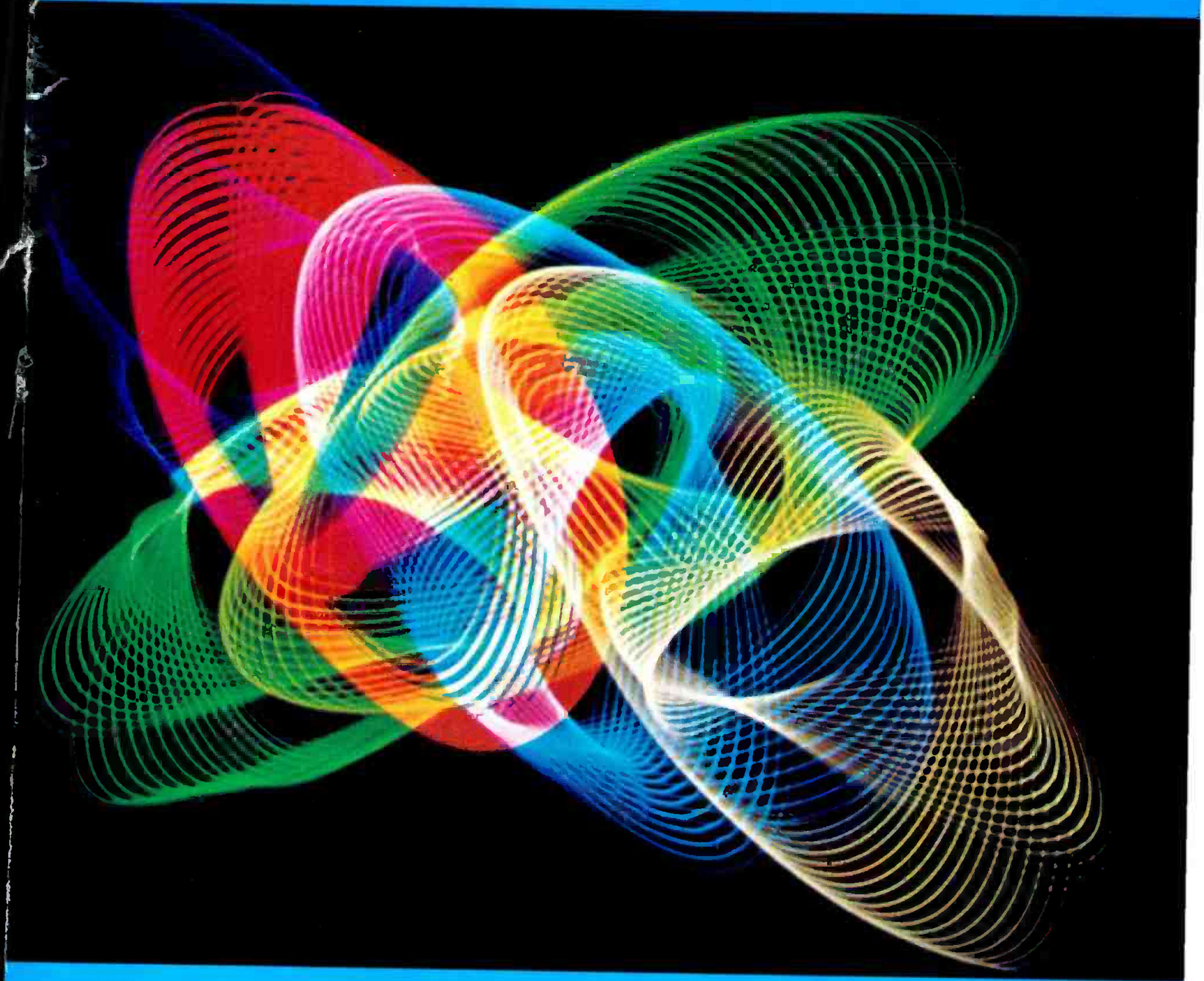
THE SOUND ENGINEERING MAGAZINE

JULY 1981
$1.95

db



SHURE

StarMaker™ SERIES

SM77•SM78 cardioid dynamic microphones

a new look for the 80's

There's so much new about these professional microphones we can only touch the highlights. Exclusive "picture perfect" SUEDECOAT™ Tan or Ebony matte finish looks great—*permanently.* Significantly smaller, exceptionally light in weight; yet so extraordinarily rugged and reliable we call them the "Light Heavyweights." They feature the crisp sound that makes Shure the world's most widely used professional performer microphones.

® GENUINE SHURE ®

For more information write:

Shure Brothers Inc., 222 Hartrey Ave., Evanston, IL 60204
In Canada: A. C. Simmonds & Sons Limited
Manufacturers of high fidelity components, microphones, sound systems and related circuitry.

Circle 10 on Reader Service Card

sales offices

Roy McDonald Associates, Inc.
Dallas, Texas 75237
First Continental Bank Bldg.
5801 Marvin D. Love Freeway
Suite 303
(214) 941-4461

Denver, Colorado Area
Englewood, Colorado 80112
14 Inverness Dr. East, Bldg. 1—Penthouse
(303) 771-8181

Houston, Texas 77036
6901 Corporate Drive, Suite 210
(713) 988-5005

Los Angeles Area
Glendale, California 91204
424 West Colorado Street, Suite 201
(213) 244-8176

Portland Area
Hillsboro, Oregon 97123
510 South First
P.O. Box 696
(503) 640-2011

San Francisco Area
Emeryville, California 94608
265 Baybridge Office Plaza
5801 Christie Avenue
(415) 653-2122

Sagamore Publishing Co., Inc.
New York
Plainview, NY 11803
1120 Old Country Rd.
(516) 433-6530

ABOUT THE COVER

• In case you don't immediately recognize it, our cover illustrates a real-time, kinetic X-Y plot of the delay interval of a slowly swept sine wave. Our thanks to Lowell Cross for the photo, and for his feature story on The Audio Control of Laser Displays, in this issue of **db**.

db

THE SOUND ENGINEERING MAGAZINE
JULY 1981 VOLUME 15, NO. 7

db is listed in **Current Contents: Engineering and Technology**

db, the Sound Engineering Magazine (ISSN 0011-7145) is published monthly by Sagamore Publishing Company, Inc. Entire contents copyright © 1981 by Sagamore Publishing Co., 1120 Old Country Road, Plainview, L.I., N.Y. 11803 Telephone (516) 433 6530 db is published for those individuals and firms in professional audio-recording, broadcast, audio-visual, sound reinforcement, consultants, video recording, film sound, etc. Application should be made on the subscription form in the rear of each issue. Subscriptions are $15.00 per year ($28.00 per year outside U.S. Possessions, $16.00 per year Canada) in U.S. funds. Single copies are $1.95 each. Editorial, Publishing and Sales Offices: 1120 Old Country Road, Plainview, New York 11803. Controlled circulation postage paid at Plainview, NY 11803 and an additional mailing office.

db Letters

MORE FEEDBACK ON THE DIGITAL QUESTION

To The Editor:

Re: The digital/analog controversy. In the past 25 years, each major new electronic development to be utilized by the audio industry was accompanied first by a setback in sound quality, and then by gradual improvement. After approximately 10 years of improvement, the new development eventually gained widespread acceptance, even by the "golden ears."

For example, in the 1960s one of the first companies to convert to (discrete) transistor technology was Langevin. That company's first transistor products were very harsh sounding and prone to failure as well. In their rush to be the audio industry "firsts," Langevin did not survive the transistor revolution. Between 1960 and 1970 (and beyond), the quality of transistors and transistor circuits improved enough so that the subjective "sound" of the mixing boards of that era eventually became quite good. The "tube" holdouts were fewer and fewer as the years went on.

In 1969 approximately, Melcor came out with the first (discrete yet modular) plug-in opamp. RCA, Automated Processes, and Aengus, among others, were the first to use these opamp modules exclusively in their consoles. The sound of these opamps was not bad, yet to my ears they were harsh to the extent that high-frequency brass instruments (especially muted trumpets) came out sounding edgy and fatiguing. (Shades of Dr. Diamond?) Note that Melcor and Aengus are no more, and Automated Processes exists in a highly-altered state (somewhere in Georgia, perhaps?).

Anyway, today, Deane Jensen has designed retrofit plug-in Opamp modules for the Melcor and Automated Processes products whose sound literally runs rings around the old sound. In ten years again, a fabulous improvement has taken place. Another development, also around 1969, was the integrated circuit opamp. Everyone rushed to put 301s and 741s into their consoles. Today those same people are considering replacing the old ICs with high-speed Bifet replacements—because they too sound much better.

Similar events occurred with the advent of the VCA and recently transformerless input (and output) amplifiers. It is noteworthy that one British manufacturer of consoles, highly respected for their sound quality, continues to avoid integrated circuits, opamps, VCAs, and FET or CMOS switches in their console audio chain.

Index of Advertisers

Coming Next Month

• In August, our featured topic is loudspeakers. For readers who are not satisfied with store-bought systems, author John Hoge goes through the mathematics of do-it-yourself design. And UREI's Dean Austin describes some redesign problems (and their solutions, of course) the company faced as it updated its 800 series of studio monitors. And, as a change of pace, Sidney Silver returns to tell us about digital filtering.

Pick a number from 9 to 52!

You've just chosen the ideal DC voltage to phantom-power these new ATM electret microphones.

Introducing four "universal" phantom-powered electret microphones. Designed to work from external power, internal regulation automatically handles any voltage from 9 to 52 VDC without adapters, switches, or rewiring. Just plug in and enjoy. With current drain a mere 0.3 mA at 9 volts (4 mA at 12-52V) a 9V battery lasts thousands of hours, not just the 60 or 70 hours typical of other mikes.

When your power supply isn't available, or isn't enough, use ours. The new AT8501 Dual Battery Supply holds two 9V batteries. One to use, and one in reserve. Instant switchover and test LED eliminates guesswork. And spares are as near as the closest shopping center. Neat!

But convenience and versatility are just two of the advantages of the new ATM models. All-new electronics provide plenty of headroom inside the microphone with no more than 1% THD even when used in acoustic fields of 141 dB SPL. Which sets new standards for clean sound even close-up to big brass or inside a powerful drum kit.

And the sound you hear is wide-range and *musical*. Presence without peaks. Highs to 20,000 Hz but without a raspy "edge." Yet despite their responsiveness, these new ATM microphones have the "Road Tough" reliability proved so often on stage and in the studio.

Before you add another microphone, compare our sound, our convenience, our reliability, and our cost. Write for literature and list of nearby ATM microphone specialists. Get great sound…right from the start! AUDIO-TECHNICA U.S., INC., 1221 Commerce Drive, Stow, Ohio 44224. (216) 686-2600.

Circle 30 on Reader Service Card

audio-technica®

All of these devices are undergoing constant development and improvement. Is it not going to be the same with digital audio? For example, at present, no digital recording system can meet the slew rate and power bandwidth specifications of an NE5534 opamp. Slew rate and power bandwidth go hand-in-hand, with, *in my opinion,* audio quality.

The same golden ears who kept us with tubes until the late '60s, and discrete transistors until just about now (when the quality of IC opamps is almost universally accepted), will hopefully keep an eye (and an ear) open to the sound of digital.

Unfortunately, the monetary stakes today are much higher than in the past. I sincerely hope that no company in the 80s will fall into the sad fate of Melcor and Langevin only because they jumped on the digital bandwagon much too soon.

BOB KATZ
Recording Engineer
New York City

TO THE EDITOR:

Concerning the controversy(?) surrounding digital recording vs. stress factors, I feel obligated to submit my thoughts. The advantages of digital recording to the audio professional, are obvious. The case for digital recording stress is *not* obvious, but that should not negate its possible merit. What is needed, *if* the stress theory is true, are the reason(s) *why* it is true. The fundamental way in which digital recording differs from analog is the fact that there *are,* however minute and miniscule, small *time* lapses between bits (the analog process, being an essentially *constant* process, and the digital being a broken one). The human brain, being the most advanced computer ever made (look at the maker!), can surely detect these differences. *If* this difference in processes *could* cause stress, this opens up an enormous question, and possibly an entire field of research. All people receive stimuli via the five senses: sight, hearing, smelling, taste and touch. How do these five senses respond to different translations of the *same* stimuli? Consider the difference between video and film. Both operate on the frame principle (there *are* time lapses between frames), but in many circles, video is considered inferior to film. Why? The answer lies in *resolution.* Video is constantly striving for better resolution, i.e., scanning at 1,125 lines instead of 525 (as explained in the "Sound With Images" article on the NHK system). Each frame of film is a complete and whole picture, while each frame of video is made of broken lines. Does one produce stress over the other? Hell if I know, but it is food for thought.

How do the other senses respond to constant vs. broken stimuli? This must be a consideration in any research of sensory stimuli. Another difference between

WOW ...it's dyna-mite™

That's what they said when we introduced DYNA-MITE at the 69th AES Convention in LA.

Why the surprise?

Dynamic manipulation—i.e., limiting, expansion, noise gating, de-essing, etc.—has been around for some time. It's ironic, in view of today's rapid pace technology, just how unsophisticated a collection of such devices exits in the marketplace. Limited function...marginal dynamic integrity...higher and higher prices. Rehashed circuits from a decade ago.

Suddenly, DYNA-MITE has struck...opening a fresh new page in the journals of audio processing.

Why all the fuss? Just because DYNA-MITE performs *all* the standard dynamics processing functions—including limiting, expansion, de-essing, KEPEXING®, gating/ducking—plus a few new ones (18 functions in all)? Or is it because it offers two independent or stereo channels, self-powered and contained in a briefcase sized, yet rack mountable package? Perhaps because it's downright affordable, even for semi-pros?

No, these are only the icings. DYNA-MITE's real forte lies in its fundamentally superior performance. Listenability which surpasses the very best of the single function devices...in any mode.

It must be stressed that DYNA-MITE employs proprietary VALLEY PEOPLE circuit principles not available from other sources; important engineering achievements, not gimmicks. The result is that DYNA-MITE performs sensationally in situations where other devices cannot even be used without severe dynamic distortions. By comparison, there is no comparison.

Only your ears can fully describe the DYNA-MITE experience. We warn you, however, not to listen to a DYNA-MITE unless you're fully prepared to buy it on the spot. It's that good.

DYNA-MITE may be ordered as a 1-channel, or Mono Unit—Model #410-1...$295.00.

DYNA-MITE may be ordered as a 2-channel, or Stereo Unit—Model #410-2...$495.00.

A single DYNA-MITE channel may be ordered to upgrade a mono unit to a stereo unit—Model #410-C...$225.00.

A rack-mount "shelf" may be ordered as Model #400-R. It will mount one or two DYNA-MITE units. $40.00.

VALLEY PEOPLE, INC.
P.O. Box 40306 • 2820 Erica Place
Nashville, TN 37204 • 615-383-4737
TELEX 558610 VAL PEOPLE NAS
a merger of **Allison Research & Valley Audio**

digital and analog recording is the ability to transmit overtones and harmonics above 20 kHz. Many experts argue that the human ear cannot perceive signals above 20 kHz (but), how do we know? (See *Theory and Practice*, "Instrumentation: For Machines and People," April, 1981.) Although the *conscious* ear and mind may not respond to signals above 20 kHz, or perceive the difference between broken lines and whole pictures, we all know the ability of the subconscious.

While I do not purport to know any answers to these questions, and do not really support either side of "the controversy," I, along with many others, would like to *know* the answers before making any investment in "the wave of the future."

David Moore
The Rock Studio
Norman, OK

To The Editor:

After reading several of the technical analyses recently written explaining the "obvious merits of digital vs. analog" recording systems, I have this sinking feeling about the future of popular recordings. The attitude taken is similar to the kind of thinking that has made the nuclear industry so attractive to technical people. On the average, one of the prime attractions for technical types are "toys." The solar industry has few such toys to attract engineers and technicians (not to mention politicians) and that has been one of its most serious stumbling blocks. The recording world has been equally fascinated with the toys of digital recording, neglecting obvious drawbacks such as low-frequency distortion, high-frequency modulation, low-signal distortion, and non-existent slew specifications. In an industry that puts more value in bells-and-whistles than in function, status rather than quality, and recreation rooms, saunas, mini-spoons, and Indian rugs over fidelity, this is understandable but no less excusable. Dynamic range means little if a significant part of the signal is distorted and even less in an atmosphere of heavy compression and general equipment misuse. I think even the manufacturers of this equipment (who now have a large investment at stake) will admit digital recording equipment will be far less forgiving in terms of tape saturation, which some of the better known and lesser talented recording types use quite regularly as an "effect."

I think the true scientific quality of this profession is accurately indicated by the fact that the two most quoted references for acoustic research and education (Beraneck's *Acoustics* and *Acoustical Engineering*) are, and have been for some time, out of print. This, plus the fact that many people calling themselves "recording engineers, sound engineers, and audio engineers" could only qualify for the title of engineer at one of today's highly-useless railroads. Of course, this does make the creation of new and more exciting theories much easier, since we are no longer hampered by the laws of nature and physics.

Tom W. Day
P.O.Box 201
Scribner, Neb. 68057

To The Editor:

I found the article by Dr. Diamond concerning therapeutic effects of music (**db**, January 1981) most interesting. That music is a factor in controlling emotions is evidenced all around us. Soft background music in places of business is supposed to enhance purchasing; certain types of music are selected for specific times of day, i.e., rousing in the morning, soothing in the evening. We have noticed, also, that rock concerts often erupt into riots. But what of the digital process?

There are at least two simple explanations for Dr. Diamond's findings. The digital process is used to reduce the overall noise level inherent in the analog recording process. In the analog system, the noise is omnipresent, but appears to diminish and increase randomly with the dynamic range of the music. That noise has some therapeutic value has been known for some time. Several years ago sleep boxes were being marketed widely. These were really noise generators, some producing "surf" and others producing animal, bird, brook, hiss or humming noises. It is possible, then, that the music provides a pleasant means of listening to the noise in the analog system, and that the noise, rather than the music, is the source of therapy. Suppose Dr. Diamond were to conduct some simple tests with the digital recordings. Mix the recording with the output of a white noise generator having a level set about 14 to 20 dB down from peak music amplitude. Try the same experiment with noise only, with "pink noise" and maybe with a low-pitch hum, all at a very low level.

There is a second possible explanation which would be more difficult to test. The digital process is one in which the music is sampled at a rapid periodic rate, and these samples are recorded. Effectively, the sound is pulsed at an extremely rapid rate, but the mind might be able to detect that "on again, off again" pulsing and reject it.

The tests using noise could be very simply done, and might very well provide some interesting avenues for further research.

E. Neil Pike
Broadcast Engineer
Radio W S I E

db replies:

Well, we still don't think Dr. Diamond has "found" anything, except perhaps an

FOR GREAT RECORDING, UP AGAINST THE WALL!

What really blows many recording engineers off their chair is the sonic *beauty*, the *accuracy*, the *reality* of the PZMicrophone.™

It's all summed up in one engineer's comment: "It's as if my ears were *in* the studio!"

The PZM™ will give you fidelity you never hoped to achieve – with minimal miking.

For instance, the evidence to date indicates that the best way to capture all the magic of a top vocalist is to have him sing to a PZMicrophone mounted on the wall. The physics of the PZM indicate that this could develop some profoundly improved recordings.

Try your next bluegrass group (or other small combo) with a PZM (or two for stereo) on the floor. Some significant trials indicate that you'll hear sound qualities hard to achieve with other miking systems.

If your current miking set up is losing something, you owe it to yourself – and to the artist – to try PZM.

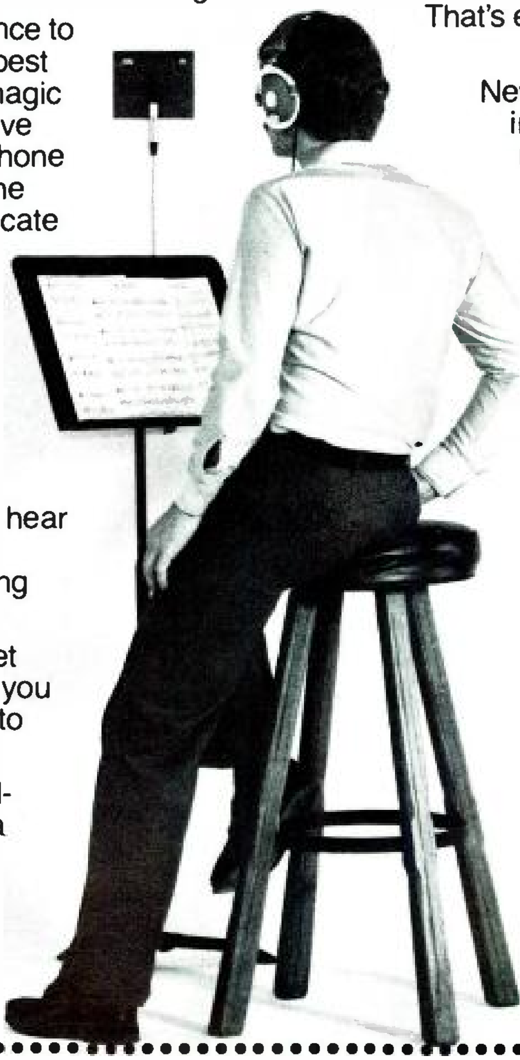
One of the most rewarding benefits of owning a PZMicrophone is the feeling of liberation, of newness, that comes with this small package.

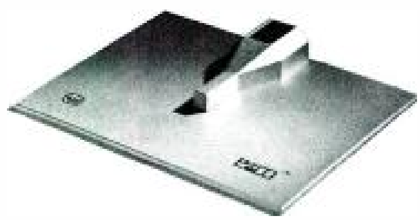
There is no textbook for miking with PZM. Those who own them are writing the textbook. With PZM, you are suddenly freed from the old rules, free to discover the sound of today, and the techniques of tomorrow.

That's exciting. It's also disturbing, and maybe a little scary.

New technology is always disturbing, but its rewards can be enormous. You know how important those rewards can be in your profession. We'd like to help you earn them.

A number of engineers have already tried PZMicrophones in a wide variety of situations, and have shared their opinions of the results with us. We've published all these ideas in PZM APPLICATION NOTES. A free copy is yours for the asking. It could be a key to unlock more of your creativity.





db7

crown®

...WHEN YOU'RE READY FOR REAL!

Send to: **CROWN INTERNATIONAL**
1718 W. Mishawaka Road, Elkhart, IN 46517

Please send me PZM APPLICATION NOTES.

Name ____________________

Company ____________________

Address ____________________

City ____________ State ______ Zip ______

Phone ____________________

Circle 40 on Reader Service Card

exposed nerve. He has a theory, not a theorem. The difference is that the latter can be proved. However, we owe Dr. Diamond credit for launching an interesting exchange of ideas, as our mailbag continues to prove. And that's *theorem, not theory!*

MUSIC—METRIC OR OTHERWISE

TO THE EDITOR:

I take issue with your amusing bit, "Metric Music" (March, 1981). Too often, we use incorrect nomenclature. In a periodical such as yours, it is intolerable. You are confusing "decimal" (base 10) with "metric." A "meter" is a linear unit—in weight it is a "gram," etc. All this in terms of "media," "wind down," "parameters," etc.—and yet more banalities to come. Please correct. The contents are stimulating, though perhaps not musical.

BEN SOBIN

TO THE EDITOR:

In regards to "Metric Music" by Hal Lion and Jim Fox (March 1981), the tones advocated as metric music seem interesting. I have not heard any of it but I can conceive of some very interesting music using the system.

However, metrication hit the field of frequency determination 20 years ago when the American National Standards Institute (ANSI) published S1.6-1960. This publication lists the "Preferred Frequencies for Acoustical Measurements" as the center frequencies for octaves, ½ octaves and 1/3 octaves. The actual frequencies specified are not even derived from octaves. They are all based on the "decade," a 10-to-1 frequency ratio. The definition used extends easily to the 1/12 octaves of the equally tempered musical scale dividing the decade into 40 geometrically equal parts. The twelfth root of two (1.059463094) is very nearly equal to the fortieth root of ten (1.059253725). Thus, a musical scale based on this ratio will reproduce itself once every "decade" instead of every octave. A frequency scale, starting at 440 Hz, will progress to 4400 Hz thusly:

440.000	782.443	1391.402	2474.302
466.072	828.806	1473.848	2620.913
493.688	877.915	1561.179	2776.212
522.941	929.935	1653.685	2940.713
553.927	985.037	1751.672	3114.961
586.749	1043.404	1855.465	3299.535
621.517	1105.230	1965.408	3495.044
658.344	1170.719	2081.866	3702.139
697.353	1204.088	2205.224	3921.504
738.674	1313.568	2335.892	4153.868
			4400.000

By the end of the first octave this scale has lost 2.09 Hz and Pythagoras will come back to haunt us; however, this scale should warm the cockles of any metrician's heart and certainly has more precedent than does tenths-of-an-octave.

This scale can be adapted to metric seconds or conventional seconds, whichever makes sense to the producer. If it makes sense to any musicians—I have this bridge in Brooklyn that I would like to sell.

FANCHER MURRAY
Sr. Transducer Engineer
James B. Lansing Sound, Inc.

TO THE EDITOR:

So! The big-city engineers are going to discuss music, are they? (March and April **db**.) Well, this musician is writing to tell you how well I think you pull it off. So long as you and your writers continue to present the intelligent, sensitive, and informative articles we've come to expect from your mag, you can be assured of my continued readership. Please, do carry on.

A point of possible interest: if you want to find out who is really responsible for equal-temperament (and it's not Andreas Werckmeister), see the new *Grove's* dictionary of music, under "Temperament." There's an excellent article with lots of references. Every home should have the new *Grove's*, a steal at a paltry $1925.00. (I use the set in the Allentown Library.)

With best wishes for your continued good health, happiness, and challenges well met, I remain

J. D. CRAIG

db replies:

We'll have to pass up the Grove's bargain. We just spent all our money buying a bridge from a guy in California.

TO THE EDITOR:

I was very interested in reading Les Stuck's experiences with M-S mike techniques (**db** letters, March, 1981—Ed.). I too have used M-S extensively.

I think something is missing in the fourth paragraph (of our December, 1980 Application Note—Ed.): "...re*member that the output polarity (of) the rear lobe of any microphone will be* reversed...." Surely this is only true in a mike which is more than half-cosine. I disagree also with the first sentence in the eleventh paragraph: *"Since the M-S technique gives us two cardioid patterns which are derived from one uni- (M) and one bi-directional (S) microphone, it follows that an M-S pickup can also be derived from two cardioid microphones."* I feel this is the most popular misconception regarding M-S miking. As John Eargle points out in *Sound Recording* (pg. 58, FIGURE 3-7C), there is no M-S equivalent for X-Y cardioids.

I have found any of the nine patterns on an (AKG) C-24 mike useful for my sum (M) mike. The figure-8 is useful as described by L. Stuck, and for eliminating unwanted sounds such as PA speakers at the sides of the stage. The

Circle 33 on Reader Service Card

cardioid is most useful when miking a quiet instrument, such as a lute in front of a noisy audience. No X-Y or other M-S configuration yields better rear rejection.

However, I feel the most misunderstood and under-estimated (method) is the M-S pickup using an omni (M) sum mike. The mike is the most natural sounding, since it behaves most like the human ear. It is useless in any X-Y application, so it is used only in spaced-mike techniques which yield questionable mono. An omni/figure-8 M-S pickup uses the two mikes which exhibit the least off-axis colouration, so you get the most natural ambience and greater clarity. This pickup is also useful in situations where desired sounds originate from all directions around the mike, such as organs and choirs placed around a church. Note that the X-Y counterpart (back-to-back cardioids) will not do the job properly because of the off-axis colouration, and because the zero axes of the mikes are facing the sides instead of the important front centre.

It is unfortunate that the M-S system was invented 50 years ago, and thus has acquired the label of "historic." At the time of its discovery, stereo was little more than a novelty, and the primitive mikes could not convincingly capture the sound of a symphony orchestra, and the matrixing transformers took their toll from the quality as well. If the system had been introduced today, it would be a major breakthrough in coincident miking. With our superb stereo mikes and matrix circuits employing the ICs instead of transformers, the system yields imaging superior to any other pickup (except X-Y figure-8s), flawless mono, and a degree of remote control over hall ambience and noise rejection (by pattern selection of the sum mike), and stereo width (by difference-mike level), which is the sole domain of the system.

I hope that when you discuss M-S, you will delve into its unique properties and qualities, not simply as an alternative to X-Y. Three procedures I have found useful, but have yet to see in print are:

a) Rolling off the bass on the different mike. This reduces interference from air conditioning, building vibrations, and standing waves, but does not affect the pickup, because there is no need of directional information below 250 Hz.

b) If using an M-S mike as a spot mike on a group, the involved panning circuits usually described are unnecessary. One need only adjust the width of the group as usual, and position it as desired by rotating the difference mike.

c) No matter how many M-S mikes are used on a recording, or what their respective widths and positions are (as Panned in "b"), they can all be directed to the same sum and difference matrix. They will not interfere with each other, and all can independently be controlled by the channel fader and EQ. I know that there are other useful hints to be discovered, and look forward to your June issue on miking. I hope that some of my thoughts are of use to you. Thanks for a great magazine.

DAVE BURNHAM

db replies:

There isn't any rear lobe unless the microphone is more than half-cosine, so we're both right. All rear lobes have a polarity reversal, and, the pattern must be more than half-cosine in order for a rear lobe to appear. This is covered in greater detail in the continuation of the Mike Math article in this issue.

John Eargle's point about their being no M-S equivalent of X-Y is correct only under the conditions which he describes, in which the M microphone is a pure cardioid. However, if the M mike is an intermediate pattern somewhere between cardioid and omni-directional, then there is indeed an M-S/X-Y equivalent. Such a condition is also described in this month's Mike Math article, and is shown in L. Stuck's second illustration, as well as in the excerpt from John Eargle's forthcoming microphone book, which we printed last month.

As for the M-S pickup using an omni and a figure-8, we'd suggest approaching this setup with some caution. As Mr. Burnham points out, this is equivalent to back-to-back cardioids, which heavily favor the extreme sides of the stereo stage, and the system has no front-to-back discrimination. If this is what's wanted, we wonder whether an M-S combination using two figure-8s wouldn't work just as well, or perhaps better.

OUR READERS FIND GARRON

TO THE EDITOR:

In the March issue of your publication there is a letter from a Mr. David Garfield who is looking for information on his Rapid-Q cartridge machines. At the last we heard, these products were being sold by Engineered Devices Company at 680 Bizzell Drive, Lexington, Kentucky 40504. Perhaps information could be obtained from them.

DONALD E. MEREEN
Director of Marketing
Professional Audio Products

db replies:

Our thanks to reader Donald Mereen, whose letter is typical of many we received. And a special word of thanks to Sr. Julio Conesa who has been kind enough to send a RAPID-Q instruction manual to us, for forwarding to Mr. Garfield. Help is on the way, thanks to **db**'s *readers. If you have a question and/or problem in which we may be able to assist, let us know. You can count on* **db**—*AND its readers—to come to the rescue.*

db Calendar

AUGUST

3-7, 10-14, 17-21 **Digital Sound Synthesis and Sound Processing Workshop.** 18 Oliver St., Boston, MA. For more information contact: Digital Music Systems, Inc., P.O. Box 1632, Boston, MA 02110. Tel: (617) 542-3042.

10-14, 17-28 **MIT Summer Session Programs on Media Technology.** Cambridge, MA. For more information contact: Director of the Summer Sessions, Room E19-356, Massachusetts Institute of Technology, Cambridge, MA 02139. Tel: (617) 253-5960.

SEPTEMBER

4-13 **International Audio and Video Fair Berlin, 1981.** Exhibition Grove, Berlin, West Germany. For more information contact: Professional Travel Management Inc., The New ASAE Building, 1575 Eye St., N.W. Suite 1250, Washington, D.C. 20005. Tel: (202) 223-6415.

13-16 **NRBA 1981 Convention.** Fountainebleau Hilton, Miami Beach, FL. For more information contact: Lisa Friede, National Radio Broadcasters Association, 1705 De Sales St. N.W., Suite 500, Washington, D.C. 20036. Tel: (202) 466-2030.

29 **Sixth Sound Broadcasting Equipment Show.** Albany Hotel, Birmingham, England. For more information contact: Carol Pottinger, Audio & Design (Recording) Ltd., 16 North Street, Reading, RG1 4DA Berks, England. Tel: (0734) 53411.

OCTOBER

25-30 **123rd SMPTE Technical Conference Equipment Exhibit.** Century Plaza Hotel, Los Angeles, CA. For more information contact: SMPTE, 862 Scarsdale Ave., Scarsdale, NY 10583. Tel: (914) 472-6606.

NORMAN H. CROWHURST

db Theory & Practice

The Importance of Basics

• Over the years, I've noticed an interesting change in philosophy about research and development, particularly as applied to systems design. Some years ago, approaching the design of a system required an ability that is now both unnecessary and almost nonexistent. This is best explained by describing it.

Back then, the engineer worked with vacuum tube triodes, pentodes, diodes, and so forth, along with resistors, capacitors, inductors and similar "passive" elements. In putting together circuits, you had to think in two sets of terms, virtually at the same time: you thought of individual blocks, and what was in them, such as amplifiers, equalizers, and so forth; and you thought in terms of a whole block diagram, where each such item was just a block, interconnected with the other blocks.

The major change today is that the contents of the individual blocks no longer bother us. We just put together chips, with some interconnecting circuitry that combines what the individual chips do, into a complete system that does everything we want it to do. And the nice thing, or so it seems, is that we don't have to bother about what goes on in those tiny chips: we just take them for granted.

Does this make the designer's task easier or harder? That's a good question. I have the feeling that it should make the whole thing a lot easier, because it should now be easier to keep the original two parts of your thinking in separate parcels, so to speak. And you would probably give much less thought to how the individual chip does its job, than back when every block had to be built of individual components, like tubes (and later, transistors), resistors, capacitors, and inductors.

In the old days of either tubes or discrete transistors, one did not put in any redundant components; both cost and room limitations usually mitigated against either form of waste. But the advent of chips changed all that. Now, the designers of the chips virtually make sure that any given chip does exactly what it is supposed to do, and only what it is supposed to do. It is now the chip designer's responsibility to consider what will happen if some unpredicted combination of "inputs" is applied to the various prongs of a chip.

As far as is possible, he has to make sure that the little thing is protected against such folly, in as many ways as possible. So, for many purposes, today's electronics has a great many redundant parts in it, if you think in terms of the old components where each chip contains maybe hundreds of electronic "parts," as we used to think of them. But, as the whole chip costs less than individual parts used to cost, that doesn't bother anybody as it once would have.

One result of this is that the chip designer works one place, while the systems designer, who puts lots of chips together to make a system, works somewhere else. But is this really an improvement? Perhaps. On the positive side, it does free the systems designer from getting bogged down in the chore of small parts selection.

But to me, the interesting thing is that basics haven't really changed. Did I really expect they would? I don't think so. But it seems as if a lot of people *think* they have changed. A lot of things have changed about electronics, it is true. When a system "goes out," it will usually be a chip that has "blown." In the old days, if a part in one component of a system went bad, you just replaced the part, not the whole component. Nowadays, of course, you never replace less than a complete chip, and some people favor replacing the complete component in which the chip is installed.

Some observers comment that this fits in with today's "throwaway" society. Nothing is repaired anymore, or so it seems. But at what level is this true? If your brand new $10,000 car quits along the highway, do you have it taken away and get another one? Perhaps that depends on the individual, but obviously this involves a matter of economics that assumes very different proportions in electronics.

There are other factors, such as a consideration of potential "down" time when a failure occurs, and the difference between protection and the fail-safe principle. If the most likely failures are in components that can be carried easily in reserve, then when they fail they can quickly be replaced, and little down time results. But if a single failure is likely to carry with it the whole system, then a whole new system must be obtained, and down time is likely to be much longer.

In modern chip technology, protection refers to additional elements added to input and output circuits, to ensure that unusual signals of any kind are unlikely, or at least less likely, to do any harm. Protection can be applied only to a single component, or chip. Elements added to protect one chip do not usually protect other chips in the same system. But protection in other chips may prevent them from receiving damage in the event

that one chip fails due to its own internal reasons.

Fail-safe may not seem to mean too much in audio. After all, what can failure of an audio system do? It can blow up an amplifier, so that it quits working. But the amplifier is unlikely to catch fire or provide any other hazard to the building. But when you are involved in designing control systems for automated electrical generating stations, or safety systems for underground mines that protect the men working below, fail-safe has a very important meaning.

In such systems, it means that failure of any component must not destroy the whole system. The whole generating system must be safeguarded whatever happens, and the men in the mine must not be subjected to unexpected risks because of some failure in the system. This means that every possible failure must be considered, as well as what will happen in the event a failure does happen. The circuitry must be engineered such that the result is safe. Systems designers for NASA are well aware of this requirement.

But when you work on such systems, you acquire a habit that can carry over into audio design: you think about what will happen in the event certain components fail. For example, in a building audio system, wrong design could result in a situation where the whole system goes into violent oscillation, subjecting the occupants to a very unpleasant noise, until someone can find the "off" switch. Fail-safe procedure would consider that, and eliminate such a possibility.

Suppose microphones on a system are equipped with some kind of automatic gain control. Failure of a component or element could deactivate the automatic control. What will happen if and when this occurs? That is the designer's problem, if he does his job well. There are three possibilities, according to the way the system is designed: the simplest would have gain go to either minimum or maximum, of the range.

If it went to minimum, then the system is effectively switched off when this happens. If it went to maximum, then acoustic feedback is bound to occur, resulting in a penetrating howl that will be painful to the audience. The third possibility is that a failure will lock gain at the setting where it is when failure occurs.

Consider each of these. With the first, the system just quits, which can be annoying, but less annoying than the painful howl. If the system locks its gain, that means that further small changes that ought to occur, won't, with the result that if a little more gain is needed for some reason, it won't come. And if a little less gain is needed, that won't happen either.

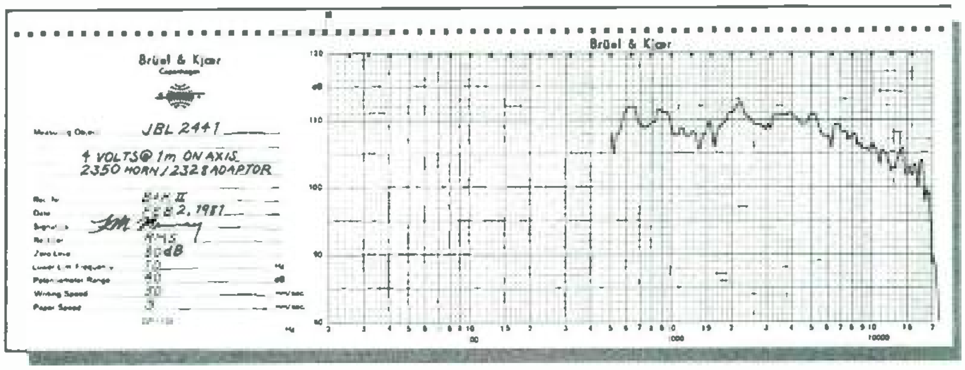
Both of those can be less annoying than either of the first possibilities. The system would still be working, but not as well as it usually does. Usually, something else must be done to compensate for the failure, but you can continue working with it until the fault can be traced, and the faulty component replaced.

Electronic technology developed quite a precise approach in such matters, so good in fact, that other disciplines have sought to copy it. We have come across applications of it in manuals for educators, for spies, for enemy infiltration, and other contexts. But invariably they lack practical application. They just apply something that was developed in an electronic context, blindly.

When you read some of these manuals, it becomes obvious that many of our national and international problems today arise from such misapplications of principles. But when you think further about it, you have to realize that nature has its own, built-in fail-safe. Periodically, something like Mount Saint Helens popping its top occurs and, for the moment, people in that locality think the end of the world has come. But with remarkable rapidity, nature heals the wound thus created on the landscape.

We can learn a lot from closer observation of such natural phenomena. In earlier days, audio people did that more than they do today. Today, the tendency is toward following instructions, or a routine, blindly. ■

This is not the only reason to buy JBL's new 2441 compression driver.



Unequalized frequency response of a typical 2441 on a JBL 2350 horn.

As you can see from this frequency response curve, the new JBL 2441 delivers impressive levels of performance. From extended bandwidth to high sensitivity and smooth, peak-free response.

But as important as these performance parameters are, they're only part of the story. Using the latest laser holography and computer analysis techniques, JBL engineers have developed a unique diaphragm design that allows the 2441 to match its outstanding response with unprecedented reliability and power capacity. That means you get exceptionally high performance without the trade-offs found in previous driver designs.

The secret behind this increased performance lies in the diaphragm's three-dimensional, diamond-pattern surround.[1] As outlined in a paper published in the Journal of the Audio Engineering Society,[2] this surround is both stronger and more flexible than conventional designs. This permits the diaphragm to combine all the traditional reliability and power capacity benefits of its aluminum construction with the extended frequency response of more exotic metals. It also maintains consistent diaphragm control throughout the driver's usable frequency range to eliminate uncontrolled response peaks.

Additionally, each 2441 is built to JBL's exacting standards. The magnetic assembly is machined from rugged cast iron and steel. Extremely tight machining tolerances and hand tolerance matching maintain unit to unit consistency. And finally, each 2441 is individually tested to ensure that it meets published specifications.

So before you buy any compression driver, ask your JBL professional products dealer about the 2441. It'll deliver a lot more than just an impressive frequency response.

1. Patent Applied For

2. Journal of the Audio Engineering Society, 1980 October, Volume 28 Number 10. Reprints available upon request.

James B. Lansing Sound, Inc.
8500 Balboa Boulevard,
Northridge, California 91329 U.S.A.

Specifications		
Horn Throat Diameter	50 mm	2 in
Nominal Impedance	16 Ω	
Power Capacity	70 W continuous program	
Sensitivity (1 watt, 1 meter)	111 dB SPL (on axis of a JBL 2350 90° radial horn)	
Frequency Range	500 Hz to 18 kHz	
Voice Coil Diameter	100 mm	4 in
Voice Coil Material	Edgewound aluminum ribbon	
Flux Density	1.8 T (18,000 gauss)	

JBL's diamond suspension diaphragm combines performance with reliability.

JBL Professional Products Division

Available in Canada through Gould Marketing, Montreal, Quebec

Circle 38 on Reader Service Card

BARRY BLESSER

The Philosophic Trap of Digital Audio

• With any new technology, such as digital audio, there is a strong tendency to focus on the close-up technological details, rather than to take the long view and examine the broad philosophic implications. This is especially true when—from a purely intellectual point of view—those details are complex and exciting. However, when we step away from the details, we see that digital audio is really a statement about the profession's attitude towards quality. Until digital technology, it was easy to attempt to build the best possible system because the analog medium produced its own real limitations. With digital technology, some of these restrictions are removed. It is a little bit like encountering the magic genie who offers three wishes. What should the audio profession wish for? Better quality? Technological improvements? It's all possible with digital.

Before we go further, consider the fact that some "improvements" are actually not improvements at all. In fact, limiting cases show that improvements can become degradations. For example, a frequency response of 300 to 3,000 Hz is poor; a response from 100 to 10,000 Hz is better. A response from 20 to 20,000 *may* be even better; but a response from 1 to 100,000 Hz will no doubt be worse. With a 1 Hz bandwidth limit, an amplifier will attempt to reproduce turnable wow and other eccentricities. These signals may result in high levels of distortion in either the amplifier or the loudspeaker. This illustrates that a change in a physical variable produces perpetual improvement only up to a certain point. Beyond that point there is no gain; and still further beyond that point there may be a degradation and "More" becomes "Less."

At the birth of digital audio, there were no such issues involved. The digital delay line, first introduced in 1971, was a practical solution to the problem of achieving a simple high-quality delay. There was no real alternative in the analog domain. Quality was not an issue, since even a 12-bit digital system was much better than the customary analog tape loop. Following the introduction of the delay line, there was the all-electronic reverberation system. This too solved problems of flexibility, and gave the user control over the variables of the reverberation process. It is only now with the advent of the digital tape recorder, digital mixing and editor, digital audio disk, etc. that the real trap of digital audio is appearing. These systems do not offer new features but they offer *QUALITY*.

QUALITY

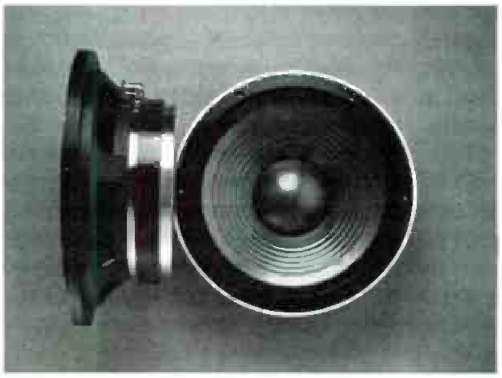
Quality is an elusive term. Do we mean that quality audio signal has *no* degradation? And do we mean to define "no degradation" as the auditory limit of *all* human beings? Hence, if we find a single listener who hears degradation, is the system therefore not of the highest

Experience previously unattainable performance in the mid-frequency range, an evolution of pure audio. Discover the best sounding 12" mid-frequency loudspeaker available. Listen... and experience the TM-1201 from TAD.

Assembled in the hands of veteran audio craftsmen, the TM-1201 is especially designed for mid-frequency applications, with a usable range of 200 to 3000 Hz. Ultra-low distortion, smooth extended frequency response and high power handling establish the TM-1201 as a hallmark in the evolution of sound.

Evolutionary in itself, with a lighter, more rigid material than currently used in the field, the polymer graphite cone has true pistonic motion throughout its specified range, with none of the peaks and valleys associated with cone break-up or doubling. In addition, a high power 4" voice coil with edgewound copper conductors better utilizes the flux within the magnetic gap to increase acoustic conversion efficiency. A corrugated cloth surround, for extreme linearity in high power applications, and a rugged 6-spoke die-cast aluminum frame assures both durable and reliable performance.

Conquer the peaks and valleys of sound with the unprecedented mid-frequency performance found in the TM-1201.



The evolution of sound begins with TAD.

TAD Technical Audio Devices

A division of U.S. Pioneer Electronics Corp.
142 Redneck Ave., Moonachie, N.J. 07074
(201) 440-8234 Telex: 133484 PIONERUSA MOON

Detailed specifications and performance data available upon request.

quality? Let us take the issue of required bandwidth. If one listener can hear the overtones of a musical instrument to 30 kHz, should I specify that all systems must have a bandwidth of 30 kHz? Clearly, nobody would design a mass market audio system on the basis of one listener.

Well, perhaps I can find 15 listeners who hear 26 kHz. Is this a sufficient number of listeners to justify a 26 kHz frequency response? Or perhaps there are 1000 people in the world who can hear the difference between 19 kHz and 23 kHz. Is this an interesting number of people? We are not arguing about the limits of auditory bandwidth in an academic sense; we are noting that there is no clear threshold such that 1 kHz more means perfection. This is the trap of digital audio: the profession will want to settle for less than the technology can provide, since the technology can provide much more than is needed. With digital audio, the technology is not, at last, the limiting factor.

Bandwidth is not the only trap; signal-to-noise, in the form of how many bits, is just as much an issue. Today, one can achieve about 90 dB of S/N but the dynamic range can be made to be over 100 dB with the addition of floating point. Is this enough, too much, or not enough? To turn attention to the new digital audio disc with extensive error correcting, we can ask how many errors should be correctable? Will the listener tolerate one uncorrectable click every 10 minutes, every hour, once per 10 hours, once per year? It is clearly possible to build such systems with some increase in cost.

I personally have had similar experiences in my own fields of expertise at the design phase. Being reasonably knowledgeable about the design of artificial reverberation systems, and having had years of experience in listening to them, I was able to hear a defect in a particular system. This defect was very clear to me and represented a kind of acoustic "red light." However, my colleagues in the audio profession had great difficulty in perceiving this defect. I would guess that there are probably no more than 100 people in the world who could detect it. The reason my ears were that sensitive is that I had been trained to know how to listen for just this effect. The solution to the defect was straightforward, but it required either the addition of money or creating another defect. As a designer, I am left with no guidance because quality is not a simple concept. Over the years, I have heard many similar stories from other designers in other areas. As an industry, we do not talk about this aspect of subjectivity.

NEED FOR QUALITY

We can change the nature of our discussion by asking about the people who feel a need for quality. The group is composed of two distinct classes: the true perceivers and the imaginary perceivers. The first group includes well trained, highly experienced sound engineers, producers, audiophiles, and performers. They often, but not always, hear real defects which the rest of the world cannot hear. The second group is composed of the emotional "true" believers who base their subjective judgement on unconscious biases. Digital is good, digital is bad; tubes are good, transistors are bad; TIM is important, TIM is irrelevant. To an outsider, it is impossible to separate the two groups. We are thus left with the only alternative: a carefully controlled scientific experiment. However, these are difficult and expensive to run and the results do not necessarily bring rationality to the discussions.

These experiments do not end the discussions because the interpretation of the results is itself a subjective judgement. The final conclusion must be in the form of determing the percentage of listeners who can hear the effect. If the industry had a criterion, such that one percent of the population will be allowed to hear a defect, then we might come to a clear conclusion. To state that 0.05 percent of the population might hear the difference between 15 and 20 kHz does not tell us which bandwidth to use. The problem is further compounded by our selection of subjects to use in the percentage reference. Should the population include the average record-buying public? In this case, the teeny-boppers will completely dominate the audiophile.

In reality, one important pressure for quality comes from the performers listening to their own music in the studio monitors. They do not listen to pressings in the local record shops. For this reason, much of the pressure for quality at the studio has no counterpart for the consumer. Another pressure for quality comes from the popular HiFi magazines that serve the audiophile market. For them, making distinctions is a service they provide to their readers. This has,

MOVING?

Keep **db** coming without interruption!

Send in your new address *promptly*.

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

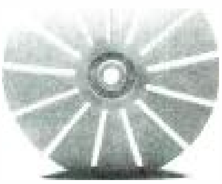
Write to:

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

Announcing the new Altec Lansing 9813 high-accuracy recording monitor.

Loudspeaker accuracy. It's a highly controversial subject. And for good reason. The most prized result of a recording session is an accurate sonic illustration of what is going on in the heads of the producer, musicians, arrangers, and composers. Recording is a process of fusion, and the monitor is responsible for an accurate painting of the completed sonic picture.

Enter our new 9813. We developed it to play its highly critical part in the fusion process with great accuracy. We did it by putting nearly half a century of audio alchemy into it.



The 9813 has an all new high-frequency compression driver that uses our famous Tangerine® radial phase plug.

There's a brand new network design the patent office is already looking at, and the smooth, accurate highs are controlled by an asymmetrical Manta-ray® horn.

The 9813 handles power like no small monitor you've ever experienced. It takes on big amplifiers as though it were addicted to watts. And if you should push it to the limit, there's a built-in system we call automatic power control, which *lowers the power* (never shuts the speaker completely off) and lights a red indicator on the front panel at the same time.

The new 9813 does everything a great monitor should: It sounds super (accuracy need not be unpleasant), handles power extremely well, mids and highs adjust through very wide amplitude ranges, and its great-looking hand-rubbed oak cabinet is small enough for even mobile recording vans (25½ H x 15½ W x 13½ D).

Next time you're visiting your favorite pro audio dealer, ask to hear the new 9813. What you'll hear will be the honest truth.

ALTEC LANSING®

Altec Lansing International
1515 South Manchester, Anaheim
California 92803 (714) 774-2900

Circle 18 on Reader Service Card

however, also created a generation of specification readers. How many audiophiles know the limits of their own perception? When one compares two amplifiers having 0.07 and 0.03 percent harmonic distortion, does one have a reason to believe that the difference is meaningful? What is *your* threshold for harmonic distortion?

If you actually try to answer that question, you will find that it is extremely difficult to come up with a number; and you may be surprised at what that number is. Now, if you really want to have fun with your friends, try training yourself to hear a particular kind of distortion with a particular type of music. After 100 *hours* of practice you can lower your threshold by a factor of 3 to 10. With 1000 hours of practice you can lower your threshold still further. Scientific psychophysics testing in research laboratories has shown that carefully selecting subjects and careful training can decrease thresholds dramatically on almost all perception tests. A sonar operator is just such an example.

We must not assume that the audio profession is providing musical entertainment to the equivalent of musical sonar operators who listen for submarine defects. Digital audio is not the first such example but it may be the most dramatic. The profession may end up dressed in the Emperor's new clothes if we do not ask the question: "How much quality is appropriate or reasonable?" The record buyer has no way to become informed by himself, and there is no public literature available to aid him. The popular news media accept the data as interpreted by the audio professional.

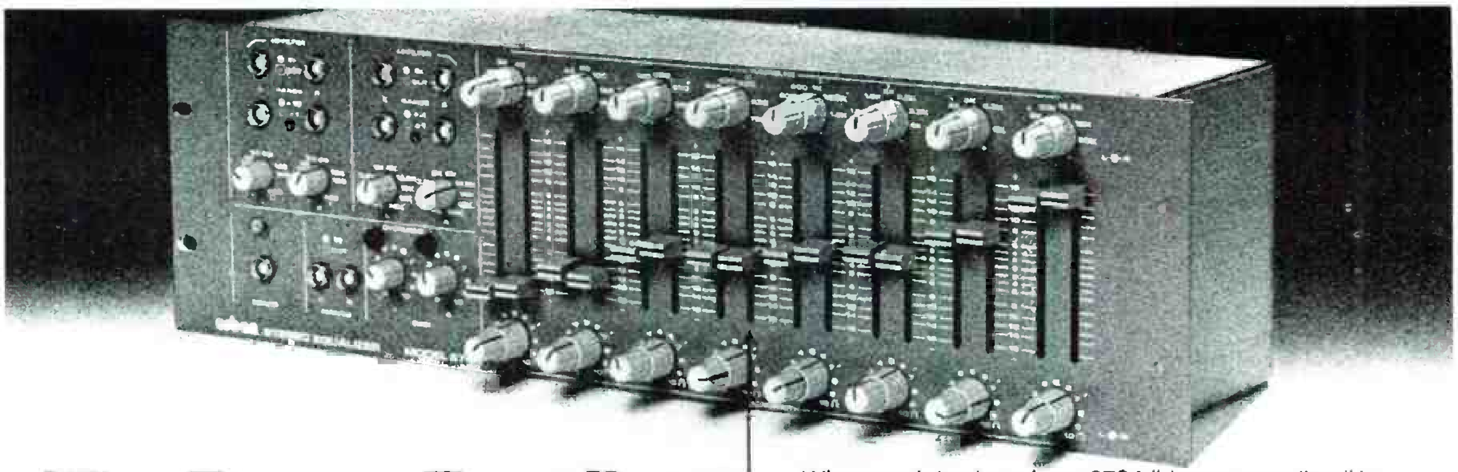
One should not read this article as meaning that digital audio is unneeded or that digital audio is pointless. Rather, digital audio presents a philosophic trap which must be examined carefully. In the limit, the trap reduces to a question of money. A 30 kHz bandwidth is perfectly acceptable if there is no effective cost difference between that and a 20 kHz bandwidth. Equivalently, the cost difference between a 10-bit and a 12-bit A/D may be so small, relative to the complete system, that we choose the larger. However, digital technology tends to have price thresholds. Registers come 8 bits wide and we need two such registers for either a 13-bit or a 16-bit word. However, the jump from 16 to 17 bits raises the cost by 50 percent in terms of our example with registers, since an additional IC must be added. Complex error correction requires more logic and there is less music information. In terms of A/D technology, there is a very sharp price curve as one goes from 14 to 15 to 16 bits of true performance. These differences are not small and they approach factors of 2.

We are now left with a very uncomfortable conclusion. The resolution of the philosophic trap is dynamic in that the "reasonable" solution for today may not be a reasonable solution tomorrow. If some manufacturer could perfect a monolithic A/D converter at 16 bits which cost $9.00, then the decision to use the better quality would be clearcut. If however, the difference between 15 and 16 bits was $500, then we might say that the marginal perceptual improvement is not worth the extra cost.

THE FUTURE

At some point, the listening public will become bored with the new advances in audio reproduction. They will realize that these improvements represent little real improvement in their own listening experience. I have no idea if these thresholds have already been crossed, are now being crossed, or will be crossed later on.

There is another dimension, however, and that is one of features. A feature is something the public can understand without being trained. Color television is a feature compared to black-and-white. Discrete four channel quadraphonic is a feature although not yet a successful one. An automatic record changer is a feature compared to manual record changing. I am of the opinion that features are much more important to the audio profession than quality, once quality has passed the magic threshold, whatever that is. ■



db Editorial

WE'VE JUST RETURNED from Los Angeles, where the Audio Engineering Society held its 69th Convention (May 12-16). For the Society, and possibly for the rest of us, it could be that the 69th will long be remembered as a turning point in the fine art of convention-going.

So far, the AES has been holding three conventions a year (one each in New York, Los Angeles, and Europe). And then, there's the NAB, NRBA, APRS, CES, NAMM and so on. It seems as though anytime someone puts a few initials together, it becomes a justification for one more convention.

Meanwhile, travel and hotel expenses continue to climb. Especially for the small-to-medium size business, the decision to attend this or that convention can be a traumatic—or at least, painful—experience. Many questions have to be answered.

Can we afford to go? Can we afford *not* to go? What's the competition doing? Will we make any sales? Will we lose any sales? And it goes on, and on. And of course, the general state of the economy raises even more questions. Given the prevailing conditions, many businessmen are now watching their "bottom line" even more closely than ever. "To go or not to go" is no longer a decision to be made casually.

Against this backdrop, the Society's board of governors is considering a plan to reduce the number of its conventions. Under consideration is a plan that would see two conventions a year, with the traditional venues of east coast, west coast, and Europe. These would be spaced at six-month intervals. Thus, the next few conventions would be: Fall '81—New York, Spring '82—Europe, Fall '82—Los Angeles, and so on. This gives each of these locations its own convention once every 18 months, rather than once a year.

What, if anything, does this mean to all of us? Frankly, we're not sure yet. The "instant reaction" by exhibitors in Los Angeles seemed to be generally positive. One less show would give them more time to concentrate on business, and be a money-saver as well.

On the other hand, a convention exhibit is also part of business. It's a chance to expose the latest products to a rather large audience. In other words, it's just one more form of advertising. Of course, if the personal contact aspect is not that critical, a manufacturer can reach an even larger audience for less money with a magazine ad.

For us magazine-types, the advertising issue may be a two-edged sword. Many companies beef up their ad campaigns at convention time to draw attention to their latest efforts. And we do our bit by printing extra copies for convention distribution.

But who are we (advertisers *and* editors) really reaching? Are the buying habits of the audio pro truly influenced by a convention and all that goes with it? Of course, it's an obvious way to find out "what's new," but without the convention, would the customers stop buying?

And speaking of buying, the AES's non-profit status prohibits exhibitors from making direct sales on the convention floor. The purpose of the exhibits must be educational, not commercial. In fact, the conventions began their history as technical sessions, with exhibit space being offered almost as an afterthought. But lately, the sheer physical bulk of the exhibits, coupled with the problems of finding enough worthwhile papers to satisfy three conventions per year, has turned the conventions into "hardware shows," at least in the eyes of many of the convention-goers.

Well, how many of us go to the convention to "get educated"? Those who do probably attend the technical sessions and skip lightly over the exhibits. Some may even skip the exhibits entirely, although we suspect that today more people skip the papers, in favor of the exhibit area.

Other conventions are differently organized. For example, The NAB convention is a "selling show" and exhibitors return home knowing just how profitable (or not) their participation was. Thus, it is comparatively easy to determine the effect of this show on the company's "bottom line." By contrast, the expense of an AES exhibit booth is more difficult to assess. Does the "educational" exposure help long-term sales in proportion to its cost? After all, most manufacturers are in the business of selling, not educating. And that being the case, would the manufacturer be better off staying back in the shop, spending the money on product development and possibly on more magazine ads spread over the entire year?

Some AES exhibitors complain about the number of "tire kickers" who show up—especially at the US conventions. It's as if someone had yelled, "Hey kids! The circus is in town!" Every school for miles around cancels its how-to-become-a-big-time-recording-engineer class, and its students descend en masse on the hapless exhibitor who is trying valiantly to sell—oops, *educate*—a prospective client.

Well, there's a bit of a contradiction here, isn't there? If the purpose of the convention is truly educational, then what could be better than turning the students loose on the exhibitors? Besides, today's "tire kickers" will be tomorrow's cash customers, anyway. However, watching the owner of "Mega-buck Productions, Inc." walk away while you're playing "twenty questions" with a crowd of hardware groupies is not a pastime for the faint-of-heart.

On the other hand, the students are impressionable, while the Big Spenders probably are not. Therefore, perhaps the exhibitor should not worry too much about snaring that big one who seems to be getting away, and concentrate on pre-conditioning the small fry who will in time become Big Spenders. Treat them right today, and maybe they'll remember you tomorrow.

Of course, the exhibitor who really can't tolerate the tire kickers can always show up with an exhibit devoid of flashing lights, and shiny knobs and switches. With no fancy gadgets to attract the unqualified masses, perhaps only the serious passers-by will stop. Just have some literature on hand, and an engineer or two to answer technical questions as they arise.

Conflicting thoughts—some no doubt similar to these—are probably running through the minds of the AES leadership, the exhibitors, the serious (and not-so-serious) convention attendees, and at least some of our readers. By the way, for us, any convention is an opportunity for "keeping in touch," for seeing what's new, for hounding potential authors, and for discovering at first hand what our readers are up to.

Of course, we see only a small fraction of our readers there. As the technical publication with the largest circulation in pro audio, we reach some 20 thousand of you every month. Typical AES convention attendance may be only about one-quarter of that figure. And of course, certainly not all are db subscribers (pity!).

So, our opinion (if only we had one) about the number and value to our readers of conventions—AES or otherwise—may not coincide with yours. Therefore, we'd value a little positive feedback. What do our readers think? Should there be more, or less, conventions each year? If you don't attend any of them, is it because of time, money, geography, interest, or what? Let us know what you think, and if there's any sort of consensus (and even if there isn't), we'll pass along the word. ■

Circle 35 on Reader Service Card

LOWELL CROSS

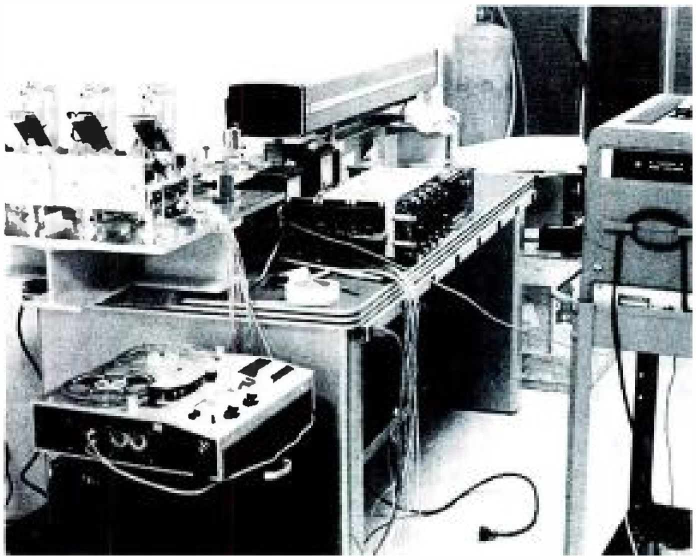
The Audio Control of Laser Displays

The use of the laser beam in the performing arts is constantly growing, showing up in everything from sci-fi movies to performances by symphony orchestras.

THE GENERAL PUBLIC is becoming increasingly aware of the many applications of lasers: in scientific research, medicine, engineering, communications, manufacturing, holography, and certainly, entertainment and the arts. These sophisticated sources of infrared, visible, and ultraviolet light energy have even become a part of science-fiction folklore: no respectable futuristic film of the "Star Wars" variety would be without an arsenal of laser-inspired weaponry and visual effects. While more earthbound, our concerns in this article relate to the technology, much of it based in audio-frequency electronics, that permits the control of laser beams for applications in the performing arts.

THE LASER

The scientific discoveries that led to the development of the laser came in the 1950s and 1960s. At present, a well-publicized suit is being argued in the courts regarding a patent application filed by Gordon Gould in 1959 that apparently disclosed techniques that could have been applied to construct a laser. However, Theodore H. Maiman, a researcher at the Hughes Aircraft Company, conclusively demonstrated the first operational laser in 1960, using a ruby rod as the lasing medium.



1. The Video/laser III during its initial testing stages.

Lowell Cross is Professor of Music at The University of Iowa, where he directs the activities of the recording studios and VIDEO/LASER projects in the School of Music.

SUPERB HANDLING. PERFORMANCE TO SPARE.

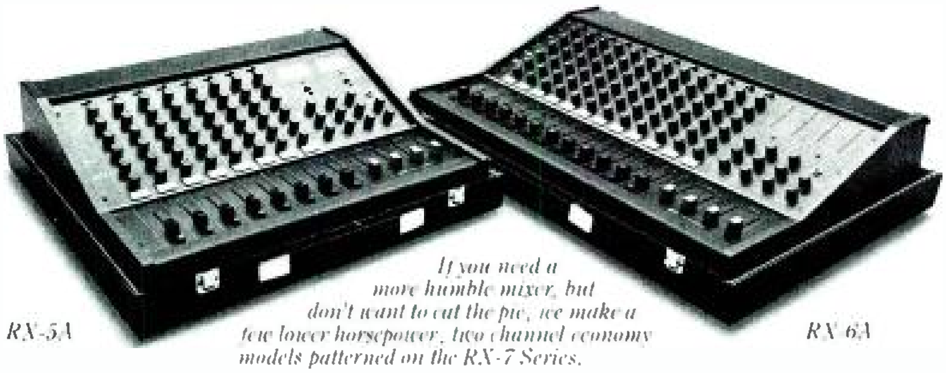
THE NEW TOA RX-7 SERIES MIXING CONSOLES.

THE NEVER-ENDING CONSOLE RACE.

Year-in, year-out, it seems like there's never any clearcut winner. With dozens to choose from, it's been easy for the sound pro to make the wrong choice.

Until now.

Introducing the new TOA RX-7 Series machines: from 16 to 32 inputs (and anywhere in-between) to 4 or 8 program outputs. The RX-7's are completely modular consoles built to take the bumps along any road they travel. Whether you need an RX-7 for reinforcement or recording, or both—we designed them to be small and agile enough to move-out easily and big enough to handle just about any conceivable fixed or remote sound requirement. They give you incredible flexibility,



RX-5A *If you need a more humble mixer, but don't want to cut the pie, we make a few lower horsepower, two channel economy models patterned on the RX-7 Series.* *RX-6A*

Every picture tells a story—like these fully professional, and easily accessible back panel interconnections.

copious features and the kind of performance you might expect from a board that can cost up to twice as much. Or more.

SIT BEHIND SOME REAL PERFORMANCE.

Before we engineered the RX-7's we looked at 'em all: Yamaha, Ramsa, Tascam—all the rest. We stripped down the look-alikes that think they're in the fast lane. We saw that a sound pro has a slim chance of seeing the winner's circle by spinning-out on yesterday's technology. So at TOA, we started from scratch. And here's why we're so confident:

DRIVER'S EDUCATION.

Because an RX-7 is fully modular, you create the board you need. Start with input modules that feature three band E.Q., two echo and foldback sends, long travel faders, peak indicators, stereo panning—and more. A lot more. Going in, you're transformer isolated via XLR from −60 to +10 dBm; insert any signal processing devices you want in the signal path; and, take a direct out for recording.

Add two foldback and echo return Buss Master modules. For your outputs, add either 2 or 4 modules that feature +4 dBm on XLR's. To complete your board, just plug-in a Phones module, a Talkback module and of course, a Power Supply module. A multipin plug connects the remote power supply unit which also gives you switchable phantom power for your condensers.

GREAT VISIBILITY.

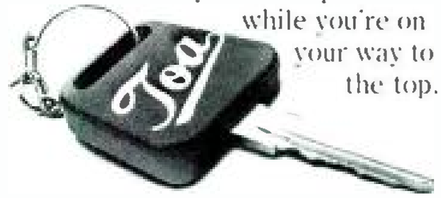
14 large, illuminated V.U.'s (Model 328) are easily seen as well as accessible on a hinged, low-profile meter pod. Easy to see, easy to use, easy to service.

EVEN A PADDED ARMREST.

And a lot more convenience and useful features make the TOA RX-7 Series the professional mixers that they are: for any critical and demanding sound environment.

THE FASTEST WAY TO SHIFT INTO FIRST GEAR.

Give us a call at (415) 588-2538. We'll rush out all the spec's and the name of your nearest qualified TOA dealer. We're confident you'll want to sit behind the superb handling of an RX-7. The one that has performance to spare—and the serious machines that won't drive you to the poorhouse while you're on your way to the top.



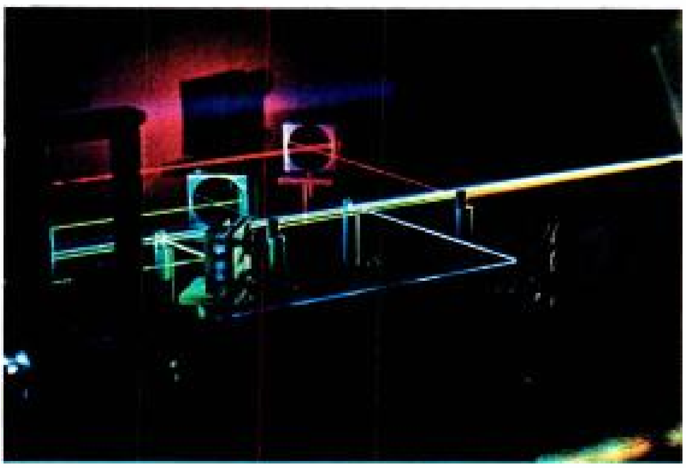
Crafted in Japan, Proven in the States.

TOA Sound Communications

TOA Electronics, Inc.
1023 Grandview Drive
S. San Francisco, CA 94080
(415) 588-2538
Telex: 331-332

Circle 17 on Reader Service Card

2. *The output end of the original four-color video/laser III system.*



3. *Carson Jeffries' custom laser beam positioning system for the video/laser III.*

Charles H. Townes (U.S.), Nikolai G. Basov (U.S.S.R.), and Alexander M. Prokhorov (U.S.S.R.) won the 1964 Nobel Prize in physics for their basic research investigations in quantum electronics that resulted in the engineering breakthroughs of Maiman and others. Lasers were originally called "optical masers," *maser* being the acronym for an earlier device which produced Microwave Amplification by Stimulated Emission of Radiation. The acronym *laser* comes from Light Amplification by Stimulated Emission of Radiation. The microwave spectrum is 3×10^9 to 3×10^{11} Hz; the infrared, 3×10^{11} to 4×10^{14} Hz; the spectrum visible to the human eye, 3.95×10^{14} Hz to 7.9×10^{14} Hz with the ultraviolet, X-ray, gamma ray, and cosmic ray spectra above the visible in the total electromagnetic spectrum. The visible spectrum can be expressed as wavelengths occupying the range of 700 nanometers (or 7×10^{-7} meter: red) to 400 nm (4×10^{-7} m: violet)—less than one "octave."

The fundamental property of lasers is the *coherent* nature of the light emitted. This *coherence* of the light simply means that all corresponding points on the wavefront are in phase. The atoms in the lasing medium emit light waves (or photons, if one prefers) that are not only in phase, but are all eventually propagated together in the same direction. Gas lasers—helium-neon, argon, krypton, and mixed-gas—are types most often used in laser light shows. They incorporate a plasma tube (a gas discharge tube not unlike a fluorescent lamp or a neon sign) which is surrounded by a strong electromagnetic field and placed within an optical cavity. The coherent, in-phase propagation of the photons results from several simultaneous actions occurring within the laser head: electrons are accelerated through the plasma tube and "spun" by the magnetic field, causing ionization and "population inversions" of the gas atoms, the stimulation of which results in the omission of the photons as light energy. The optical cavity in a typical argon or krypton ion laser is about one meter in length, within which standing waves on the order of 10^6 in magnitude occur, at the visible wavelengths of light. Resonances, standing waves, and oscillations are produced in the cavity in the 3.95×10^{14} to 7.9×10^{14} Hz range. This optical cavity is defined as the distance between two mirrors at either end of the laser head, often outside of the plasma tube.

The rear mirror is almost totally reflective, while the front transmitting mirror is partially reflective and allows the output beam to exit from the optical cavity. This beam of light exhibits very special properties owing to its coherence: it is usually quite intense (very concentrated in cross-sectional area), collimated (the light beam consists of almost-parallel rays, diverging very slightly over great distances), and monochromatic (totally saturated in hue). This last property can be ascribed to the single wavelength nature of the photon emission, which defines a discrete frequency in the visible spectrum. By comparison, all other forms of light are emitted with phase relationships of the photons distributed in a random, non-coherent manner. The analogy of white light, such as sunlight, to random "white" noise in the audio spectrum is both appropriate and meaningful.

In addition to solids such as the ruby, and gases such as krypton, lasing media can include liquids. Exotic liquids containing dyes have been used to construct lasers whose wavelengths are tunable over a wide range in the visible spectrum. However, gas lasers, especially krypton or mixed-gas krypton-argon, offer the most attractive possibilities for laser art, because they produce several specific wavelengths of pure spectrum colors simultaneously. The plasma tubes of mixed-gas lasers contain a mixture of these two "noble" gases (each usually having a valence of 0), which become ionized to a higher positive valence when the plasma tube is operating. The laser produces the spectral lines of krypton and argon, all of which are bound up in a single output beam approaching

4. *Carson Jeffries and the video/laser III system in Guanajuato, Mexico for the IV Festival Internacional Cervantino.*

FOR THE BRIGHTEST STARS.

Molly Hatchet — Walter Murphy Band — Rush — Bob Seger & The Silver Bullet Band — George Thorogood & The Destroyers — The Whispers
Ronnie Prophet — Raydio — Sun — Supertramp — Johnny Guitar Watson — Betty Wright — ZZ Top — ABBA
Barbra Streisand — Styx — Donna Summer — Grover Washington, Jr. — Atlanta Rhythm Section
Walker — Joe Walsh — Dionne Warwick — BT Express — Bad Company
Meri Wilson — Bell & James — William Bell
ABBA — Aerosmith — The Blues Brothers — Blondie
Average White Band — Bootsy's Rubber Band
Bar-Kays — Bee Gees — Alicia Bridges — Bad Company
Blue Oyster Cult — Jimmy Buffett — The Buggles
Debby Boone — Richard Clayderman
Brass Construction — Norman Connors — Bob Dylan
Peabo Bryson — Crosby, Stills & Nash
Cheap Trick — Chicago — George Duke Band
Con Funk Shun — Eagles — Earth, Wind & Fire
Bill Cosby — Rick Dees — Emotions — Jon English
Neil Diamond — Firefall — Funkadelics
Dramatics — Bob Dylan — Fleetwood Mac — Dan Fogelberg
Electric Moog Orchestra — J. Geils Band
Fatback — Roberta Flack — Graham Central Station
Dan Fogelberg — Ace Frehley — Emmylou Harris
GQ — Crystal Gayle — Dr. Hook — Hot
Andy Gibb — Chantal Goya — Hiroshi Itsuki — Millie Jackson
Merle Haggard — Hall & Oates — Jefferson Starship
Heart — Rupert Holmes — Robert John
Instant Funk — Isley Brothers — Sachiko Kobayashi
Millie Jackson — The Jam — Kiss
Waylon Jennings — Billy Joel — Evelyn "Champagne" King — Kinks — Mary Macgregor — Maze
Journey — Chaka Kahn — Kansas — Eigo Kawashima — Johnny Logan — Lynyrd Skynyrd — Leslie R. McKeown — Meat Loaf
Kool & The Gang — Led Zepplin — Ramsey Lewis — Barry Manilow — C. W. McCall — Ted Nugent — Parliament — Chuck Mangione
Peter Maffay — Chuck Mangione — The Manhattans — Willie Nelson — Ronnie Prophet — Raydio — Rush — Molly Hatchet
Meco — Steve Miller Band — Molly Hatchet — Walter Murphy Band — The Pretenders — Spyro Gyra — Steely Dan
Dolly Parton — Tom Petty & The Heartbreakers — Pink Lady — Bruce Springsteen — George Thorogood & The Destroyers
Bob Seger & The Silver Bullet Band — Shalamar — Carly Simon — Slave — Supertramp — Jerry Jeff Walker — Joe Walsh
Barbra Streisand — Styx — Donna Summer — Sun — Gino Vannelli — Grover Washington, Jr. — The Whispers — Pink
Pete Townshend — Trooper — Dionne Warwick — Johnny Guitar Watson — Barry White
Robin Williams

The talents of recording stars and their studios provide the magic that turns a reel of recording tape into an outstanding creative achievement.

The Ampex Golden Reel Award honors those achievements which were mastered on Ampex professional recording tape. They have earned a place in the ranks of the world's most successful recorded albums and singles.*

Included in the award is a gift of $1,000 to the recording artist's favorite charity.

Over the past three years, there have been more than 200 Golden Reel recipients. And more than $200,000 donated on their behalf.

Congratulations to all of them on a masterful performance.

AMPEX

Ampex Corporation, Magnetic Tape Division, 401 Broadway, Redwood City, California 94063 415/367-3889

Ampex 1981 Golden Reel Winners as of 1/81

Circle 25 on Reader Service Card

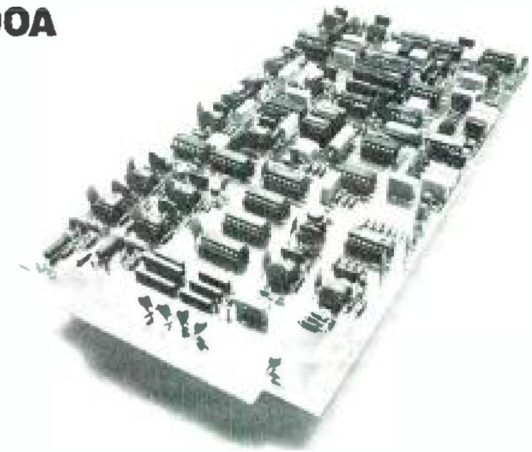
*RIAA Certified Gold.


1500A TAPE RECORDER TEST SYSTEM
SOUND TECHNOLOGY
FD L 1.5% -00.9dB R 1.4% -00.9dB
10dB/□ L-19.0dB R-19.5dB 1.00kHz
POWER
ON
CURSOR POSITION
LOW FREQ LIMIT SET
COPY
OUTPUTS
LEVEL
ATTEN
LEFT
RIGHT
MAX 0.5V
INPUTS

The Sound Technology 1500A

It's the first microprocessor controlled audio measurement test system. It can do in minutes what used to take hours with more conventional and ordinary test set-ups. And, it can show you things you've never seen before.

Designed around the most advanced microprocessor hardware, the 1500A will show you the whole story on an integral CRT with adjustable cursor. Push a "Copy" button, and it delivers a hard-copy printout from the optional VP-150 Video Printer.



What Will It Do?

Conceived to be the ultimate precision test instrument for tape recorder analysis, the 1500A evolved into a comprehensive audio test system for many applications. Here's just a small sample of the varied jobs it will do:

- Complete tape recorder mechanical and electronic performance checks
- Thorough phono cartridge analysis
- One-third octave spectral analysis
- Evaluation of audio quality for VTR's
- Acoustical room analysis including microphone and loudspeaker measurements
- Quality control for high speed tape duplication systems
- Semi-automated production testing
- Research and development for the audio tape manufacturer
- Quality assurance for the audio distribution network
- Exclusive asynchronous inputs and outputs for remote location testing (satellite, transmitter, studios, etc.)

Here's the kind of data you can get:

- Frequency Response
- Azimuth at 4 discrete frequencies
- 2nd and 3rd Harmonic Distortion Vs. Level
- Wow & Flutter; noise; weighted or flat
- Channel Separation 20Hz – 20k
- Delta Speed & Drift

Because of the modular plug-in design, the 1500A is designed to grow with you. The first plug-in option will be available soon: a spectral noise and flutter card. Other current accessories include a hard-copy printer, flight case, rack mounting ears, and our own test record for phono cartridge analysis.

Who Can Use It?

Broadcaster. Recording studio. Film sound studio. Audio manufacturer. Audio dealer. Service technician. Researcher. Virtually anyone whose job requires accurate evaluation of audio equipment performance. Wherever you are in the audio spectrum, it can make life a whole lot easier.

Clean up your act with the 1500A. It's intelligent. And so is a phone call to Sound Technology. We'll be pleased to send full information on the 1500A and our other industry standard test equipment.

Please send me more information on how the Sound Technology 1500A System can help me clean up my act.

NAME ____________________

COMPANY ____________________

ADDRESS ____________________

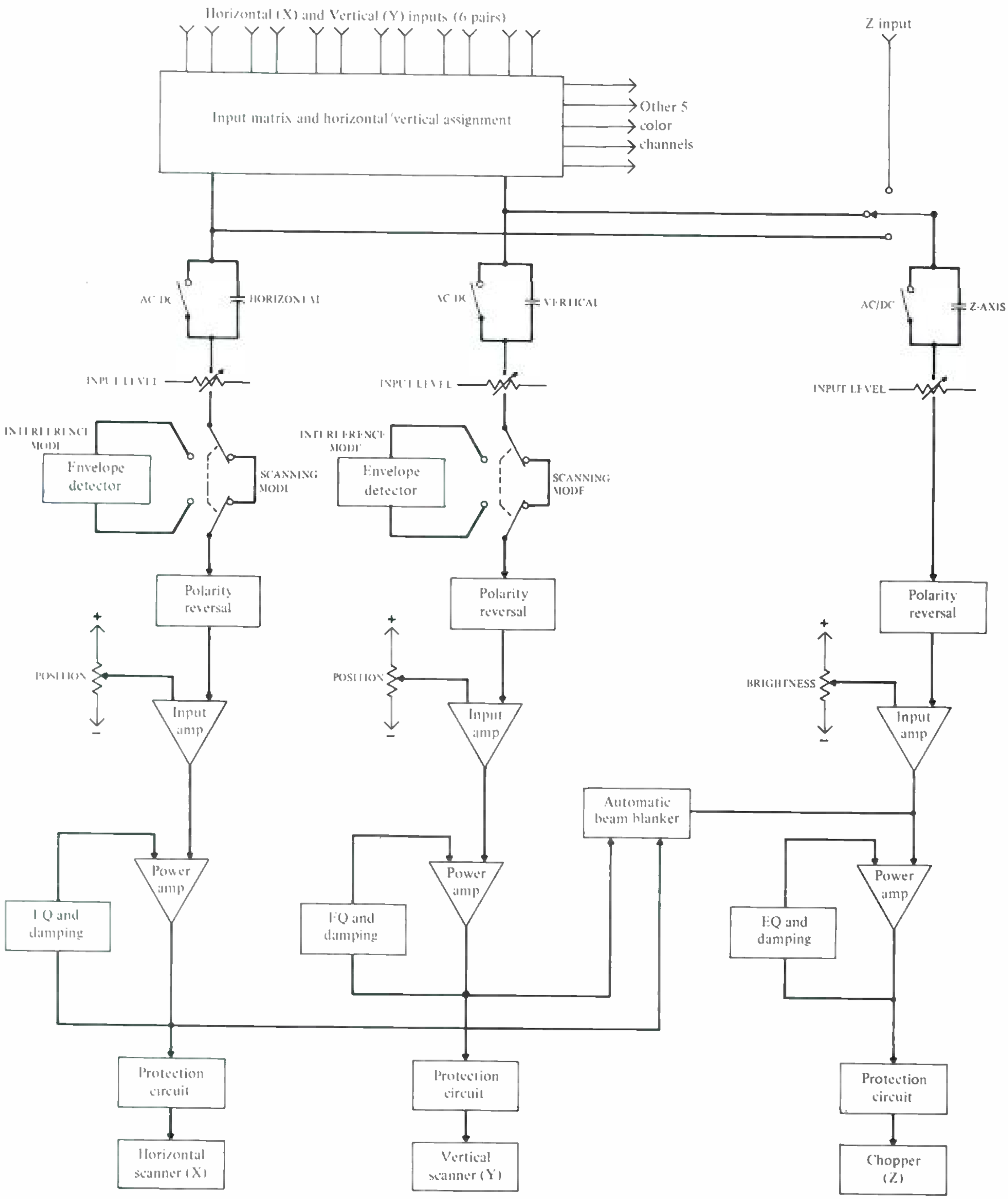
CITY ____________ STATE ________ ZIP ________

PHONE ____________________

SOUND TECHNOLOGY Department 3003102, 1400 Dell Avenue, Campbell, CA 95008 (408) 378-6540 TELEX: 357445

TEST SET THAT YOUR ACT.

Circle 36 on Reader Service Card



5. Block diagram of one color channel of the VIDEO/LASER system.

white. The total beam may be refracted into its colored component beams by a prism. An example of a mixed-gas laser is the Coherent Inc. model CR-MG, used in the VIDEO/LASER deflection systems at The University of Iowa and The Adler Planetarium, Chicago.

Coherent Inc. and a major competitor, Spectra-Physics, sell ion lasers like the CR-MG beginning at $15,500 (plus shipping). These "scientific" or "industrial" lasers require careful setup and maintenance procedures; a certain level of competence is expected of the user. Such lasers do not operate on ordinary house current (208 V, 3 phase, 10 kW AC service is required); an adequate supply of water for cooling must flow through both the laser head and its separate regulated power supply; and precise operating conditions of voltage, current, and pressure within the plasma tube must be maintained.

As many as thirty discrete wavelengths from the infrared through the visible to the ultraviolet may be generated from ionized krypton and argon gases. Following are the ones most useful for kinetic laser art systems.

Wavelength (nanometers)	Color	Lasing Medium	Approx. Power (milliwatts) Coherent Inc. CR-MG laser
647.1*	red	krypton	250
568.2*	yellow	krypton	100
530.9	yellow-green	krypton	80
520.8	green	krypton	100
514.5*	green	argon	250
488.0*	blue-green	argon	250
476.5*	blue	argon	100
468.0*	deep blue (violet)	krypton	less than 50

*Used in the VIDEO/LASER systems. The 647.1, 514.5 and 476.5 nm wavelengths are found very close to the "corners" of the chromaticity diagram of CIE (*Comite International d'Eclairage*). They were used as the red, green, and blue primary colors in a laser-generated projection color television system exhibited in the Hitachi Pavilion at Expo '70, Osaka, Japan.

Since all of these wavelengths are available together, the total (unrefracted) output beam of a mixed-gas laser can measure up to 2 watts. An ordinary light bulb with an output rating of 100 watts can deliver only about 15 to 20 watts of light power output. Even this would appear to be much more than the 2 watts from the laser, yet at 30 meters from either source, the laser beam is on the order of 1,000,000 times more intense. Because of this concentration of power in collimated laser beams, stringent safety precautions must be observed when lasers are operated in public.[1] However, with the proper safeguards, viewing a laser event is as safe for the eyes of the audience members as watching a motion picture.

EARLY DEVELOPMENTS: VIDEO/LASER I AND VIDEO/LASER II

My own work in the laser-art medium developed from my concern and experimentation with a problem peculiar to electronic music on tape: the total abstraction of the sound materials. Public performances of tape music offer no visual information whatsoever about the processes by which the sounds are made. At first I was attracted by the abstract nature of electronic music, since by necessity the process of audition requires concentration on the sound itself, to the exclusion of all other factors; the rich visual rituals of performed music are absent. Even the reproduction of traditionally performed music, via recording or broadcast, operates within a visual frame of reference, the "persistence of memory" of our knowing the manner in which the music was originally performed. The most satisfactory performances (i.e., reproductions) of recorded electronic music may take place not in the concert hall, but at home. Here, one does not expect to participate in a performance ritual. Yet wherever the locale of reproduction, listeners to tape music are still denied visual information, either about the musical process, or from interrelated materials that could contribute to the appreciation of the sound materials.

Perhaps unconsciously trying to deal with these abstract properties of recorded tape music, composers in the '50s and '60s experimented with at least two approaches: the direction and mobilization of sound in space (including the construction of specialized equipment at electronic music studios in Paris, Cologne, Columbia-Princeton and Toronto, as well as my own development, the "Stirrer"), and the idea of having the composer/performers come out of the studio onto the stage, heralding the advent of "live electronic" music. Having no interest in establishing a performance ritual for electronic music, I decided to try the second possibility of devising a means for making *electronic* visual correlates to the sound materials. Assisted by Anthony J. Gnazzo at the University of Toronto Electronic Music Studio, I composed some tape pieces, including *Video II (B)*, whose signals in performance were fed simultaneously to a two-channel audio system and to the inputs of an X-Y oscilloscope. The oscilloscope was unsatisfactory for an audience presentation, and I subsequently modified a television receiver into an X-Y display device. About this time (1965), John Cage and David Tudor, leading figures in "live electronic" music, took an interest in these developments, and by 1966 the three of us had given several public collaborative performances with electronic sound and X-Y television devices. The equipment which I modified for these purposes included black and white sets, color receivers, and a black & white television projector. The projector offered the most interesting possibilities for audience presentations, but it did not withstand an evening of use following the modification: all of our X-Y patterns became permanently etched on the phosphor screen of its cathode-ray tube, owing to the high intensity of the brightness levels which the modification permitted.

At this time, the first argon and krypton ion lasers were in the process of being developed and marketed, and soon I had the opportunity to pursue my ideas for using a laser to overcome the inherent deficiencies of modified television as a medium for public performances. In 1968, shortly after becoming Artistic Director of the Tape Music Center, Mills College (Oakland, California), I had the very fortuitous opportunity of becoming acquainted with Carson Jeffries, a gifted and innovative sculptor of kinetic art systems, and significantly, a renowned Professor of Physics at the University of California, Berkeley. I somewhat timidly proposed to Professor Jeffries my ideas for using a laser and a tandem galvanometer mirror system, driven by audio electronics, to make a laser X-Y projection device for public performances. I suggested that the collimated nature of laser beams offered possibilities for X-Y projection via mirror galvanometers, since the persistence of the eye would permit the perception of kinetic imagery when scanning frequencies in the audio range were employed. Prof. Jeffries agreed that this was indeed a feasible idea, and he embraced the plan with his characteristic enthusiasm and artistic dedication. On May 9, 1969, David Tudor, Carson Jeffries, and I gave at Mills College the first public performance with a multi-color, X-Y laser projection system, programmed by electronic music, operating in conjunction with modified X-Y television equipment, and entitled *Audio/Video/Laser*. Jeffries and I called the X-Y device that we put together, with equipment borrowed from Honeywell, Coherent, and other companies, VIDEO/LASER.[2]

This first VIDEO/LASER was the prototype for a larger commissioned project, VIDEO/LASER II, which we built in Jeffries' Berkeley sculpture studio. It was installed under my supervision, with faithful attendance by David Tudor, in the Pepsi-Cola Pavilion at Expo '70 in Osaka, Japan. The overall project for the Pavilion was undertaken under the auspices of Experiments in Art and Technology, Inc. (EAT), a cooperative group of artists and engineers based in New York. Tudor was one of the four initiating artists for the Pavilion, which was attended by over 2,000,000 Expo visitors while the 1970 World Exhibition was open. This four-color system (red, yellow, green, blue) utilized a Coherent mixed-gas laser, Bell & Howell mirror galvanometers, Jeffries' own custom-machined mounting and alignment system, and my electronic control unit. The Coherent laser was later repatriated by Jeffries, donated by him to the UC-Berkeley Physics Department, and converted to argon operation, where he uses it today in his research.

Within about three years, the first imitators of VIDEO/LASER II appeared. Among the better known of these was a group calling themselves "Laser Images," based in the Los Angeles area. They have made projectors for use under planetarium domes, at football halftimes, and elsewhere, using recorded music from science fiction film soundtracks, pop music, rock, etc., as part of the "light show." Their "Laserium" equipment has spawned similar approaches by other groups. The "Laserium" was introduced after Elsa Garmire, laser scientist, disclosed the techniques of the Expo '70 system, VIDEO/LASER II, to Ivan Dryer, a filmmaker. He later became the president of Laser Images, Inc. Dr. Garmire requested technical information from Jeffries and me regarding the design of VIDEO/LASER II for a publication which she wrote on the various systems within the Pepsi-Cola Pavilion; she was also present during the installation of this system at Expo '70[3].

VIDEO/LASER III

Not long after my wife and I returned from our 1970 adventures in Japan and India, laser-related and otherwise, we relocated at The University of Iowa, where I was hired to direct the audio and recording operations in a recently completed performing arts facility for the School of Music—and where my work with lasers was looked upon with favor by William Hibbard, a composer and the director of the then flourishing Center for New Performing Arts (CNPA). After the funding for a new laser system was secured under the auspices of the CNPA, Jeffries and I began to work in 1971, 2,000 miles apart, on VIDEO/LASER III.

During the interim since the 1968-69 design period for VIDEO/LASER I and II, General Scanning Inc., Watertown, Massachusetts, had patented and introduced a series of optical scanners with vastly improved characteristics over previous designs. (The mirror galvanometers in VIDEO/LASER I and II, made by Honeywell and Bell & Howell, were intended for strip-chart recording instruments which used light sources and moving photographic paper; Jeffries and I had to adapt them for X-Y laser scanning.) The General Scanning units offered a controllable bandwidth up to 10 times greater than that of the other galvanometers; they also could be mounted in the proper geometrical relationship for unimpeded X-Y scanning applications. Like its predecessor, VIDEO/LASER III was designed to have independent X-Y scanning in each of four principal colors (red, yellow, green, and blue), so we ordered eight scanners from General Scanning, carefully specified for our new system. The design parameters for these scanners were optimized for bandwidth, frequency response, and scanning angle. We also ordered four beam "choppers" to provide Z-axis modulation and to control the relative brightness levels in the four colors. The mechanical construction of the system was undertaken by Jeffries in his Berkeley sculpture studio, with assistance in Iowa City from the University's Physics Machine Shop. I supervised the electronic design and assembly in the new facilities of the School of Music, which included a suitable area for a laser studio, now referred to as "Laser Hall." VIDEO/LASER III was given its premiere on November 29, 1972 in Hancher Auditorium on the Iowa campus, in my piece *Electro-Acustica* for orchestral instruments, electronic sound ("live" and on tape), soloists, and laser system. William Hibbard conducted, and Carson Jeffries assisted me in the laser performance. Between 1972 and 1980, the system was seen in numerous performances within the U.S., and on tours to Mexico (1976), Germany (1977 and 1979), and most recently, Italy and Austria (1980), where David Tudor and I gave performances at the Venice *Biennale*, at the Roman Forum, and at Ars Electronica, associated with the International Bruckner Festival, Linz. Touring with VIDEO/LASER III requires advance planning for shipping, customs clearance, local laser safety regulations, and performance logistics. The standards for 3-phase AC power vary in voltage and configuration in different localities (wye vs. delta; neutral and grounding conventions) as do those for hose fittings for the water supply. Our air freight shipment to Italy and Austria weighed 630 kg (1,385 lb).

Before the 1980 European tour, VIDEO/LASER III was converted to six-beam operation with funding from a grant by The University of Iowa Foundation. The mechanical reconstruction was performed by Jerry Swails, machinist, University of Iowa Medical Instruments, and the electronic drive amplifiers for the scanners and choppers were redesigned and optimized by Stephen Julstrom, engineer for the Recording Studios and VIDEO/LASER Projects in the School of Music. There are now 12 optical scanners in the system, one each for horizontal (X) and vertical (Y) for the six beams: red, yellow, green, blue, either blue-green or deep blue (approaching violet), and "white"—derived from the total output beam of the mixed-gas laser (see wavelength chart above). There are now also six choppers, including an example of a new design proposed by Stephen Julstrom, with improved mechanical characteristics, and custom-fabricated for us by General Scanning Inc.

SCRIABIN: PROMETHEUS

One of the graduate seminars I attended at the University of Toronto was entitled "Scriabin, Busoni, and Reger—The End of an Era?". My research project for this course was an analysis and report on *Prometheus: the Poem of Fire* (1909-10) by Alexander Scriabin (1872-1915), a remarkable turn-of-the-century Russian composer with strange, monumental, yet prophetic notions relating to the synthesis of the arts. (The first electronic music studio built in the Soviet Union is named for Scriabin and is associated with the Scriabin Museum, Moscow.) My study of the score of *Prometheus*, considered by his biographer Faubion Bowers to be his fifth (and last) symphony, was concerned mostly with its musical elements, but I was naturally intrigued by its compelling theosophical cover design (by Jean Delville), the symbolism of its theosphical/philosophical extramusical program, and especially, the top line of the score, marked "Luce"—for a lighting instrument.

Scriabin wanted much, much more than just music and lighting for his final works (perhaps mercifully uncompleted), the *Mysterium* and its *Preliminary Action*: dance, mime, gestures, processions, fires, incense, perfumes, tastes, caresses, pain, tactile experiences, theatrical effects, etc. After seven days and nights of these activities (according to Scriabin), there would be a final "suffocation of ecstasy" among the faithful who had assembled in the Himalayan Mountains for the occasion—during which he expected a new, exalted race of men to be born, Scriabin himself leading them. *Prometheus* is a rather less cataclysmic affair, yet it may be identified as an early forerunner of current multimedia explorations in the arts.

While the piece exists in the standard orchestral repertoire, no truly satisfactory performances of the "Luce" part were possible until devices such as VIDEO/LASER III became available. In 1975 I suggested to James Dixon, conductor of The University of Iowa Symphony Orchestra, and James Avery, professor of piano, that we perform *Prometheus* according to the composer's intentions. The noted Russian-born lexicographer, composer, and authority on Scriabin, Nicolas Slonimsky, has stated that, "Perhaps the nearest approximation to Scriabin's scheme was the performance of *Promethee* by the Iowa University Symph. Orch. on Sept. 24, 1975, under the direction of James Dixon, with a laser apparatus constructed by Lowell Cross."[4] For this performance in Hancher Auditorium, we used a special electronic keyboard, played from the "Luce" part in the score, and programmed the laser system with Scriabin's own orchestral sounds via a stereophonic (X-Y, of course!) microphone system. The keyboard performer operated the choppers in VIDEO/LASER III as well as the dimmers in supplementary lighting equipment, providing color changes in synchronization with the music, in accordance with Scriabin's pitch-to-color associations (C = red, F# = blue, etc.). In addition to deploying the symphony orchestra, the keyboard-controlled laser and lighting equipment, a huge projection screen, and all of Scriabin's other musical requirements (piano, chorus, organ), we burned Russian incense ("Cathedral Blend No. 3," provided by Faubion Bowers, the biographer) and propagated a dry-ice cloud into the auditorium (infused with No. 4711 Eau de Cologne), onto which laser effects were projected toward the end of the 23-minute work. We played two performances to accommodate an attendance of over 4,200: the auditorium has a seating capacity of 2,500. Our production was made into a 16mm color film by Franklin Miller, a professor in the University's Division of Broadcasting and Film. It has been shown in the U.S. on CBS-TV, in Germany on WDR, and in the Netherlands, on NOS.

VIDEO/LASER IV: SIGNAL ROUTING IN THE SYSTEMS

A significant collaboration between The University of Iowa and The Adler Planetarium, Chicago, led to the construction of VIDEO/LASER IV. This six-beam system, based on the design of VIDEO/LASER III, was built on the Iowa campus during 1979-80 and installed under the dome of the Planetarium's Sky Theatre in March 1980. It is used there daily to provide added visual interest to the sky shows and other educational presentations designed by the staff of the Planetarium. VIDEO/LASER IV was commissioned in commemoration of the 50th anniversary year of The Adler Planetarium.

FIGURE 5 is a block diagram of one color channel of VIDEO/LASER III and IV, allowing for slight differences in the two systems. The audio frequency input signals may come from two-channel music of virtually any type, from electronic generators, or from natural sounds in the environment via microphones. These signals enter the system via the input matrix switches, which also determine the horizontal/vertical assignment of the input pair. It should be noted here that stereophonic audio reproduction bears striking similarities to the X-Y design characteristics of these laser systems. Anyone familiar with Lissajous patterns displayed on an oscilloscope screen from two-channel audio signals will appreciate the potential applications of projected X-Y laser imagery. After the signals have been selected at the input matrix, they are presented to the horizontal and vertical electronic drive circuitry and made available simultaneously for special forms of Z-axis modulation. Undesirable DC offsets in the input signals may be removed by switching in the blocking capacitors; otherwise the system is DC-coupled. The response of the overall scanning system is quite uniform from DC up to the resonant frequency of the scanners (2.2-2.4 kHz), with useful response extending to above 3 kHz. This range is sufficiently wide to produce satisfactory X-Y displays from most forms of music. The uniformity of response has been achieved in the design of the damping and equalization circuitry of the drive electronics. Following the input level controls, a set of switches determines either the scanning or the interference mode. The scanning (X-Y) mode is the basic operating condition of the system, for which all design criteria have been optimized. However, another interesting vocabulary of visual effects can be made to operate from the input sound materials by using the interference mode, which takes advantage of the coherent, single-wavelength nature of laser light. The control systems for VIDEO/LASER III and IV include remote switching for sets of solenoids which move various textured translucent materials in front of the scanners; the beams may be either stationary or deflected when passing through these materials. The amplitude detectors cause the resulting interference patterns to evolve at sub-audio rates, in response to the dynamics of the music or other audio information. The polarity switches provide the option of 180-degree phase reversal, inverting the horizontal or vertical coordinates of the scanned imagery. The polarity reversal functions, in combination with the horizontal/vertical assignment possibilities, permit the six scanned displays to be oriented in numerous configurations, including several symmetrical arrangements. The position controls function exactly like the horizontal and vertical centering controls on an oscilloscope. The DC-coupled input and power amplifiers incorporate frequency-shaping and damping characteristics to provide control over the mechanical properties of the scanning transducers. To prevent overdriving and possible damage to the scanners, protection circuits limit the output of the power amplifiers (nominal maximum output, approximately 5 watts).

The Z-axis, or intensity modulation circuitry is very similar to the scanner channels. Chopper signals may be derived from either the X or Y scanner signals or may be obtained from other sources. The AC/DC switches and input level controls perform the same functions as in the scanner electronics. The Z-axis polarity reversal switches determine whether the chopper aperture will open, or will close, from an input signal of given polarity. The "brightness" controls provide a continuously variable opening or closing of the chopper apertures and determine the threshold levels of Z-axis modulation. The resonant frequency of the choppers is approximately 700 Hz (useful response to 1 kHz), again controlled by frequency shaping and damping circuitry. While more rugged than the scanners, they are also protected from being overdriven. The optional use of an automatic beam blanker prevents a stationary (undeflected) beam from exiting the system. In realizing these designs and applying them in public performances, I have been invaluably assisted not only by Carson Jeffries and David Tudor, but also by Stephen Julstrom, Thomas Mintner, Peter Lewis, Joel Carl, Peter Elsea, Tim Barrett, and Thomas Henry.

THE PRESENT AND THE FUTURE

Like other audio-frequency applications, our VIDEO/LASER projects are becoming increasingly more involved with digital technology, microprocessors, and hybrid devices. Examples of these techniques are found in the "3-D Laser Display Processor" built for VIDEO/LASER III in the spring of 1981 (see photo). This equipment processes three or more input signals, and via rotation techniques and other clues, imparts to two-dimensional X-Y displays the illusion of an existence in three dimensions. As digitally-based technology becomes more widespread, it is almost certain to become even more miniaturized, less software-oriented, more user-oriented, and it should therefore offer improved prospects for truly interactive performance conditions in the arts. Paralleling these developments, the future of laser technology accessible to audiences may include holographic movies and 3-D color television generated by lasers on a large scale. Beyond this, most attempts at prophecy become mere speculation. But there is an excellent chance that yet another unimagined "breakthrough" technology will emerge in the very near future, to the delight and surprise of scientists, artists, and the public, just as laser technology has emerged to surprise and delight us within the very recent past. ■

REFERENCES

1. See Deborah Van Brunt: *Laser Light Show Safety: Who's Responsible?* Rockville, Maryland: U.S. Department of Health and Human Services, Bureau of Radiological Health, 1980. FDA Publication No. 80-8121.
2. Lowell Cross: "Audio/Video/Laser." *Source, music of the avant garde*, Issue No. 8 (1970), 28-36. "Musica Instrumentalis, Video II (B)/(C)/(L)." *Source*, Issue No. 9 (1971), 3-10. "Laser Deflection System." In Kluver, Martin, and Rose, eds., *Pavilion*. New York: E. P. Dutton, 1972, 235-237.
3. Elsa Garmire: "An Overview: Laser Deflection System." In Kluver, Martin, and Rose, eds., *op. cit.*, 184-186.
4. Nicolas Slonimsky, ed., *Baker's Biographical Dictionary of Musicians*, 6th edition, article on Scriabin. New York: Schirmer Books, 1978, 1569-70.

NIGEL BRANWELL

Dynamic Processing

Presenting a close look at the applications of dynamic processing—along with a review of the terms associated with this practice.

COMPRESSORS AND LIMITERS assist the engineer by providing system overload protection (preventing distortion, hopefully, and the potential for damage to following equipment). In addition, reducing dynamics permits a more acceptable or comfortable range of sound levels as well as increasing the apparent loudness and impact.

The above applies to all media—broadcast transmitters, tape, optical film tracks, public address, etc.

The Compressor/limiter is essentially a linear audio amplifier with a voltage-controlled attenuating element (VCA, FET, photo-cell or, looking ahead, digital). The control voltage is derived from the signal being processed in what is often termed the "side chain." The characteristics of the side chain determine the dynamic performance of the device; its sensitivity establishes the threshold level (the point at which gain reduction begins), and its loop gain above threshold controls the slope (or ratio) of the input-to-output level. The way it integrates and derives the control signal establishes its attack characteristics (peak, averaging, or RMS sensing). The system's performance, and its application potential, is largely a function of its slope (ratio) and attack/release characteristics.

LIMITING

Limiting can be described as overload protection; its purpose—to "limit" the signal at a specified level. Transients, composed of peaks of high energy but short duration, can exceed the pre-determined limit threshold—this is known as "overshoot." The degree of overshoot will be determined by the attack time.

Nigel Branwell is vice president of Audio + Design Recording, Inc., Bremerton, WA.

The effect of stopping every transient, no matter how fast, will almost certainly result in a lower average level with objectionable side effects such as "gritty" sound and switching spikes. Delay-line limiting can have zero overshoot without such side effects, but it will result in a lower-than-average modulation level.

To avoid these unpleasant side effects, it is preferable to have limiter attack times between 250 μsec and 1 msec. This allows the fastest transients (too fast for a peak sending meter) to overshoot and, in the extreme instance, saturate the tape. These overshoots are often undetectable, due to the momentary duration of the transients. This preserves the essential wavefront information which gives the transient its characteristic, reduces the side effects, and also increases mean level and overall sound for a given amount of gain change. However, in the case of broadcast transmitters, PCM links, disc cutters and optical film tracks where overshoot cannot be tolerated, a clipper can be included. It has been demonstrated that a limiter with a medium attack time followed by a clipper 1.5 dB above the limiter threshold, sounds far superior to an extremely fast (and gritty) attack time in the limiter.

There are limiters available with ratios of 100:1 (some even labelled ∞:1) which means that for every 100 dB change in level at the input (above threshold) there is 1 dB change at the output. Under normal conditions, the difference between ratios of 20:1 and 100:1 will be microscopic in terms of increased output, while the tighter ratio will certainly be more audible (and objectionable). Using the system described above, a ratio of 20:1 or 30:1 is all that is necessary.

The action of limiting must involve a peak sensing side chain, as it is peak level that is being controlled. When limiting, program dynamics are not greatly modified since gain reduction, when it occurs, is usually of small magnitude and duration. A fast release time is usual so that the action of recovery is inaudible; 6 dB of limiting, however, can make all the difference between background noise being audible or inaudible.

Limiting, then, allows the engineer to reduce his headroom, or overload margin, resulting in extending the dynamic range of the recording or transmission medium without fear of overload. In limiting, the ratio (slope) is termed "tight" because, whatever the increase in level at the input, the level at the output cannot rise significantly.

COMPRESSION

Compression refers to gain reduction which is more or less continuous. The original dynamics are reduced or "compressed." Compression ratios can vary from 1.5:1 or 2:1 (very soft and subtle) to the tightest limit slope of 20:1, depending on the effect required.

The ratio or slope refers simply to the relationship between the input and the output levels. In an amplifier, the normal ratio is 1:1. In compression and limiting, this relationship changes above the selected threshold and the output level rises and falls less than the input level; e.g., with a ratio of 2:1 selected, for every 10 dB rise above threshold at the input, the output will rise by only 5 dB.

To retain the maximum dynamics of the original signal while compressing, it is best to use a soft ratio with a slow or multiple release time. For a given amount of compression or gain reduction, the threshold on a soft slope/ratio will have to be lower than for a tight limit threshold—if we are to allow for headroom.

For the same degree of compression (or gain reduction) at two different ratios, the resulting sound will be different. On the softer ratio, it will be virtually undetectable: at 20:1 the sound will be more noticeably limited or "squashed."

FIGURE 2 graphically shows the relationship of input to output—notice how 10 dB of compression or gain reduction can be achieved. With a 1.5:1 ratio, the threshold is lower, which allows the device to act on a 30 dB range at the input. At 2:1, the threshold is higher and 10 dB variation at the input above threshold results in a 5 dB range at the output. At a ratio of 10:1 at a threshold of −10 dB, 10 dB of input level change results in only 1 dB of output level change. The softer the ratio, the more suitable are the effects of gain reduction and also the less apparent are the effects of fast release times.

ATTACK TIME

The attack time determines the shape and size of peaks allowed to pass through the system before gain reduction occurs—in effect dynamically modifying the static sine wave response of the chosen compression ratio. Slow attack can be observed on a PPM (peak reading meter) as overshoot and can be heard as a softening on a tight limit ratio. As attack time lengthens, a subtle change takes place in the spectral energy balance as more and more high frequency content passes unattenuated—in extreme cases leading to sibilant accentuation. FIGURE 3 shows various attack characteristics on a pulsed sine wave.

RELEASE TIME

Release time is very important in that it determines the moment-to-moment gain change in the device: and it is this which controls loudness. Under conditions of considerable compression, very fast release time, and a tight ratio, the mid to low level signal content is raised to peak level (see FIGURE 4); this increases apparent, or subjective, loudness. The program "sounds" louder but is at the same peak level. Taken too far, fast gain change becomes noticeable as "pumping" or "breathing," as background ambience and reverberation rise and fall in level. This can be used for effect, but when unwanted it can be minimised by either increasing the release time, or using a program-controlled (or "auto") release network; or reducing the amount of compression or softening the ratio.

A program controlled release is achieved by using a multiple network giving two or more release times, dependent on signal level. Its purpose is to provide maximum gain change without noticeable or objectionable pumping. This usually means a fast release over 4-to-6 dB gain reduction before slowing down to a medium or long release time. Often described as a gain riding platform, the auto-release function is ideal when considerable overall long-term compression is required (e.g. broadcasting).

In recording, where a fast rate of compression is desired, side effects can be greatly reduced by recording in a dead acoustic environment with good microphone separation and compressing prior to tape. By reducing reverb, ambience and leakage, it is surprising how much compression can be achieved.

Note that as release time is reduced, low frequency content is increasingly flattened by the attacking action on each cycle; the ear however, is very tolerant of LF distortion and therefore this is not a major problem and can be used to great effect while slowing the attack will "round" the distortion. For bass instruments, a release time greater than 0.4 seconds will give a totally clean sound.

NOISE & MODULATION EFFECTS

Self-generated noise in compressors and limiters should not be a problem in professional units. However, studio noise can be raised through compression of acoustic noises such as ambience, rumble and leakage from other instruments. Compressing off tape (certainly analog tape) can cause problems; 15 dB gain reduction will mean an increase of 15 dB in tape noise (unless an expander is used). Either way, there is little or no masking of high-frequency noise with a bass instrument and it is best to compress for effect prior to going onto tape.

Modulation of the signal by specific instruments can best be avoided by separately compressing instruments, or groups of similar instruments; there then being no dominant line to modulate another. It is impossible to compress or limit a high level low frequency signal without a very obvious and unpleasant modulation of the high-frequency signal and ambience—unless bandsplitting techniques are used.

There is often the need to compress a balanced (mixed) program where the dynamic range of the new medium may be more restricted (e.g. broadcast transmitter, disc-cutting). Modulation effects can be minimized by using a soft slope, a program controlled release network, or an averaging side chain. An increasing number of units allow for the insertion of an equalizer in the side chain (or control voltage) to modify the system response (Audio & Design's Vocal Stresser, SCAMP S01 Compressor, and Easy Rider Compressor/limiter are examples). Reducing the low frequency content in the side chain through an equalizer will reduce any modulation effects caused by bass instruments, so that compression is controlled from the mid-band signal. This can only apply to compression, since limiting may produce the unexpected as low frequency signals exceed the established limit threshold.

Boosting frequencies in the side-chain can also be used to advantage. The correct high-frequency lift will cause the compressor to operate on sibilants—normally using a tight slope with a fast attack and release time. Compressor gain should be adjusted so that attenuation only occurs in the presence of sibilance. This is best employed on a separate vocal track to avoid modulation of the whole program. Compressors should incorporate some system gain (usually 20 to 30 dB) so that normal line levels can be compressed by the amount of gain available and still appear at the output at standard operating level. This also allows A/B comparison between the direct and processed signal.

Audio + Design's model F769X-R Vocal Stresser.

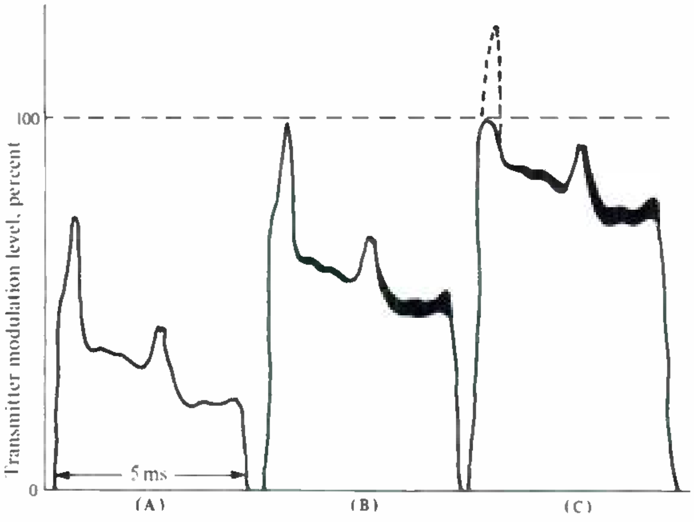


Figure 1. The effects of limiting and clipping. (A) The unprocessed signal, with peak level set to all headroom. (B) A delay-line limiter with 4 dB limiting at peak level. (C) A limiter/clipper combination. (The solid sections of the curves show the effect of a 25 msec release time.

Most compressor/limiters offer a range of ratios. While it may often be preferred to use the softest slope (1.5:1 or 2:1), this can only be done on a well-controlled signal. On a more unpredictable signal (e.g. vocals), the need for overload protection (limiting) as well as compression is often desirable. In this case, a compromise is necessary with the selection of a 5:1 or 10:1 ratio which may well sound inferior. The ideal, of course, is to be able to compress at a soft ratio while retaining a limit slope above the compressor for overload protection. The ability to vary the relationship between the compressor and limiter thresholds, and so determine the amount of compression before limiting commences, allows additional creative control. Simpler systems adopt fixed thresholds which, after 10 dB of compression (for example), tighten to the slope of a limiter.

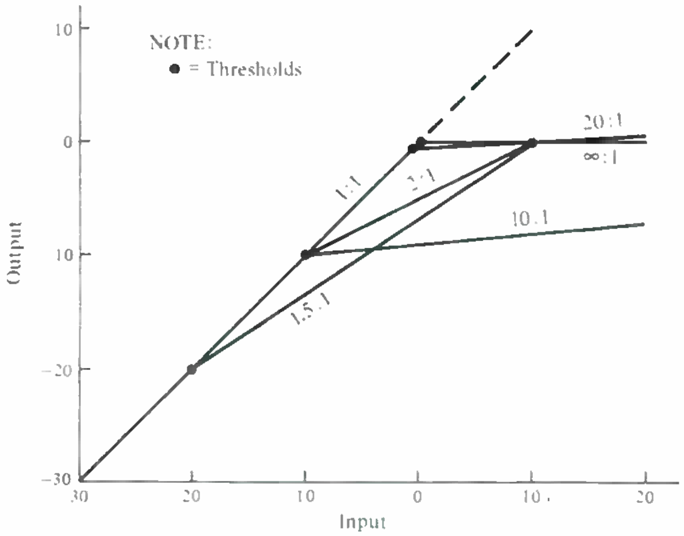
EXPANSION

The addition of an expander or gate adds a great deal to the effectiveness of a compressor/limiter. Apart from reducing increased source noise due to compression, the expander section can clean up tracks and dramatically reduce leakage from other instruments.

Gates can be described as limiters in reverse; typically, for a change of 1 dB at the input *below* threshold, the output falls by 20 dB, the rate being dependent on release time. Often critical to set up due to the switching characteristic, gates work well on punchy well-defined signals.

The expander, likewise, can be compared to the compressor; again, instead of acting on high level signal, it operates on low levels. The softer the slope, the easier it is to use without modulation side-effects; but the softer the slope, the less useful it is in attenuating noise effectively. In most recording applications, the purpose of the expander-gate is not to expand the signal, but to operate below the lowest level signals and attenuate the channel gain in the presence of noise only. For example, if the noise on a particular program lies 10 dB below the wanted signal, by setting the expander threshold just below the program it will be possible to lower the noise a further 10 dB with a 2:1 expander slope beyond which it will be held on the noise itself. A tighter slope of 4:1 would increase the separation of program-to-noise to 40 dB, but would also be more susceptible to modulation problems.

Figure 2. Typical compression and limiting slopes.



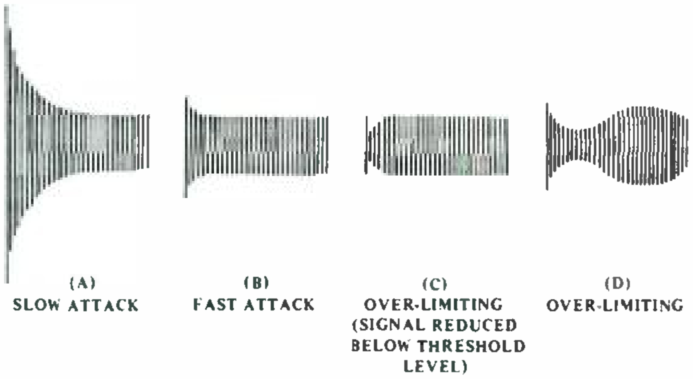


Figure 3. Attack characteristics on a pulsed sine wave. (A) and (B) show good waveform envelopes, as the signal is smoothly attenuated at the threshold level. (C) and (D) will sound restricted, and are examples of overlimiting and poor design.

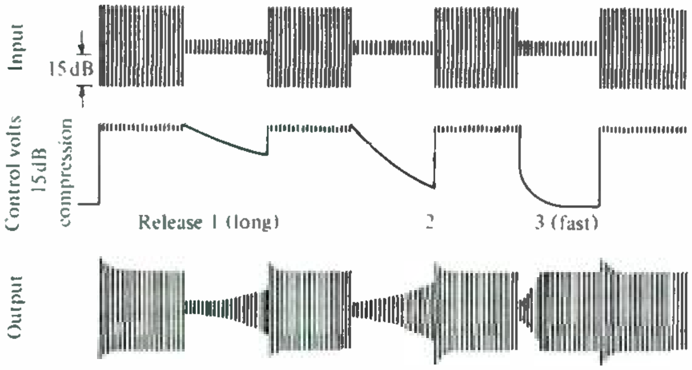


Figure 4. The effect of release time on mean level.

STEREO MATCHING

Stereo matching is an important aspect since gain reduction must track very closely between channels if there is to be no image shift during compression—although operating dual mono can increase energy level or density at the expense of image stability. Although mono units are often supplied with coupling possibilities, the potential user should investigate the manufacturer's stereo matching tolerances. Stereo units are likely to be more predictable in performance.

MULTI-BAND COMPRESSION

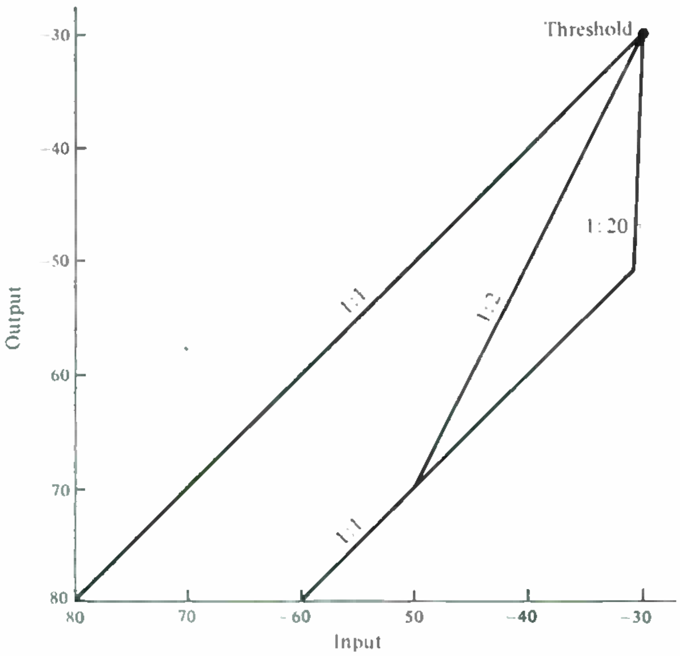
Multi-band compression, or "discriminate processing" (as it is often termed in broadcasting) has come to the fore recently, particularly in broadcasting but increasingly in those studios working in the areas of video and film post production, and disc-cutting facilities.

As previously discussed, there is a limit to how much fast gain reduction can be achieved from a broadband device without having the low frequency content modulate the high frequencies. By splitting the balanced program into two or more frequency bands, further gain reduction can be accomplished before modulation problems occur prior to recombining the broadband signal. This technique is of particular use where, in the following medium, the available dynamic range is less than the recorded material, or where more impact is required from the final medium; e.g. broadcast transmitters and recordings intended primarily for broadcast including all aspects of audio for video, optical film tracks, disc-cutting, and cassette duplication.

The process of band splitting (the accuracy of the filters and summing, the filter slope, and turnover frequencies used), the quality and flexibility of the band compressors, will all determine the overall quality of the results obtained from multiband processing techniques.

The creative possibilities of multi-band processing are many—from achieving a denser but dynamically-flat response to dynamic equalization (either for effect or to compensate for deficiencies in the following medium).

Figure 5. Expander gate slopes.



GAIN REDUCTION METERING

Using a meter or light column, the moment-to-moment gain change within the device may be displayed. In a combination device, this can include limiting, compression, expansion, and gating; although preferably there will be an additional indication of which section is functioning. Better still if the meter follows the release time—an important point to remember is that, while there may be a good deal of gain reduc-*tion* being read on the meter, it does not necessarily follow that there is a lot of gain *change* happening. One indication of increased loudness will be the *speed* at which the meter moves. Little movement of the gain reduction indicator will mean little loudness increase (other than a long-term increase in lower level signal). Whenever the change in dynamics of the program is faster than the release time, the program will hardly be affected dynamically whatever the amount of compression. The *rate* of gain change in a compressor/limiter determines loudness.

VU metering is widely used today. However, since the VU meter doesn't indicate peak level, the following variation in set-up procedure is advisable when using a peak sensing compressor/limiter.

If set up on tone, the VU will under-read by some 6 dB when operational on a compressed or limited dynamic signal, only approaching zero VU under conditions of fast gain change. Any system that uses VU metering must have good headroom (peaks of 10 dB or more being quite common). Thus it makes sense to set up under dynamic conditions so that the VU meter reads zero VU at least. Using a fast attack in this circumstance, the engineer can be confident that peak level is being well controlled 6 dB higher without fear of overload—well within the normal operating range. This may not apply to an RMS or averaging device where peak levels may be less predictable.

APPLICATIONS

In any recording, classical to rock, it is best to apply compression to the sections needing it rather than overall. If this is not possible, gain reduction will be restricted to between 6 dB and 10 dB, if its effect is to be inaudible. Up to about 6 dB can be achieved if limiting with a fast release (fast enough for recovery to be inaudible); beyond this, it is probably best to use an auto-release network, with a soft slope and a limiter standing guard above the compressor. In this way maximum dynamics are retained.

The effect of compression on signals containing plenty of presence frequencies, especially with ambience (i.e. choral work), is for the signal to recede as gain reduction takes place; using softer slopes will allow the sound to get louder and reduce the impression of a receding image. On bass or bass drum, using a tight slope with a fast attack and medium to slow release, a "bigger" sound will result as the decaying signal is lifted to the level of the original peak, creating a sustain. The acoustic environment will affect the sound, and is worth experimenting with.

Piano will sound great using a tightish slope, medium to slow attack and fast release. Likewise for rock vocals, where high average levels must be maintained to retain intelligibility. Some presence can be added after compression if desired. Otherwise vocals can be processed using softer slopes to retain expression and dynamic range. Compression with fast release will compensate for movement around the microphone. Where direct injection is possible, try compressing the direct signal (to avoid leakage) and mixing this with acoustic signal.

Weaker instruments (like violins) can be given more body using compression but watch out for leakage from headphones. Should this happen, a good expander will help maintain a clean track. Often with vocals or hand-clap overdubs, leakage from headphones will be a problem, and in this case gating will probably do a great clean-up job.

Gating or expanding the kick drum (depending on separation) can be effective. A fast attack will give a sharp edge (like a stick), while slower attack will result in a mellower, rounded "leather pedal" sound. Using a fast release, threshold should be adjusted until the cleanest sound is obtained. If there is a lot of leakage from cymbals (due to a badly placed mic?), a slower attack will help, the device responding to the drum rather than the cymbal. A gate with a frequency-conscious side-chain could also be very effective.

Selective expanders and gates or dynamic filters can be very useful. A highpass version can be used to attenuate low level acoustic rumble or hum until sufficient wanted low frequency signal masks it. At this point, the system would be adjusted to give a flat response. Similarly, a low pass version can attenuate electronic hiss or high frequency leakage around a bass instrument—opening to give a flat response in the presence of wanted high frequency transients and signal. On stage and in P.A. work, expanders can be very useful on vocal mics and direct injection instruments. Compression can add to the effective power of the system, while limiting is useful in the protection of amplifiers and speakers. Allowance should be made for increased gain on recovery, which will affect feedback levels.

WIDE DYNAMIC PROGRAMS (Classical)

In classical or other wide-dynamic range recording, high level compression causes a reduction in upper level dynamic contrast. Here an alternative form of compression can be arranged. By placing a compressor/limiter in parallel with the direct signal, it is possible to achieve low level compression, leaving the higher levels dynamically intact. As the program level rises, the slope gets progressively softer until finally returning to a 1:1 condition. To retain a correctly-related internal dynamic balance, it is best to have a very soft slope with a low threshold level. Compression then commences just above the lowest signal level; the compressed signal can then be a true reduction of the original.

One of the hidden advantages of this system is to soften even further the slope selected: for example, a 2:1 ratio is reduced to less than 1.5:1; while a 1.5:1 slope becomes 1.25:1 with a threshold of 60 dB down on peak level. The procedure is as follows: adjust the direct signal for required peak output; connect a compressor in parallel and select the lowest ratio available to give 20 dB reduction. Adjust the compressor to give 20 dB compression at peak input level; then set the peak output level of the compressor to be 10 dB below the peak level of the direct signal. The two signals are mixed and the effect will be around 12 dB overall compression.

This is similar to the Dolby system, but it would be unwise to use Dolby units as single-ended compressors since there will be considerable spectral energy distortion due to the action of the band processor. Using the above system in a multiband processing technique results in even subtler dynamic processing.

CHOOSING A COMPRESSOR/LIMITER

An ever increasing plethora of compressors and limiters are available today. Some are just excuses to make money (while they may reduce dynamics, the resulting noise and distortion negate their validity), while most are legitimate. Simpler devices, while easier to operate, must compromise on the versatility available, which in turn will restrict the contribution to creative processing. The more flexible units naturally require a higher degree of operational competence on the part of the user and demand that he understand the workings of the device to achieve the desired effect—hence this article.

For those engineers who take the time to fully explore and experiment with dynamic processing, the results can be extremely rewarding. ■

Ed's note—In September, author Branwell returns with a discussion of multi-band processing. Stay tuned!

TOM LUBIN

The Changing Art of Composing and Recording

Thanks to companies like TEAC, Fostex and Roland Corp., contemporary composing and recording can be financially and creatively satisfying.

FIFTY YEARS AGO, songwriting consisted of sitting at a piano, working out the parts, and writing the notes down. When (and if) the composition finally made it to the recording studio, it would have been completely scored, with all the parts for the individual instruments written out long in advance. The musicians would arrive and play what had been written, and the composer or arranger would get exactly what had been composed. Recording was more a documentation process than integral part of the creative process.

THE FIRST GENERATION

The introduction of the first tape recorders had little impact on this process, other than introducing the possibility of post-session editing. With tape recording in its infancy, the equipment was simple to use, and did not require an extensive arsenal of peripheral hardware. The cost of a complete recording system was well within the means of successful recording artists, and an extensive background in studio engineering technology was not really required.

Well, times have changed. Today, the complete recording studio is a formidable expense, and well beyond the means of all but a handful of the most successful recording artists.

In the meantime, the nature of the recording process has also changed (almost beyond recognition). Now, the aspiring (and perspiring) studio composer/arranger must write for a medium that rarely allows the luxury of "instant replay," especially if the music will be recorded over a time span of days, weeks or even longer. Today, in a successful production, the recording studio is an integral part of the creative process. Composition by committee is commonplace, with the producer, every member of the band, and possibly even the recording engineer all claiming a piece of the creative action.

For the musician who wishes to record a demo, or develop an idea at home, the difference between what can be achieved in and out of the studio grows ever larger.

THE SECOND GENERATION

When the Teac-Tascam series of tape recorders and mixing boards was introduced, many saw them as the much-needed pre-production tools for the artist at home. And although some artists have indeed acquired such hardware, its greatest impact has been in the exploding "semi-pro" recording industry.

Although the "semi-pro" label seems to be cordially disliked by all who are branded with it, it seems to be tenaciously hanging on as a catch-all description of anything that lies between "hi-fi" and the full-bore multi-track commercial recording studio operation.

However, even a well-equipped "semi-pro" operation may be more than the artist is ready to contend with. Many a performer who bought in, now finds himself the proprietor of a struggling recording studio, trying to rent time to others in order to support the hardware costs. Like the most ultra-sophisticated state-of-the-art recording system, this second generation of recording hardware can easily develop into a system too complex for the artist who simply wants to perfect his own craft, rather than become enmeshed in a new career as a recording engineer.

THE THIRD GENERATION

Re-enter Teac, along with Fostex (a "new kid on the block"). Both companies have taken the lowly cassette format and built it into a "personal" (*not* semi-pro) four-track recording system. The Teac 144 Portastudio combines a cassette recorder and a

Tom Lubin is the president of Creative Space. Formerly, he was the editor of Recording Engineer/Producer.

four-input mixer in one package. Two tracks can be recorded at a time, and it is possible to combine three tracks onto an open fourth track.

The system has high and low-frequency equalization, variable-gain input preamplifiers, a simple busing system, separate auxiliary sends that are used for echo, and an auxiliary return that combines to the stereo output bus. There is a separate cue system, and it is possible to monitor three previously recorded tracks while recording four inputs onto the fourth track.

The transport uses Dolby B noise reduction, a non-standard equalization curve, and runs at 3¾ ips. A variable-speed control and return-to-zero function are also provided.

The Fostex model 250 is similar in many respects to the Portastudio. It too is a self-contained unit, but it has provisions for a guitar-type footswitch to control punching in and out. Also, it is possible to simultaneously record all four tracks and to get four separate track outputs from the machine. Thus, safety copies and transfers to larger format machines can be done, and during mixdown it is possible to patch in external outboard gear in each track.

A digital display takes the place of the Portastudio's mechanical counter. The 250 also comes with two earphone outputs and peak indicators, in addition to the VU meters. The audio specs include the use of Dolby's new C type noise reduction; unfortunately, this makes the tapes made on the two systems incompatible.

Both of these units definitely satisfy the cost simplicity requirements for a personal multi-track system. The at-home musician can develop his ideas with relative ease, without going broke or getting preoccupied with coping with the technology of recording.

Of course, sooner or later four tracks aren't enough. If the Elcaset were still available, it might have become the ideal medium for an eight-track system, but since it's not, Fostex seems to have come up with a viable alternative. They have combined an inexpensive eight in/four out console with an eight-track/quarter-inch tape recorder. The tape runs at 15 ips, uses the IEC equalization curve, and Dolby C noise reduction. The recorder will reproduce all eight tracks simultaneously, but will only record four tracks at a time.

The console has eight mic/line inputs, each with two bands of parametric equalization. The auxiliary send bus is used for monitoring during recording and for echo sends or other types of signal processing during mixing. The board also has four phono inputs, with built-in RIAA networks. There are two earphone outputs connected to a bus-monitoring matrix and volume control network.

The transport has a return-to-zero feature, and provisions for a punch-in/punch-out foot switch. Like the four-track cassette systems, this transport has a single record/play head. The alignment controls are straight-forward, and not overly complicated. Performance is more than acceptable, and it is likely that with a price tag of about $3,500 for the board and recorder, it will find an avid marketplace waiting for it.

AUTOMATING THE MUSICIAN

As high technology has become more accessible—as witness the new-generation hardware just described—the performing musician has become more electronically sophisticated than in the past. However, many attempts to bring recording studio techniques to the concert stage have been marginally successful at best. This can be—to say the least—frustrating to the artist who tries, unsuccessfully, to duplicate (or at least, to simulate) his recorded sound during on-stage live performances.

Roland Corp. has long been well-known for its line of studio-type signal processing devices, re-designed to meet the specific needs of the performing musician. And now, Roland Studio Systems (a newly-formed division of the company) has introduced the CPE-800 Compu-editor, which bring automation technology within reach of the concert performer, as well as the recording studio whose needs and/or budget do not call for more costly systems.

The CPE-800 Compu-editor is specifically designed for semi-pro (sorry!) studios, live performances, and the expanding audio/visual market. Its versatility deserves some description, since it lends itself to several voltage-control and automation applications.

The Compu-editor is made up of a fader/control desk, an umbilical cord, and a rack-mounted voltage-controlled amplifier housing (VCA-800). For control room applications, the VCA mainframe should be patched into line level insertion points on the recording console. If these are not available, the VCA-800 may be inserted between the tape machine outputs and the console's line inputs.

The audio signal is confined to the amplifier mainframe, and does not pass through the Compu-editor's control desk. In order to keep it simple and affordable, the system does have some limitations in its software. For example, there is no update mode, so when a program is re-written the previous information is lost.

The Compu-editor system may be used as an audio automation system and/or to program lighting cues and machine controls. The basic system is capable of handling 15 audio signals, on the assumption that the 16th track will be reserved for time code, and does not need audio-type control functions. If more inputs are required, it is possible to interlock two Compu-editors.

The system has a large (32K) internal program storage capacity, so it is unnecessary to have an additional tape machine or pen track on the master tape. There is an internal clock, but the unit can interlock to any SMPTE code. For intricate mixes that require precise interlock to the tape machine, the Compu-editor provides an SMPTE data train output which can be recorded on an open track of the master tape.

The digital display provides a running indication of the program progression. Depending on the mode of the board, the readout will show the SMPTE code, the free-running internal clock output, or the scene number.

The internal mode is particularly well-suited for multimedia pre-programmed presentations, since it can provide a master clock for other elements in the presentation. This mode is also ideal for controlling the balance between synthesized or processed signals during a complicated composition. A synthesist can program an entire composition and at some later date recreate the blend. The Compu-editor is definitely a tool for live performance, and it has the "roadability" of other Roland products, as well.

For lounge-type shows, where there are no sound or lighting operators, the performer must often control these functions as well as entertain. Here, the CPE-800 seems to be ideal, since it can control both sound and lights. An act can preset all of its lighting/sound cues, and then control the advance of the program with a foot switch.

The "scene" mode is primarily designed for this sort of application. Thus operated, it can program thousands of different scenes in advance. This mode is also ideal for live theater, where a director wants lighting and audio settings accurately repeated for each performance. The operator does not have to remember the different elements of the cue, and only needs to advance the scene as he follows the script. A particular scene can be repeated without going to the top of the entire program.

In its normal operating mode, the system will go to the end of a scene and hold indefinitely. It is possible to repeat a scene and to halt the program in any of the modes. To go backward in the program it is necessary to return to the top and manually advance by using the "stroke" keys.

The memory capacity of the Compu-editor is approximately 100 hours. Because the memory is internal, and has a limited capacity, the amount of hours of information that can be stored will vary with the number of changes contained in the program. There is a display that shows the percentage of remaining storage while the program is being written. Typically, this indicates that more memory has been used than is actually the case. This is because the computer is able to reevaluate the information and compress it once the program has been completed.

The mixer has a volatile memory, and will lose the program if the power is disconnected. (A modification featuring a battery-operated memory is now being considered.) However, provisions have been made to connect a standard tape recorder to the unit and use it as a storage medium. The date can later be re-loaded into the computer. There is also a safety feature that provides verification that the recorded data is accurate and duplicates the program.

There is an additional indicator on the console that shows when there is sufficient data interlock signal. LEDs are also used for level matching when a program is being modified. Lights above each fader indicate if the present fader position is above or below the previously-written program. The Compu-editor also permits manual control of individual faders without disturbing the program. And, when the operator is sure that a change in the program is necessary, each of the inputs can be independently re-written.

The voltage-controlled amplifier uses an opto-isolator to control the gain of the amplifier. This type of device generally has a turn-off response that is slower than its turn-on time. In this case, a fast fade-in can occur in 10 ms, while the fade-out averages 100 ms. This is not a problem under normal conditions, but the amplifier logic circuit does provide a means of "rapid kill." When the control voltage drops to 0 (i.e., off), an FET mute switch turns off the audio signal.

The fader control voltage is segmented into 128 individual steps (110 steps of 0.39 dB from +9 dB to −36 dB and 17 steps of 2.7 dB from −36 dB downward). The last step is the zero-voltage position. The other controls on each input are the mute, read and write, along with a manual key. To prevent the accidental erasure of a program, the board automatically returns to a read mode after a program has been written.

There are provisions for an additional display of fader status. The console can be connected to an oscilloscope with XY inputs to give a fader position graph on the screen.

While many of the largest manufacturers were at the recent AES show to herald a digital explosion, and the increasing availability of highly automated control systems, a few companies had on display equipment that may have much more of an impact on the music recording industry because they will more closely effect the person who writes the song. Amongst the heady atmosphere of high technology one mustn't forget that increasing the signal-to-noise will not make better lyrics or a stronger melody. However, personal multi-tracks, and automation systems that can be used by the performing musician will change the way music is developed. The creators of composition will again be able to develop their ideas without someone looking over their shoulder. The cost for these systems is comparatively inexpensive and their simplicity and compactness will make them a standard tool for the song-writer. Musicians will take them on the road and turn any hotel room into a recording suite. By the time the composition reaches the studio (and for that matter, the other members of the band), the concept and the direction of the material will be developed to its author's satisfaction. Long hours working out parts will be cut down, and tighter budgets will be more prudently spent. The composers will know what they want to get out of the studio before they go in, as a result of their pre-production efforts. Songs will again be created by composers and not by committees. ■

JOEL SILVERMAN

Getting Those Effects Boxes to Behave

In this note, author Silverman gives some helpful hints on the proper use of effects boxes.

AFTER THE BEATLES, with everyone trying to out-produce everyone else, the sounds a musician could produce live were no match for the textures and effects coming out of the studios. And so, "home-made" effects suddenly began appearing on stage, followed almost immediately by manufactured effects boxes. In 1972, MXR appeared on stage with effects that were reliable and also very compact, so that many could be used together. As the '70s progressed, the musician gained access to all the effects the studios had, and in some cases demanded the stage devices be used in the recording process. This brought on engineering headaches galore—impedence, level, noise and distortion—as musicians and engineers tried to get the respective equipment to "talk to each other."

It is not uncommon for a musician to use several effects in a series. It is also not uncommon to hear a lot of noise and unwanted distortion from such a set up. Why? Usually, the order of effects is wrong. A wah pedal should not be the first hook-up in the effects chain! Unless the instrument's output

A do-it-yourself pedal board.

Joel Silverman is National sales and marketing manager, professional products group, for MXR Innovations, Rochester, NY

is already buffered, or frequency response is unimportant, a buffering preamp should be the first device. You can boost the signal level at the source, instead of adding gain at the end of the chain. A lot of so-called noise problems are simply due to gain at the wrong place. A strong clean signal to start with is mandatory. It's easy to attenuate at the end.

The same rules that apply in the control room should also be followed in the studio. In other words, don't put +4 dB into something that can only handle –20 dB. Some effects are designed for musical instrument (*not* line) level. In fact, FET phase shifters and many other devices can even be made to distort by hot pickups. This can result in a pleasant growl on signal peaks or just plain old (and wanted) distortion. By following the buffering preamp with a limiter or compressor, you can control the instrument's dynamic range *and* prevent overloading of effects down the line. But, don't use dynamics-dependent devices after the limiter. Since these devices vary their effect relative to the signal, their performance is severely degraded.

Distortion devices work very nicely after the limiter. The 140+ dB SPL needed to get that "singing quality" can be simulated very well with a Distortion II and limiter. Distortion works very well ahead of phase shifters and flangers. The increase in harmonic content offers more signal for the phaser or flanger to work with. The result is a more-intense sound.

The proper input level is very important with flangers, chorus and other analog delay devices. What can appear to be a too noisy device may just be a simple level mismatch.

A volume pedal at the beginning of an effects chain is also a disaster in the studio. All the residual noise is greatly amplified while the input level to the effects chain is almost nil. A volume pedal should be the very last device. A noise gate could be nice at the end if no volume pedal is used (no signal-no noise).

The Wah pedal usually works best at the end of the chain. The filter peak produced by the Wah can easily overdrive anything preceding it. TIME DELAY devices have been used to create a number of fascinating effects. If the device allows one to "sweep" or modulate the amount of delay, even more effects are possible. Flanging occurs in a time delay range from 0.25 milliseconds to 20 milliseconds. The sweep is usually wide and a fair amount of regeneration (or feedback) is used. In flanging (as in all "swept delay" effects), the more rapid the sweep, the less wide the sweep. This avoids a pitch-bending effect that is usually undesirable. The "Chorus" effect occurs within the range of 10 to 30 milliseconds. Here the sweep is not as wide as in flanging, and regeneration is generally not used.

Doubling may be obtained with a time delay of from 15 to 80 milliseconds. The sweep width is narrower still and, in fact, the sweep function may be defeated entirely. Usually little or no regeneration is used. A very good doubling may be achieved with a pitch shifting device. Merely offset the pitch by a small fraction of one semitone (or 1 percent of the input frequency). Again, no regeneration is necessary.

Discrete echo begins in the neighborhood of 80 milliseconds. Here, the sweep function is very rarely used and regeneration determines the number of repeats desired.

The use of batteries to power effects has been taking a lot of heat lately as a great lack of convenience, but it is definitely an advantage in the studio. Obviously, battery-powered effects will not generate AC hum in the effects chain. Also, their use will simplify grounding hassles. The local ground from a chain of battery powered effects can be easily tied to one point, avoiding ground loops.

Which type of battery should one use? As a rule of thumb, if a device draws an average current of 10 milliamperes or less, a regular transistor battery will do nicely for well over 24 hours of continuous use (assuming the device uses a 9-volt battery). If the device averages over 10 milliamperes, a heavy-duty or alkaline type battery should be used.

If you feel you must stock a supply of batteries, it may be effectively done in a refrigerator (*not* the freezer). Try to allow the battery to come up to room temperature prior to use. If the battery is leaking its electrolyte, discard it. ■

JOHN M. WORAM

The Mathematics of the Microphone, Part II

There was Jaws II, there was Exorcist II, so why not Mike Math II?

INTRODUCTION

In PART I of this article, we learned that all first-order polar patterns may be represented by the general equation, $\rho = A + B\cos\theta$. For comparison purposes, we may set A + B equal to 1, so that all microphones will have an on-axis sensitivity of 1 (that is, 0 dB attenuation).

When A = B, the polar pattern is a perfect cardioid, or, *limacon of Pascal*. As A becomes larger than B, the pattern moves toward omni-directional. The rear area expands until when A = 1 (and therefore, B = 0), we have the perfect circle of the omni-directional pattern.

On the other hand, if B becomes greater than A, the cardioid lobe becomes smaller and rounder, and a rear lobe begins to appear. Finally, when B = 1, the front and rear lobes are both circles, and we have the familiar figure-8 pattern.

Note that the value of A determines the omni-directional characteristic of the microphone, and is therefore referred to as the omni-directional component. Or, it may be called the pressure component, since omni-directional microphones are simply pressure-responsive devices.

In like manner, the value of B may be called the cosine or velocity component. Thus, each microphone may be thought of as having say, a pressure and a velocity component, whose relative values determine the microphone's polar response.

M-S MIKE MATH

In our letters column this month, reader Dave Burnham discusses some practical applications of various M-S combinations. Since M-S, as well as other stereo miking techniques, are once again becoming popular, let's extend our "mike math" into this area, and see just what is supposed to happen when various microphone pairs are combined.

More often than not, we encounter an M-S pair consisting of a cardioid and a figure-8 microphone, although as reader Burnham reminds us, there are many other combinations that merit investigation. When we use the cardioid/figure-8 pair, the M microphone is pointed straight ahead (0°) and the figure-8 is at right angles to it (90°). When the M and S outputs are combined, we get left- or right-oriented resultant patterns, depending on whether M and S are added or subtracted.

Through trial-and-error, most of us arrive at the combination of M and S that sounds best under the circumstances, and it doesn't really matter much that we don't know what the resultant polar equations are. After all, if it sounds good, who cares about A and B anyway?

On the other hand, a little M-S math—studied up front—may help us find our ideal setup just a little faster, or help us recognize which direction to go in when things don't sound as they should. As an added bonus, it will help us predict just

John Woram is the editor of **db** *magazine.*

what will happen when *any* two microphones are combined, at any angle.

First, let's talk the problem through: when the outputs of any two microphones, M and S, are combined, the resultant output, T, is M + S. You don't need to be Einstein to figure that out, nor to recognize that at *some* angle between the microphones the resultant output will reach its maximum. Obviously, this will be the "resultant angle," at which our M-S combo is pointing. We can even go a step further, and write $T_s = M + S$, and $T_d = M - S$. And, with lots of coffee and graph paper, we can eventually dope out the shape of the resultant outputs, and the angles at which these are pointing. Or, we can take the easy way out, by applying a little calculus.

At this point, some may opt for the easiest way out, that of forgetting the whole thing. But we needn't despair—the math is not really that fearsome, and little of it needs to be committed to memory.

To begin, just visualize a graph of almost anything that rises to some maximum, and then falls off again. Although the graphed curve may have a slope which varies continuously, at only one point will its slope be equal to zero. That is where the curve reaches its maximum.

Now, let the curve represent the combination of two polar equations. We know that at some angle, θ, we will reach our maximum output, and now we know that at that angle, the slope of the curve will equal zero. Well, we already know the equations for the two microphones ($M = A_1 + B_1 cos\theta$, and $S = A_2 + B_2 cos\theta$). With a little calculus, we can find the equation for the slope of this curve. Since we're interested in what's going on when the slope reaches zero, we can set the equation equal to zero, and solve it to find the angle at which this happens. This, of course, is the resultant angle that we are looking for.

To spare those readers who are not that excited about calculus any further grief (and our typesetter from heartburn), the math is worked out in a boxed insert at the end of this opus, where it may be cheerfully ignored by those who wish to do so. In either case, when we solve the equation, we find that when *any* two microphones are combined at 90 degrees, the angle of the resultant pattern is simply a function of B_1 and B_2. To be specific,

$tan\theta = B_2/B_1$, where
θ = the resultant angle between the microphones.

So, if one of the microphones is a cardioid ($0.5 + 0.5cos\theta$) and the other is a figure-8 ($0 + 1cos\theta$), then $tan\theta = B_2/B_1 = 1/0.5 = 2$, and therefore, $\theta = 63.43$ degrees ($tan 63.43 = 2$). Note that if we were subtracting outputs, our ratio would be $-B_2/B_1 = -2$, giving us an angle of -63.43 degrees.

In other words, the resultant polar patterns are angled at ± 63.43 degrees, with respect to the on-axis M microphone. This is a rather wide included angle (126.86 degrees), and chances are we would prefer to have an angle of say, ± 45 degrees. To get this, we must make $B_2 = B_1$, so that $B_2/B_1 = 1$ ($tan 45 = 1$). We can do this in several ways.

First, let's make that M mike a figure-8 instead of a cardioid. Now, $M = 0 + 1cos\theta$, instead of $0.5 + 0.5cos\theta$. Therefore, $B_2/B_1 = 1$, and we have our resultant angle of 45 degrees.

As another alternative, let M remain a cardioid, but with an output that is twice that of the S microphone ($M = 1 + 1cos\theta$, and $S = 1cos\theta$, as before). Once again, $B_2/B_1 = 1$, and we're back at 45 degrees.

As a third alternative, choose *any* microphone for M, and adjust the output so that, as before, $B_2/B_1 = 1$. For example, try a hyper-cardioid and a figure-8. The figure-8 must be $0 + 0.75cos\theta$, since the hyper-cardioid is $0.25 + 0.75cos\theta$. This gives us an M-S output ratio of 1:0.75, or 4:3.

From all of the above, we can see that when two microphones are combined at 90 degrees, the resultant angle is strictly a function of the cosine (or velocity) components (B_2 and B_1) of each microphone. Since these must be kept equal in order to maintain a 45-degree resultant angle, then we see that as the M mike varies from figure-8 to cardioid, its relative sensitivity must be continuously adjusted, to keep $B_2/B_1 = 1$.

But now, what are the actual polar equations of the resultant patterns? It may be important to know what these look like, so that we can adjust the pattern of the M microphone *before* recording if necessary. We can find out what the patterns look like by examining the contribution of each microphone to the resultant output.

The amplitude of the resultant output is a function of the pressure and velocity components of each microphone. Therefore, when we combine these components at the resultant angle, we discover the effective pressure (P) and velocity (V) components of the resultant polar pattern. For example, consider the typical M-S combination described earlier. The M mike is at 0 degrees, and the S is at -90 degrees. When the M and S outputs are *added*, the resultant angle is +45 degrees, and so we have

$$\begin{aligned} M &= 0.5 + 0.5cos[\ 0 + 45\ ]^* = 0.5 + 0.5(0.707) \\ +S &= \quad 0.5cos[-90 + 45] \qquad\quad + 0.5(0.707) \\ &\qquad\qquad\qquad\qquad\qquad \overline{0.5 + 0.707} \end{aligned}$$

*[The angles within the brackets indicate each microphone's own angle, and the resultant angle between the microphone pair.]

If we *subtract* the outputs, $B_2/B_1 = +1$, and the resultant angle is +45 degrees, giving us

$$\begin{aligned} M &= 0.5 + 0.5cos[\ 0-45\ ] \quad = 0.5 + 0.5(0.707) \\ -S &= \quad -0.5cos[-90-45] \quad = \quad -0.5(-0.707) \\ &\qquad\qquad\qquad\qquad\qquad \overline{0.5 + 0.707} \end{aligned}$$

Note that in either case, the M-S combination gives us a polar pattern whose on-axis sensitivity is $\rho = 0.5 + 0.707cos\theta = 1.207$. This means that the M-S system outputs at the resultant angles are $20 log 1.207 = 1.6$ dB greater than the M microphone's own on-axis output.

Now, in order to compare the resultant polar patterns with those we have examined earlier, we may set the sensitivity equal to 1, which gives us $\rho = 0.414 + 0.586cos\theta$. The pattern is thus seen to be cardioid-like. Since B is slightly greater than A, there is a small rear lobe. The 180-degree off-axis sensitivity is $0.414 + 0.586\cos 180 = -0.172 = -15.28$ dB attenuation at 180 degrees.

We have thus verified that our M-S combination produces resultant polar equations of $0.414 + 0.586\cos\theta$, and that the resultant angles are ±45 degrees. Of course, we can change the resultant angle later on, simply by varying the proportion of M to S. However, at each resultant angle, we will have to accept the resultant polar pattern that occurs. If we want a different polar pattern, we must adjust the M pattern before recording, as we shall see later on.

If the M and S outputs are combined before recording, and the resultant left and right outputs are recorded on separate tracks, what sort of pattern will we get when these tracks are combined later on? Of course, it should be a perfect (mono) cardioid, but let's see if it is:

$$
\begin{aligned}
R_l &= 0.414 + 0.586\cos(+45)\\
+R_r &= \underline{0.414 + 0.586\cos(-45)}\\
&\quad 0.828 + 0.828\cos\theta = 1.657, \text{ or}\\
&\quad 0.5 + 0.5\cos\theta = 1.
\end{aligned}
$$

Since the A and B components are equal, we know we have a perfect cardioid pattern, as expected.

Now, let's try combining two cardioid microphones at 60 degrees. If the microphone outputs are equal, we may realize intuitively that the resultant angle will be midway between them (+30 degrees) when the outputs are added, and at right angles to this (−90 +30 = −60 degrees) when the outputs are subtracted. This can be verified by using the longer equation described in the boxed insert.

Proceeding as before, we find that

$$
\begin{aligned}
X &= 0.5 + 0.5\cos[0 + 30] = 0.5 + 0.5(0.866)\\
+Y &= 0.5 + 0.5\cos[-60 + 30] = \underline{0.5 + 0.5(0.866)}\\
&\qquad 1 + 0.866\cos\theta
\end{aligned}
$$

and

$$
\begin{aligned}
X &= 0.5 + 0.5\cos[0 - 60] = 0.5 + 0.5(0.5)\\
-Y &= -0.5 - 0.5\cos[-60 - 60] = \underline{-0.5 - 0.5(-0.5)}\\
&\qquad 0 + 0.5\cos\theta
\end{aligned}
$$

In other words, adding these microphones gives us a cardioid-like pattern ($1 + 0.866\cos\theta = 0.536 + 0.464\cos\theta$), and subtracting them produces a figure-8 pattern (since the subtraction leaves us with a cosine component only).

X-Y FROM M-S

Since X-Y recording uses two cardioid microphones angled at 90 degrees, it may be possible to create an X-Y cardioid pair from an M-S microphone system, should the need arise. Using the math described above, we find that the following values are required:

$M = 1 + 0.707\cos\theta$
$S = 0.707\cos\theta$
or
$0.586 + 0.414\cos\theta = 1.0 = 0$ dB
$0 + 0.414\cos\theta = 0.414 = -7.6$ dB.

This should be well within the capabilities of any M-S microphone with a continuously-variable polar pattern selector. The M microphone's pattern is set so that its off-axis sensitivity is $0.586 + 0.414\cos 180 = 0.172$. This is equivalent to an off-axis attenuation of 15.28 dB. The S microphone is set so that its output is 7.6 dB lower than the M microphone.

Of course, unless you happen to have an anechoic chamber lying around, finding these precise values may be easier said than done. However, with a little pre-session trial-and-error, it should be possible to arrive at a reasonable approximation of these theoretical points.

And, if you want to know what microphone to use to get a certain amount of attenuation at any specified angle, the computer program here may be of interest.

```
100  INPUT "ENTER ATTENUATION, IN DB. ":N
110  INPUT "ENTER ANGLE, IN DEGREES.  ":TH
120  C = EXP(-.115*N)
130  FOR X = 1 TO 2
140  B = (1-C)/(1-COS(.01745*TH))
150  A = 1-B
160  IF B >1.001 THEN END
170  IF A <.0001 THEN 210
180  PRINT INT(A*100 +.5)/100
190  IF B <.0001 THEN PRINT: GOTO 230
200  PRINT TAB(5)"+";
210  PRINT TAB(6) INT(B*100 +.5)/100;
220  PRINT TAB(10)"COS(TH)."
230  C = -C
240  NEXT X
250  END
```

The program calculates the value(s) of A and B, for any desired amount of attenuation, at any angle (TH).

Depending on the values you enter, you may get one or two pairs of numbers for A and B. Or, you may get nothing at all. For example, if you ask for 90 dB attenuation at 10 degrees (nice try!), the computer will ignore your request, since this is beyond the capabilities of any first-order microphone.

CONCLUSION

After all the calculations described here have been done, remember that most of this is microphone *theory*. The practice part is up to you. However, it helps to know what the laws are, so that you can break them creatively, instead of by accident. ■

Two microphones are separated by an angle, α. Therefore, their polar patterns may be written as

$M_1 = A_1 + B_1\cos\theta_1$, and

$M_2 = A_2 + B_2\cos\theta_2$. Therefore,

$M_1 + M_2 = A_1 + B_1\cos\theta_1 + A_2 + B_2\cos\theta_2$. And,

$\theta_2 = \theta_1 - \alpha$. Differentiating the sum, S,

$\partial S/\partial\theta = B_1\sin\theta_1 + B_2\sin\theta_2 = 0$. Therefore,

$\dfrac{B_1}{B_2} = \dfrac{-\sin\theta_2}{\sin\theta_1} = \dfrac{-[\sin\theta_1\cos\alpha - \cos\theta_1\sin\alpha]}{\sin\theta_1}$. Or,

$B_1/B_2 = -\cos\alpha + \cot\theta_1\sin\alpha$. Therefore,

$\cot\theta_1 = \dfrac{B_1/B_2 + \cos\alpha}{\sin\alpha}$, and

$\tan\theta_1 = \dfrac{\sin\alpha}{B_1/B_2 + \cos\alpha}$. When $\alpha = 90$ degrees,

$\tan\theta_1 = B_2/B_1$.

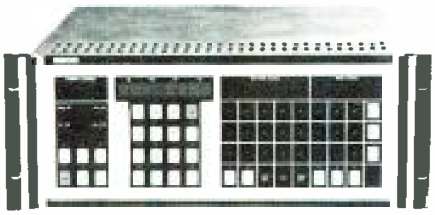
In the above derivation, θ represents the resultant angle when any two microphones are combined. When the microphones are separated by 90 degrees, the simplified form, $\tan\theta_1 = B_2/B_1$ may be used.

New Products & Services

MULTI-CUE SYNCHRONIZER

• EECO Incorporated has announced the availability of its MQS-100A Series Multi-Cue Synchronizer, a frame-accurate, microprocessor-based SMPTE/EBU time code instrument. The new model, an enhanced version of EECO's MQS-100, is equipped with several new features designed to increase the efficiency, precision and flexibility of video or audio tape production. Enhancements include transfer of time code information from any machine to any cue or Event register, variable pre-roll, Event offset capability, three scratchpad memories accessible from the keyboard and the ability to make mode changes on-the-run. With the MQS-100A, three Events can be programmed, each with its own time designation, and time offsets can be specified for events one and two. Typically, offsets are used for "punch in" recording to accommodate record erase delays. In addition, a third special Event has been added which automatically enables, rolls and synchronizes Machine three when the Master Time Code reaches the stored Event time.

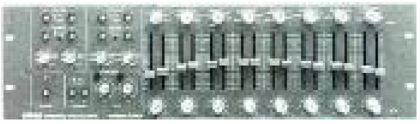
Mfr: EECO Incorporated
Price: $13,900.00
Circle 50 on Reader Service Card



STEREO EQUALIZER

• The 674A Stereo Equalizer features eight bands, graphic-type EQ controls, and continuously variable center frequency and bandwidth in each band. Wide-range high and low-pass filters with 12 dB/octave Butterworth slopes follow the EQ section which can be used as independent,tunable two-way electronic crossovers. Each of the eight bands tunes over a 3:1 frequency range and offers 16 dB boost or cut with reciprocal curves. The Q typically can be varied between 0.3 and 20 for extra-narrow notches. The high and low-pass filter sections are continuously tunable over 100:1 frequency range in two decades. Each section is independently switchable. The input is electronically balanced; the output is unbalanced with the balanced option available. Nominal output level is +4 dBm with the maximum output level before clipping being greater than +19 dBm. Total noise at the output is less than -78 dBm, giving a dynamic range of greater than 97 dB. THD and SMPTE IM are both less than 0.08% at +18 dBm out.

Mfr: Orban Associates Inc.
Price: $1,149.00
Circle 52 on Reader Service Card



DIGITAL CAPACITANCE METER

• The lightweight IT-2250 hand-held Digital Capacitance Meter measures capacitance values from .1 pF to 199.9 mF (0.1999 Farad). The IT-2250's Kelvin terminal design permits measurement directly at the capacitor leads to minimize error, according to the manufacturer. A remote extension cable is also provided for measuring capacitors which cannot be connected directly to the meter. An auto-ranging feature automatically selects the correct range of measurement from a choice of ten ranges. The value of capacitance is then shown on a 3½-digit LED display. A special low test voltage is provided for measuring electrolytic and other capacitors which have a low operating voltage. Clamp diodes, a fuse and resistor provide protection from excessive current.

Mfr: Heath Company
Circle 51 on Reader Service Card

SIGNAL PROCESSING EQUIPMENT

• The Dyna-Mite is a self-contained, self-powered processing tool capable of being rack-mounted or remaining portable. In the Mono version, the Dyna-Mite offers limiting, expansion, de-essing, noise gating, Kepexing and voice-over ducking. The Stereo version offers inter-coupling capability at the push of a switch, allowing any number of processing combinations, such as an expander followed by a limiter or a dual threshold peak and average limiter with independent release timer. Dyna-Mite is easy to interface and use, as it plugs in instantly with ring/tip/sleeve jacks to -10 or +4 lines, and is capable of driving 600 ohm loads. An optional battery pack is available.

Mfr: Valley People, Inc.
Price: Mono—$295.00, Stereo—$495.00
Circle 53 on Reader Service Card

AUDIO MONITOR MIXING CONSOLE

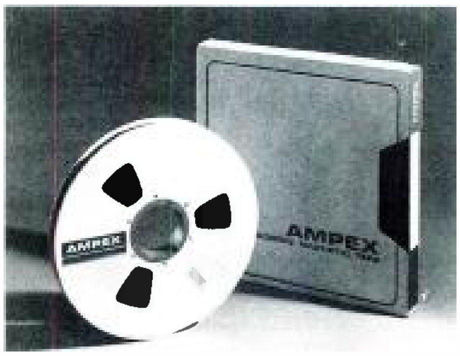
• A new monitor mixing console with six separate output mixes for on-stage monitor mixing, sound reinforcement, and recording applications has been introduced by Audy Instruments, Inc. The Audy Series 2000M provides 16 inputs (stackable to 32) with separate output mixes that permit control of up to six independent monitor sends. Using high speed, low noise IC op-amp technology, it minimizes transient and slewing-induced intermodulation distortion. A dual LED system assures proper adjustments of input attenuation switches and maintains 25 dB of headroom throughout for clean sound. Other features include: input and output channel patching; EQ in/out switch for each input mix control; individual channel muting; talkback; 20-segment LED bargraph, and 6 auxiliary inputs.
Mfr: Audy Instruments, Inc.
Price: $6,995.00
Circle 54 on Reader Service Card

LAVALIER MICROPHONE SERIES

• The 647C Series consists of three models—the 647CH, 647CL and 647CLS. They are high-impedance, low impedance and low impedance with an integral locking on/off switch, respectively. The 60-10,000 Hz frequency response of these three models is designed to match the range of the human voice and will fit virtually any public address or sound reinforcement application. The 674CH, 647CL, and 647CLS all measure 3⅝-in. long by ¾-in. diameter.
Mfr: Electro-Voice
Price: 647CH—$95.00, 647CL—$92.00, 647 CLS—$99.95
Circle 58 on Reader Service Card

DIGITAL AUDIO TAPE

• The newest addition to Ampex Corporation's professional audio tape line—466 High Energy digital—made its debut at the 69th AES convention in May. Ampex 466 High Energy digital tape uses a durable binder system that improves runability and reduces dropouts for sustained low error rates. Its greater packing density in comparison to standard energy tapes accommodates narrower track widths. The tape is backcoated to reduce static generation and improve handling and winding characteristics. The new tape is available in appropriate configurations for all digital open reel audio recorders.
Mfr: Ampex Corporation
Circle 55 on Reader Service Card



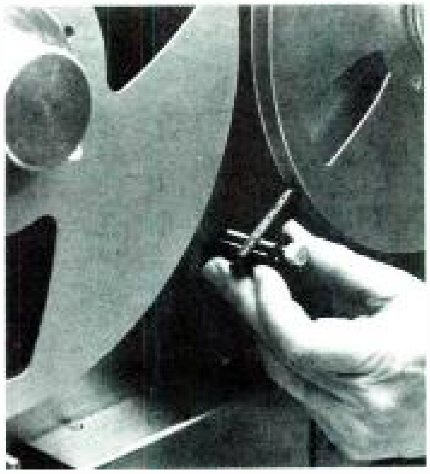
OSCILLOSCOPE

• The V-1050 100 MHz oscilloscope features a sensitivity of 500 V/div (5 MHz). Four channel capability permits the simultaneous display of four signals. A total of eight traces can be seen with operation of alternate time base feature. The V-1050 offers calibrated delayed sweep capability. Time base A sweep rate ranges from 20ns/div to 0.5s/div in 23 calibrated steps. Time base B sweep range is 20ns/div to 50ms/div in 20 calibrated steps. There are four horizontal display modes: A Only, A Intensified, Alternate and B Delayed. The 6-in. CRT has an acceleration voltage of 20 kV. Internal graticule, variable scale illumination and P31 phosphor is standard. A TV synchronization circuit for video applications is standard in the V-1050. Other features include variable trigger hold-off, beam finder, single sweep capability and front panel X-Y operation.
Mfr: Hitachi Denshi America, Ltd.
Price: $2390.00
Circle 56 on Reader Service Card

8-TRACK MIXER

• The M-35 modular mixing console has 8 mic/line inputs, 4 buss outputs and an independent 8-track monitor mix. Microphone inputs are transformer isolated. The M-35 also features an 8-track cue system, 4 effects returns and direct outputs on each input channel. Equalization on the input channels is the parametric sweep type. Either of two low frequency ranges can be selected (60 Hz-400 Hz or 400-1.5 kHz) as well as either of two high frequency ranges (1.5 kHz-7.5 kHz or 7.5 kHz-12.5 kHz). Boost or cut for both is ± 12 dB. If more input channels are desired, an expander is available, offering up to an additional 12 mic/line inputs. A talkback module is also available.
Mfr: TEAC Corporation
Price: $2,300.00
Circle 57 on Reader Service Card

RECORDER CARE PRODUCTS



• Nortronics has introduced two additions to its Proformance line of recorder care products—the PF-710 and PF-720 alignment gauges. The PF-710 provides three critical adjustments for any broadcast cartridge player. The zenith adjustment is 90 degrees ±5 minutes (compares to NAB standards of ±15 minutes). The PF-710 also checks tape guide and track height to ±.001-in. The PF-720 open reel alignment gauge adjusts to check tape guide on all currently available recorders. A locking thumb screw prevents movement while verifying the consistency of all tape contact points.
Mfr: Nortronics
Circle 59 on Reader Service Card

db New Literature

PROFESSIONAL AUDIO BUYER'S GUIDE

• SIE Publishing has introduced its new audio professional buyer's guide. According to its manufacturer, the guide is the most complete source of audio sound and recording equipment information. Included in its pages are over 70 manufacturers and thousands of professional products ranging from speakers and amplifiers to wireless microphones and computers. Mfr: **SIE Publishing, Box 4139, Thousand Oaks, CA 91359.**

HEATHKIT CATALOG

• In a new edition of its free 104 page catalog, Heath Company is introducing 15 new products plus educational courses and computer software. The publication features products in stereo high fidelity, test instruments, microcomputers, and televisions. Some of the new kit products being introduced include: a 2-meter amplifier and a deluxe antenna tuner. In addition to the electronic kits, the catalog also highlights an educational section of self-study courses. For the engineer and technician, there are high technology specialties such as microprocessors and optoelectrics. Mfr: **Heath Company, Dept. 350-800, Benton Harbor, MI 49022.**

REVISED D SUBMINIATURE CATALOG

• A 60-page revised catalog of D Subminiature rectangular connectors for use in aircraft, missile and ground support systems has been published by ITT Cannon Electric. Catalog D-15 includes 75 photographs, eight cutaways and 70 drawings of the Original-D, Burgun-D, Golden-D, Royal-D, Hermetic-D, Filter-D, and Mas/Ter-D connectors. Additional features include accessory information, voltage/current ratings, panel mounting and tools assembly instruction. Mfr: **ITT Cannon Electric, 666 E. Dyer Rd., Santa Ana, CA 92702.**

CONNECTOR CATALOG

• Switchcraft, Inc. has published a new 36-page catalog on audio and general purpose connectors and AC receptacles. The catalog includes product descriptions, full engineering specifications, detailed drawings, and mating charts showing connecting compatibility with similar products. Mfr: **Switchcraft, Inc., 5555 N. Elston Ave., Chicago, IL 60630.**

EASYRIDER BROCHURE

• Audio & Design have produced their brochure for the Gemini Easyrider Compressor/Limiter in six languages. The brochure offers full details of the Easyrider. With help from Audio & Design agents worldwide, the brochure was translated into French, German, Spanish, Italian and Portuguese. Plans are also in the works for a Japanese translation. Mfr: **Audio & Design (Recording) Ltd., 16 North St., Reading, RG1 4DA England.**

SOUND EQUIPMENT CATALOG

• Lear Siegler's Bogen Division has announced its new catalog of sound equipment. In sixteen pages, the catalog covers all the firm's public address products, beginning with Bogen's line of amplifiers. The catalog also contains complete information on Bogen's mixer-preamplifiers, tuner, receiver and equalizers, as well as speakers, microphones and stands and accessories. Mfr: **Bogen Division, Box 500, Paramus, NJ 07652.**

MI SERIES CATALOG

• The new MI Series catalog details Sescom's audio transformer line with pertinent data including complete technical specifications, mechanical specifications, and user data. Mfr: **Sescom, Inc., 1111 Las Vegas Blvd. No., Las Vegas, NV 89101.**

Circle 32 on Reader Service Card

Circle 28 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

Minimum order accepted: $25.00
Rates: $1.00 a word
Boxed Ads: $40.00 per column inch
db Box Number: $8.50 for wording "Dept. XX," etc.
Plus $1.50 to cover postage

Frequency Discounts: 6 times, 15%; 12 times, 30%

ALL CLASSIFIED ADS MUST BE PREPAID.

FOR SALE

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.**

BGW: FOR IMMEDIATE DELIVERY. **UAR Professional Systems, 8535 Fairhaven, San Antonio, TX 78229. 512-690-8888.**

FOR SALE: YAMAHA and BGW power amps, AKG-BX10, MXR EQ, Yamaha PM180 mixer, Ashley crossover. **Call George (516) 921-9421.**

Sony ElCaset-EL7, 3 head, 3 motors, Dual Capstan closed loop, Dolby, ¼" tape cassette format. 3 of them, new, @ $350.00 ea. **Willie Heinz, ComCorps, (202) 638-6550.**

DISC CUTTING SYSTEM, Westrex 2B head rest of system stereo, Scully Lathe 2 track Ampex, Limiter, Eqs., Altec/McIntosh Monitors, priced to sell. **Soundwave, 50 W. 57 St., N.Y.C. 10019 (212) 582-6320.**

ORBAN. All products in stock. FOR IMMEDIATE DELIVERY. **UAR Professional Systems, 8535 Fairhaven, San Antonio, TX 78229. 512-690-8888.**

TASCAM, BGW, JBL, EV, Nikko, Technics, Onkyo, Hafler, Audio Research, Conrad Johnson, dbx, etc. **P. K. Audio, 4773 Convention, Baton Rouge, LA 70806. (504) 924-1001.**

FOR SALE: EMT 250 Digital Reverb Unit, mint condition. $17,500. **RPM Sound Studios, 12 E. 12th St., New York, NY 10003. (212) 242-2100.**

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, Colo. 80218.**

AGFA MASTERTAPE AND CASSETTES. Super prices. Example: ¼" x 2400' bulk = 9.82 and C-60 for $.86 (case quantities) send for wholesale price list. **Solid Sound, Inc., Box 7611, Ann Arbor, Mich. 48107 (313) 662-0667.**

dbx 155: FOR IMMEDIATE DELIVERY. **UAR Professional Systems, 8535 Fairhaven, San Antonio, TX 78229. 512-690-8888.**

AMPEX, OTARI, SCULLY—In stock, all major professional lines; top dollar trade-ins; write or call for prices. **Professional Audio Video Corporation, 384 Grand Street, Paterson, New Jersey 07505. (201) 523-3333.**

FREE CATALOG & AUDIO APPLICATIONS

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

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

UREI: FOR IMMEDIATE DELIVERY most items. **UAR Professional Systems, 8535 Fairhaven, San Antonio, TX 78229. 512-690-8888.**

FOR SALE: STEREO SELA MIXER 2880 St 8x2 fully equipped, equalization, peak meters. **Call Ed or Phil (212) 777-5580.**

AMPEX MM-1100 16-track, $11,000. Electrosound 2tk in console with variable speed, $1800. Audio Designs noise gates (4) in rack, $800. **(303) 473-1114.**

Lexicon Prime Time: FOR IMMEDIATE DELIVERY. **UAR Professional Systems, 8535 Fairhaven, San Antonio, TX 78229. 512-690-8888.**

ELECTRO-VOICE SENTRY Studio Monitors, Otari Recorders, dbx Systems, AB Systems Amplifiers, Harmon/Kardon hi/fi products. Best prices—immediate shipment. **East: (305) 462-1976, West: (213) 243-1168.**

101 RECORDING SECRETS MOST ENGINEERS WON'T TELL, $7.95 **Tunetronics, P.O. Box 55, Edgewater, N.J. 07020.**

AKG, E/V, Sennheiser Shure, Neuman; FOR IMMEDIATE DELIVERY most models. **UAR Professional Systems, 8535 Fairhaven, San Antonio, TX 78229. 512-690-8888.**

PROFESSIONAL AUDIO EQUIPMENT

Shop for pro audio from N.Y.'s leader, no matter where you live! Use the Harvey Pro Hot-Line. (800) 223-2642 (except NY, AK, & HI) Expert advice, broadest selection such as: Otari, EXR, Ampex, Tascam and more. Write or call for price or product info:

Harvey Professional Products Division
2 W. 45th Street
New York, NY 10036
(212) 921-5920

AKG BX20 reverberation and C414 microphones. FOR IMMEDIATE DELIVERY. **UAR Professional Systems, 8535 Fairhaven, San Antonio, TX 78229. 512-690-8888.**

NEW DEALER SALE—Entire line: Shure, Electro-Voice, Tapco. 20% to 30% off Pro User Net. Shure Pro Master 700 $799, EV PL5 $54, Tapco CP120 Pwr Amp $478. **Sound Communications Co., Box 121, Olpe, KS 66865, (316) 475-3537.**

LEXICON 224 Digital Reverberation. FOR IMMEDIATE DELIVERY. **UAR Professional Systems, 8535 Fairhaven, San Antonio, TX 78229, 512-690-8888.**

FULLY RECONDITIONED and warranted Ampex AG440-A-2 stereo ¼" in console, $3450, FOB California. **Barrett Associates, Inc., 318 Juniper Avenue, Carlsbad, CA 92008, (714) 729-9020.**

AMPEX, OTARI & SCULLY recorders in stock for immediate delivery; new and rebuilt, **RCI, 8550 2nd Ave., Silver Springs, MD 20910.** Write for complete product list.

SCULLY, NEW and used: FOR IMMEDIATE DELIVERY. **UAR Professional Systems, 8535 Fairhaven, San Antonio, TX 78229. 512-690-8888.**

SERVICES

MAGNETIC HEAD relapping—24 hour service. Replacement heads for professional recorders. **IEM, 350 N. Eric Drive, Palatine, IL 60067. (312) 358-4622.**

JBL AND GAUSS
SPEAKER WARRANTY CENTER

Fast emergency service. Speaker reconing and repair. Compression driver diaphragms for immediate shipment. **NEWCOME SOUND, 4684 Indianola Avenue, Columbus, OH 43214. (614) 268-5605.**

VIF INTERNATIONAL will remanufacture your Ampex or Scully (Ashland/Bodine) direct drive capstan motor for $200. Average turn around time—2-3 weeks. For details write **PO Box 1555, Mtn. View, CA 94042, or phone (408) 739-9740.**

ACOUSTIC CONSULTATION—Specializing in studios, control rooms, discos. Qualified personnel, reasonable rates. Acoustilog, Bruel & Kjaer, HP, Tektronix, Ivie, equipment calibrated on premises. Reverberation timer and RTA rentals. **Acoustilog, 19 Mercer Street, New York, NY 10013 (212) 925-1365.**

CUTTERHEAD REPAIR SERVICE for all models Westrex, HAECO, Grampian. Modifications done on Westrex. Quick turnaround. New and used cutterheads for sale. Send for free brochure: **International Cutterhead Repair, 194 Kings Ct., Teaneck, N.J. 07666. (201) 837-1269.**

ELECTRONICS DESIGN ENGINEERS

Beltone, one of the most respected names in precision electronics manufacturing, is undergoing unprecedented growth and is seeking engineering-oriented personnel for a major expansion of the Research & Development Division. We are located in Chicago in a very desirable northwest side area. The following positions offer an opportunity to work in a sophisticated and advanced technical environment.

MANAGER/ELECTRONIC ENGINEERING

Must be capable of supervising high technology engineers involved in the design and development of miniature low-power analog electronics. Your responsibilities will be to initiate new product concepts and develop them through the introductory stage into production.

SENIOR ELECTRONIC DESIGN ENGINEERS/AUDIOMETRIC SYSTEMS

Responsible for the design and development of audiometers and related products; required qualifications include a BSEE, 3-5 years professional experience, plus proven ability in the areas of analog circuit design and commercial product development.

PROJECT ENGINEER/ADVANCED DEVELOPMENT

Responsible for advanced development work in circuit analysis, linear low-voltage analog circuit design and electroacoustics; qualifications include a BSEE plus strong analytical capabilities.

These positions are accompanied by competitive salaries and comprehensive fringe benefits. To arrange a confidential interview, please send your detailed resume, complete with salary history and requirements, to:

BOB RIEFKE

Beltone

ELECTRONICS CORP.
4201 W. Victoria
Chicago, IL 60646

An Equal Opportunity Employer m/f

WANTED

Full Time **INSTRUCTORS** Part Time

FOR AUDIO RECORDING TECHNOLOGY

To teach basic and advanced theory and operation of multi-track recording studio equipment.

Audio field experience required. Teaching experience preferred.

Location—historic Greenwich Village.
Salary—attractive; commensurate with background, experience and responsibility of position.

Send detailed resume, including salary history to:

Philip Stein, Director
Institute of Audio Research
64 University Place
Greenwich Village
New York, N.Y. 10003

People/Places/Happenings

• **James L. Camacho** has been appointed to the position of director of marketing at **Lexicon, Inc.** According to **Ron Noonan**, president of Lexicon, Jim Camacho will assume responsibility for all aspects of Lexicon's audio products marketing including direction of distribution and field sales operations. Jim Camacho has been active in the audio field for 19 years, previously holding marketing and sales posts at **dbx**, **Delta Lab Research**, **Acoustic Research** and **H. H. Scott**.

• **MCI, Inc.**, of Fort Lauderdale, Florida, and **Sony Corporation** of Tokyo announced today that MCI has adopted the common format for multi-channel stationary-head digital audio announced last year by Sony Corporation and **Willi Studer** of Switzerland. MCI is the largest manufacturer of multi-track recorders and studio consoles in the United States.

• **Ampex Corporation**, a subsidiary of **The Signal Companies, Inc.**, announced the appointment of Ampex executive vice president **Charles A. Steinberg** to the additional post of chief operating officer. Steinberg will be responsible for the day-to-day operations of the company.

• **Dolby Laboratories Inc.** of San Francisco announced the appointment of **Bradley C. Stribling** as vice president, Product Development; **David P. Robinson** as vice president, Advanced Development; and **Scott P. Schuman** as manager, Recording Industry Products.

• **R. T. (Rudy) Bozak**, a pioneer in the high fidelity reproduction of sound, received an honorary Doctor of Engineering degree from his alma mater, **Milwaukee School of Engineering** at the college's commencement exercises on May 23. Bozak's early career involved the development of controls for sound equipment for the **Allen-Bradley Company** in Milwaukee. In 1949 he built the first loudspeakers bearing the Bozak name. From the beginning, the **R. T. Bozak Manufacturing Company** grew to be a major factor in the manufacture of sound reproduction equipment, both for the home and for commercial applications in theaters, concert halls and outdoor amphitheaters. His achievements have previously been recognized by the **Audio Engineering Society**, which has named him a fellow, awarded him its bronze medal of outstanding accomplishment in 1970 and which elected him to the **Audio Hall of Fame** in 1977.

• The appointment of **Robert Cook** to the position of national sales manager at **Electro-Voice** was recently announced by **Bob Morrill**, E-V's vice president of Marketing and Sales. Cook comes to Electro-Voice from **Magnetic Video Corporation, a Twentieth Century Fox** subsidiary, where he was national sales manager.

• **Paul Nagle** has joined the **Fostex** sales team at **Interlake Audio** as corporate sales manager, and will be developing an intensive dealer support program for the more than 50 dealers currently handling Fostex transducer products throughout the United States and Canada. The announcement was made by **Michael A. Gillespie**, president of Interlake Audio Inc. Mr. Nagle brings over 13 years of experience in electronics technology, sound reinforcement system design, marketing and sales management to the Fostex operation.

• **Westwood Recording Studios** has announced the installation of another MCI JH-114 16-Track Recorder with Autolocator II and TVI, a **BTX** 4500 SMPTE Time Code Synchronizer, and a BTX 4100 Time Code Generator. The new MCI LH-114 16-track is currently being used in sync with Westwood's existing MCI-JH-114 24-track recorder. The two MCI multitracks with SMPTE give Westwood Recording Studios a 40 track capability with 38 usable tracks, (less 2 tracks for SMPTE). Westwood Recording Studios is proud to be the first studio in Arizona with SMPTE for +24-track applications, as well as having facilities for complete Automated Mixdown with the MCI JH-636 console.

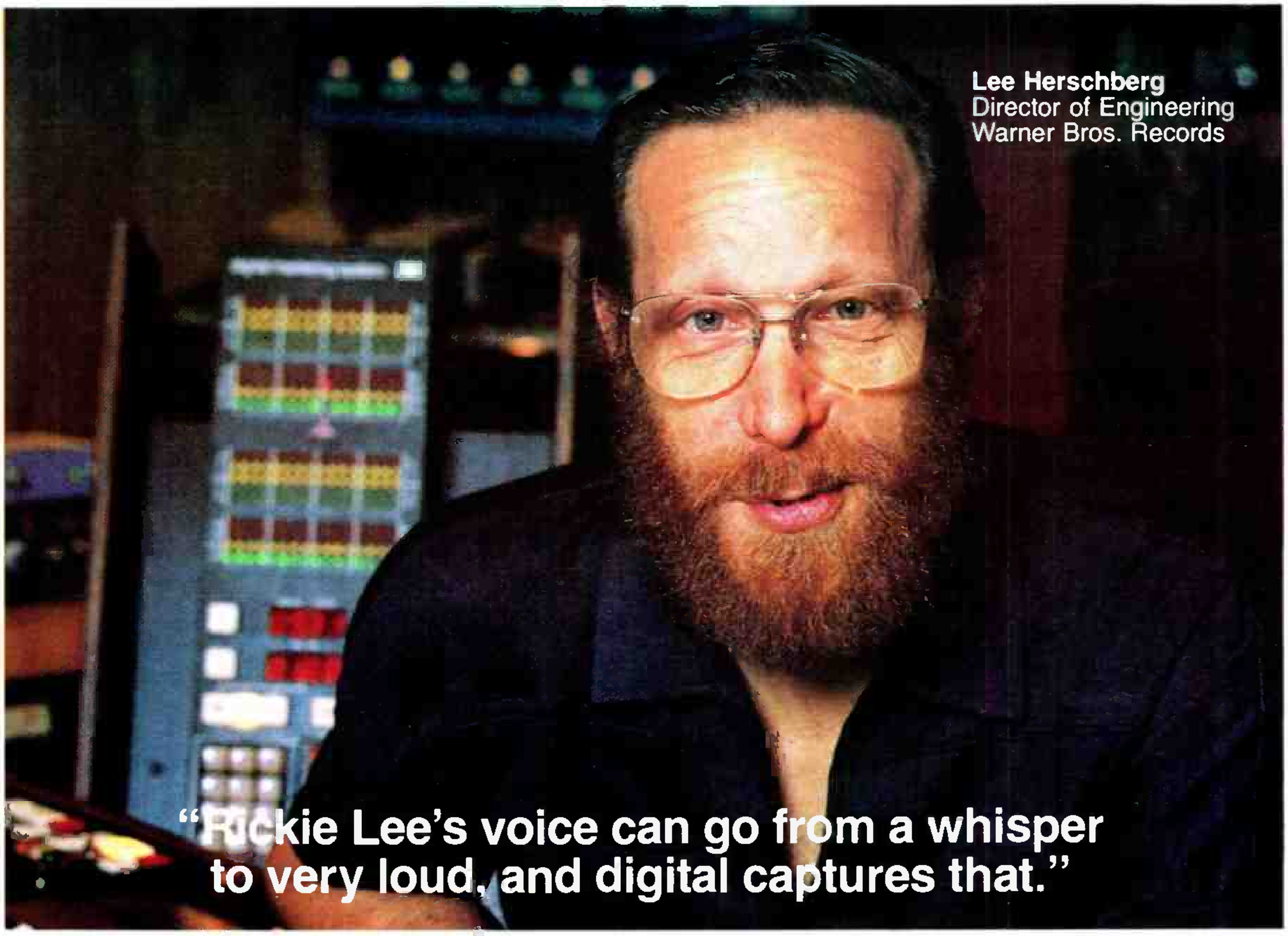
• **Empirical Audio** has just delivered a custom 56 channel automated recording/remix desk to **Record Plant Studio** in New York City. The desk, built by **Trident Audio Developments**, U.K., is the first of its kind to be installed in a studio in the United States. Trident's tradition has been to have an independent input and monitoring section. This board, however, built to the exacting specifications of Record Plant, necessitated extensive modifications to accommodate 56 channels of automated remix capability. The TSM at Record Plant is an "In Line Monitor" style console. This console is fitted with another first; Melquist VCA bypass faders on each input.

• **WNCN** (104.3 FM), New York's 24-hour classical music radio station, has recently celebrated its fifth anniversary under **GAF Corporation** ownership. According to **Matt Biberfeld**, general manager, WNCN has doubled its audience in those five years. GAF Corporation purchased WNCN on June 6, 1976, reinstating its classical music format which had been changed to rock-and-roll by the previous owner. WNCN's five-year record is exemplified by its recent capture of the **George Foster Peabody Award**, the Pulitzer Prize of broadcasting.

• **Gotham Audio Corporation's** president, **Stephen F. Temmer**, announced the return to Gotham of **Jerry Graham** after several year's absence. Graham will be in charge of Gotham's dealer sales organization which markets **Neumann** microphones, **TTM** Noise Reduction Frames and Gotham Cable throughout the United States and Canada.

• **Norman Baker**, president of **Valley People, Inc.**, has announced the appointment of **Ray Updike** to the position of vice president and general manager. Updike's audio experience dates back to 1969, when he became service manager for **R. A. Moog, Inc.** Since that time, Updike has worked for **Willi Studer America, Inc.**, and most recently, **Technicon Marketing Inc.**, where he served as president. In addition to the Updike appointment, Baker also announced the appointment of **Liz Clark**, previously executive assistant at Valley People, to the position of sales and marketing coordinator for products manufactured by Valley People.

• **James B. Lansing Sound, Inc.** recently supported two live experimental music broadcasts aired by Los Angeles radio station **KPFK** with the loan of JBL Professional Series amplifiers and studio monitors. Made possible by a grant to KPFK from the **National Endowment for the Arts**, the first concert was an electronic music piece titled "Busobong"; the second featured works by composers **Morton Subotnick** and **Joan La Barbera**. The mid-January performances were also taped for syndication over 200 public radio stations this fall, making the NEA-funded series the first programs of experimental music to be distributed nationwide via satellite.



Lee began his engineering career with Decca in 1956, moved to Warner Bros. in 1966, and became Warner's Director of Engineering in 1969. His experience spans the recording of such artists as Frank Sinatra, James Taylor, and most recently, Rickie Lee Jones. Herschberg is a true believer in digital recording, and agreed to tell us why.

Q. You've probably had as much experience with the 3M Digital System as anyone.

A. Yes, probably. I've been working with it for two years and had one of the first systems. We've been through the ups and downs and it's been well worth it. At this point, the 3M digital machine works as well as most analog machines.

Q. How do you justify the extra expense of digital recording?

A. Well, I think from any studio point of view, you've got to have the equipment that will bring in the artists. And if digital recording is truly the state-of-the-art, you've got to consider the clients you'll attract, and their needs.

Q. You've obviously done a lot of projects digitally. Why?

A. To me, digital recording is almost like the tape machine is nonexistent. You don't have any of the inherent problems you have with analog. I think everybody is aware of the major benefits of digital recording. No wow or flutter, lack of tape noise and no need for noise reduction. And digital allows you to do things you couldn't do with analog. Like compiling 3 or 4 tracks onto one. There's no degradation of quality.

Having 32 tracks has helped, and so has the addition of a digital editor.

Q. What do you say to an artist who's considering a digital project?

A. I'd say, yes, if it's up to me, go ahead and do it with digital.

Sometimes, on an analog session when the digital is available, I'll record the first couple of tracks on both machines. Then, on the first couple of playbacks, we'll listen to them side by side. That usually does it right there. There's no comparison.

There's nothing wrong with analog recording. And never has been. It's just that, with digital, you're hearing on playback what you just did in the studio. And you begin to hear all the shortcomings of analog machines — the things you've come to accept. And suddenly, those things are no longer acceptable.

Q. What musical formats are suited to digital?

A. Any format, really. It's particularly good for music with a lot of dynamic range. Like Rickie Lee.

Q. What would you say to other engineers and producers considering digital?

A. Well, digital isn't for everybody. And I'm not trying to say it is. There will always be people who prefer analog, and a lot of great records are made that way. It's just that, to my ears, digital is far superior, and it's the next logical step.

Lee Herschberg recently recorded Rickie Lee Jones on the 3M Digital System. The album, Pirates, is available from Warner Bros. Records.

3M Hears You . . .

3M

Circle 11 on Reader Service Card

The MTR-10 Series

Bridging the gap between the new technology and old reliability, they're the logical extension of the innovative technology built into our multichannel MTR-90. The new MTR-10 Series gives the professional unprecedented control:

- Measurable and audibly discernible performance improvements. More than chips and buzz words, a balanced engineering approach: adjustable phase compensation with internal square/sine wave generator; electronically balanced I/O with direct coupled outputs; Sendust™* heads for maximum accuracy and long life.
- Full servo, D.C. PLL transport governed by an on-board microprocessor—an industry first.
- Unmatched production features—exclusive multiple edit modes, reverse-play, standard alignment level presets, and dual-mode varispeed. Other features include controlled wind, preset master bias switching, three speeds and AES, NAB and IEC selectable. Both include return-to-zero and offer an optional tape locator with ten position memory and tape shuttle.
- Comprehensive servicing and support. Fully modular power supply, audio and transport electronics card frame. A small and highly dedicated dealer service network with factory trained and supported technicians.

Contact us at (415) 592-8311 for the name of your nearest, selected MTR-10 Series dealer. Let the awesome performance of the Newest Workhorse put you in control.

*TM JVC Corporation

The Newest Workhorses

OTARI

Otari Corporation, 2 Davis Drive
Belmont, California 94002
(415) 592-8311 Telex: 910-376-48

Circle 12 on Reader Service Card