special interest these days is in finding new ways to modify both instrumental and vocal sounds. Best of all, he helped design the ARIES SYSTEM 300 ELECTRONIC MUSIC SYNTHESIZER. Let us tell you about it - just send the coupon for full information. (Also available in kits for those who would like to save money by building their own).

( ) Yes, I would like to explore sound. Please send free information on the ARIES SYSTEM 300 ELECTRONIC MUSIC SYNTHESIZER. Full information on kits too, please.

( ) I would like a demonstration cassette of the SYSTEM 300. I enclose $5.00 to cover costs.

NAME..

ADDRESS..

....

SEND TO: ARIES, Inc., 119 Foster Street Peabody, Mass., 01960 (617) 532-0450

Circle 35 on Reader Service Card

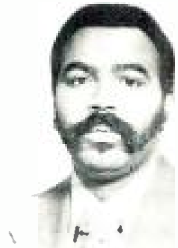
db people/places/happenings

SZEGHO

FAROOQ

COLODNY

TRUEMAN



DUNN

DILLEY

• Well-known electronics engineer **Dr. C. S. Szegho** has been appointed to the board of directors of the **Rauland-Borg Corporation** of Chicago. Dr. Szegho. who has been associated with Rauland-Borg research, pioneered many advances, first in radar tube design and later in t.v. picture tube development. He is now serving as a consultant to the **Zenith Radio Corporation.**

• **Abid Farooq** has been appointed senior project engineer for single sideband product development at **Coastcom,** of Concord. California. Mr. Farooq will be responsible for SSB program channels for use with both satellite and CCITT FDM networks.

• **Dr. Peter C. Goldmark,** president of **Goldmark Communications Corp.** of Stamford. Connecticut, has been granted a patent for a new video learning system to be used with home t.v. sets. The new system. called Rapid Transmission and Storage Mark II. makes it possible to transmit pictures and sound at extremely high speeds for broadcasting by satellite or cable t.v. and for storage and playback over home television sets. The system can provide 60 different half-hour programs from a single hour-long videotape. Up to 30 of the programs can be selected from the single tape and shown simultaneously on different t.v. sets.

• **Omega State Institute** has moved to a new location at 237 E. Grand Ave., Chicago. Ill. The school offers trade school facilities in broadcasting. FCC license training, and electronic alarm systems.

• Serving the Midwest, **Tom S. Butler** has been appointed Central sales manager by **McMartin Industries, Inc.** of Omaha. Nebraska. Mr. Butler's responsibilities will include broadcast. engineered sound and background music product sales. He comes to McMartin from the **Collins Radio Group, Rockwell International.**

• Two new personnel changes have taken place at **CCA Electronics Corporation,** of Gloucester City. N.J. **Samuel H. Colodny** has been promoted to director of engineering, and **A. W. (Bill) Trueman** as director of marketing. Mr. Trueman came to CCA from **RCA** in 1974. Mr. Colodny. who holds several patents for his inventions, was previously with **AEL.**

• A new nickel/steel alloy for magnetic heads was unveiled at the NAB Show in Chicago by **Nortronics.** The alloy. a high permeability magnetic substance named Wear-Resistant Hy Mu 800™, was developed by **Carpenter Technology,** of Reading, Pa. The long-wearing heads are projected as a boon to high use applications. such as broadcasting.

• Overseas orders from Egypt. Malaysia. and Hong Kong have been received by **Rupert Neve and Co.** of London. The **Cinema Organization** of **Cairo** ordered an 8036. 24-channel. 16-track music recording console. **Film Malaysia** has requested a BCM 10/2 sound control console and **Radio Hong Kong** has placed an order for three BCM 10/2 consoles.

• Swiss based **Eastlake Audio,** recently founded by **Tom Hidley,** president of **Westlake Audio Inc.,** of Los Angeles. has commenced its operations from Montreux. Representation for the corporation in Scandinavia and the United Kingdom will be **Scenic Sounds Equipment. Kent Duncan** of **Sierra Audio,** Burbank. Ca. will represent American interests. Additional dealers are **3M France,** and **Studer International** in Zurich. Milan. and Tel Aviv.

• The **Rectilinear Research Corporation** of the Bronx, N.Y. has named **Claude Dunn** as Metropolitan New York sales manager. **Mr. Dunn** comes to Rectilinear from the **Sony Corporation.**

• **William G. Dilley,** president of **Spectra Sonics,** of Ogden. Utah. has been selected for inclusion in **The International Register of Profiles.** Mr. Dilley is a Fellow of the **Audio Engineering Society,** a Fellow of the **Intercontinental Biographical Association,** holder of fourteen U.S. and foreign patents and is the holder of numerous aircraft speed records.

• Assigned to the design of circuits and products in the areas of broadcast control room equipment and small broadcast transmitters, **Edward M. Corse** has joined **LPB Inc.** of Frazer, Pennsylvania as staff engineer. Mr. Corse was formerly with **U. S. Electronic Services.**

• **Cindy Guzzo** has been named marketing manager at **Pacific Recorders & Engineering Corporation** of San Diego. California. Ms. Guzzo will be responsible for sales and marketing. She was formerly with **La Salle Audio.**

• **The Plastic Reel Corporation** has established a new customer service department which will work with audiovisual suppliers to develop methods of packaging. storage. and handling new tape products. The company is based in Carlstadt. N.J.

Copies of all issues of **db—The Sound Engineering Magazine** starting with the November 1967 issue are now available on 35 mm. microfilm. For further information or to place your order please write directly to: **University Microfilm, Inc.**
300 North Zeeb Road
Ann Arbor, Michigan 48106

We added your inputs to ours.

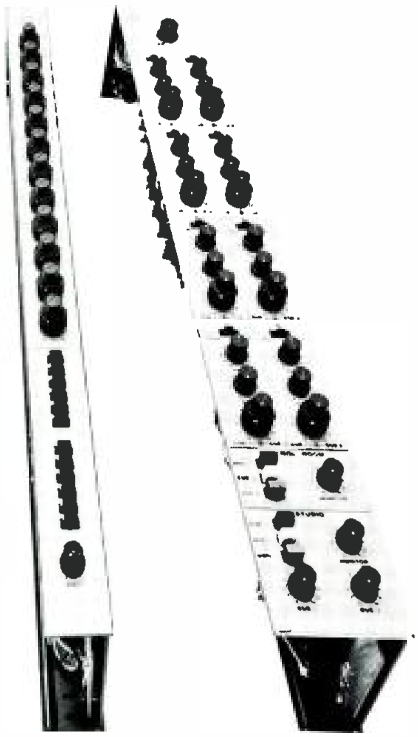
The result is the Model 10B—a good thing made better. Now there are peak reading LED indicators on each input, chassis mounted ¼″ phone jacks for added stability, and the echo busses can be used with the program busses for 8-out capability.

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

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

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

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



Model 120 Model 116

Model 10B

TEAC
TASCAM SERIES

TEAC Corporation of America, 7733 Telegraph Rd., Montebello, CA 90640
© TEAC 1976

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 11 on Reader Service Card