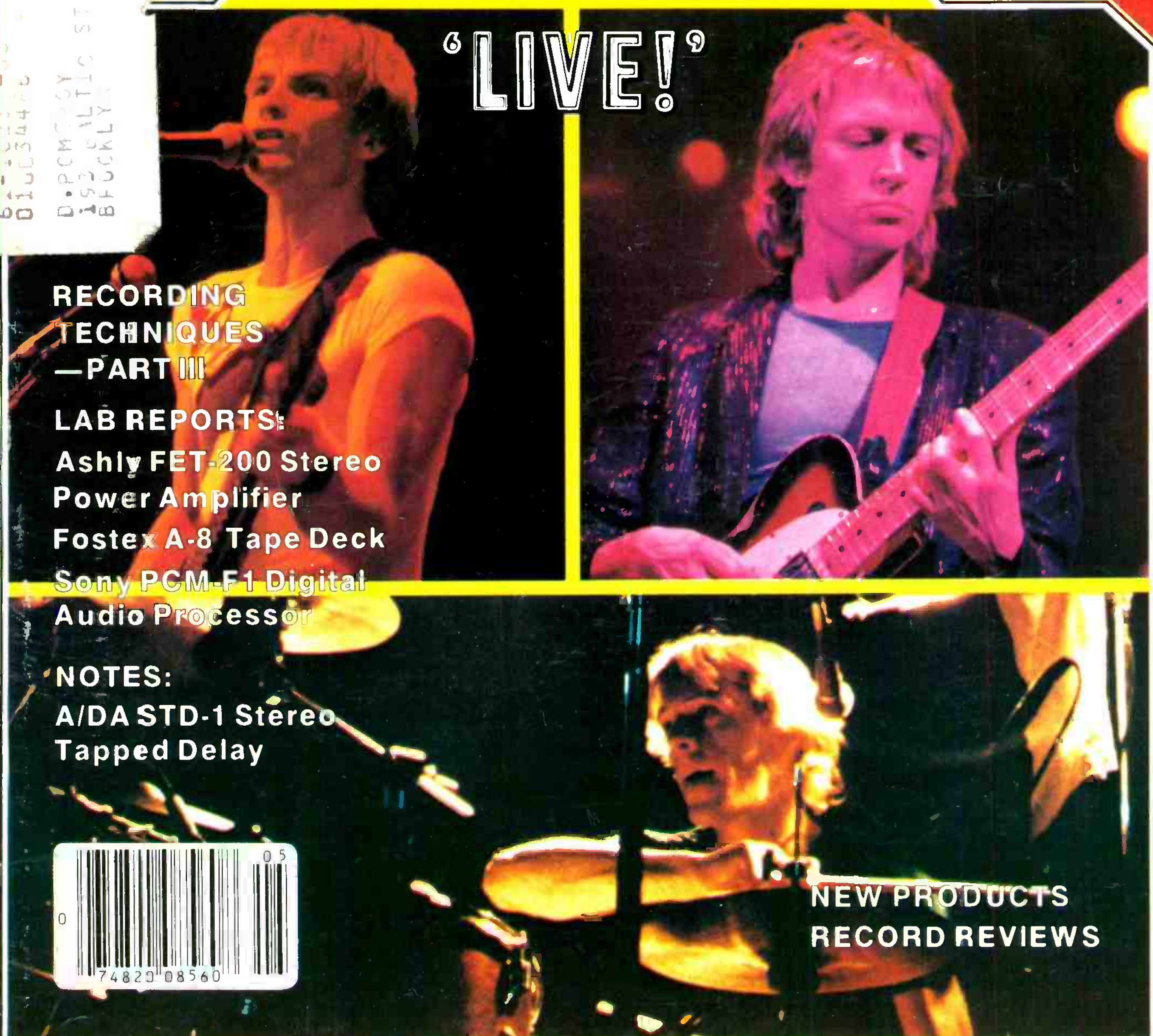
Profile: MOE BANDY

MODERN RECORDING & MUSIC

ICD 08560
$1.95

THE POLICE 'LIVE!'



RECORDING TECHNIQUES —PART III

LAB REPORTS:
Ashly FET-200 Stereo Power Amplifier
Fostex A-8 Tape Deck
Sony PCM-F1 Digital Audio Processor

NOTES:
A/DA STD-1 Stereo Tapped Delay

NEW PRODUCTS
RECORD REVIEWS

More time on your hands from MXR.

Providing extended delay times at an affordable price, the new MXR Model 151 Delay System II gives you over three full seconds of delay (three times that of similarly-priced digital devices). Specifically, the Delay System II can offer you up to 800 milliseconds of clean, quiet delay at a full 16 kHz bandwidth (over 200 milliseconds more than the closest competitor). As a digital recorder, the Delay System II's exceptional memory capability lets you capture entire musical phrases or obtain a wide variety of dynamic and *musical* studio-quality effects from flanging and chorus to echo and doubling in one rugged package.

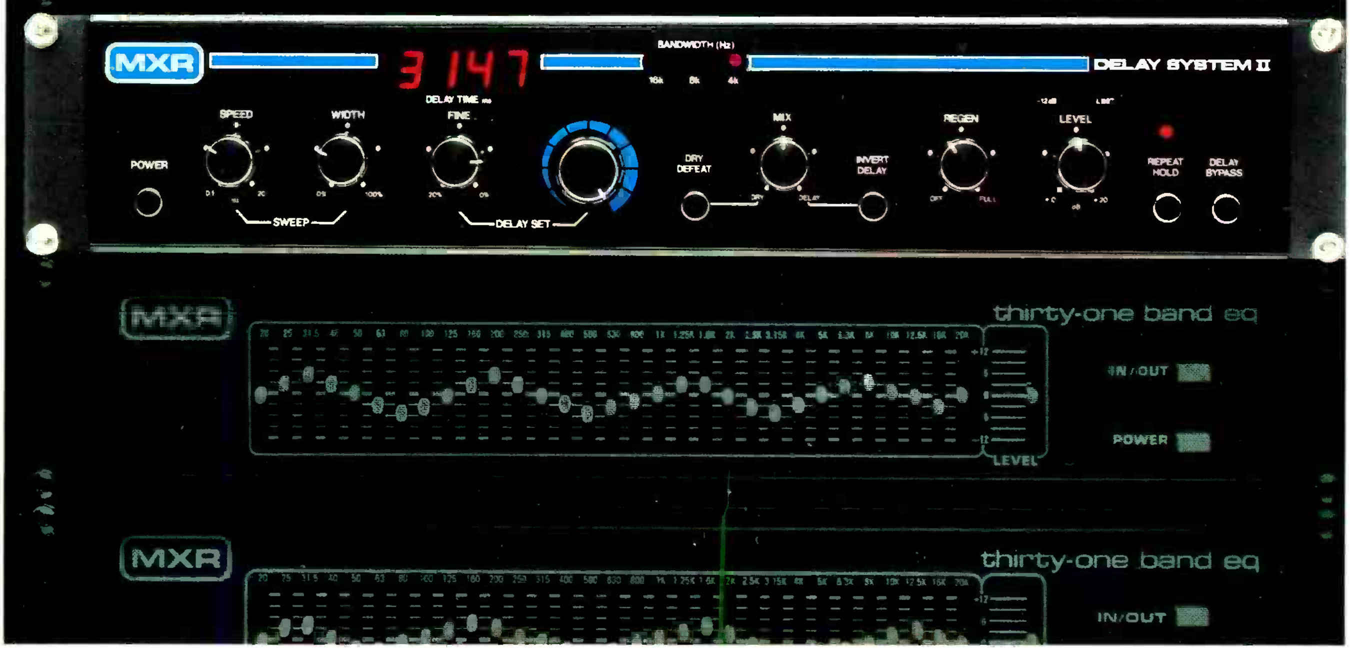
And it's easy to use in real time. The large front panel and simple control format make it a snap to quickly select from a vast range of time delay effects. The Delay System II's high-resolution four digit readout displays the precise amount of delay and the bandwidth is indicated by LED's, so there's no "squinting & thinking" to find out exactly where you are. Level-indicating LED's let you set up the optimum level in seconds.

The Delay System II fits right into your rack, looks great and provides clean, noise-free performance. A level switch is provided to optimize signal-to-noise for professional/home recording and onstage applications. The Delay System II also features easy access with both XLR and phone jack connectors (inputs *and* outputs) on the rear panel for instant interfacing with your patch bay.

MXR Delay System II — More time on your hands and more ways to use it as a creative tool on stage and in the studio. Hand-assembled in the U.S.A. with the finest components available in a compact, easy-to-use rack-mountable package.

MXR Professional Products Group

MXR Innovations, Inc.
740 Driving Park Avenue
Rochester, New York 14613
(716) 254-2910



Start a Studio. Under $400.

The TASCAM MM-20 Mixer is the heart and the start of a professional quality four-track recording system, which uses the same quality components as our larger systems.

And you can start recording with it today for under $400.

We were able to keep the price so reasonable by taking the unconventional, and extremely logical step of replacing all the switches — which are expensive and have fixed, limited logic — and replacing them with patch points all along the signal path.

The MM-20 incorporates four line inputs and two XLR transformer isolated microphone inputs. Need more mics? Add them four at a time with our EX-20 mic expander. You've also got six output busses, ready to change with the job just by changing the patches.

When you go from basics to overdubs to remix, again, just repatch.

When you need EQ, plug in our four-channel, three-band PE-20 Parametric Sweep type equalizer. For extreme flexibility in shaping sound.

And to read signal outputs anywhere in the system, patch in our MU-20 Meter Bridge with its four VU-type averaging meters and peak LED's.

There's a lot more to this ingenious system. Just visit your TASCAM dealer.

Or, write us for details.

So if you're going somewhere in recording, look into the mixer that can grow there with you.

7733 TELEGRAPH ROAD, MONTEBELLO, CA 90640

TASCAM MM20 Recording Mixer.

CIRCLE 123 ON READER SERVICE CARD

MAY 1982
VOL. 7 NO. 8

MODERN RECORDING & MUSIC

THE FEATURES

COMING NEXT ISSUE!
Recording With Stevie Nicks
Profile: The Tubes
An Overview of Synthesizers, Part II

Modern Recording & Music (ISSN 0273-8511) is published monthly by Cowan Publishing Corp., 14 Vanderventer Ave., Port Washington, N.Y. 11050. Design and contents are copyright 1982 by Cowan Publishing Corp., and must not be reproduced in any manner except by permission of the publisher. Second class postage paid at Port Washington, New York, and at additional mailing offices. Subscription rates: $15.00 for 12 issues; $26.00 for 24 issues. Add $3.00 per year for subscriptions outside of U.S. Subscriptions must be paid in advance in U.S. funds. Postmaster: Send Form 3579 to Modern Recording & Music, Cowan Publishing Corp., 14 Vanderventer Ave., Port Washington, N.Y. 11050.

THE STAPLES

Cover Photos: Lynn Goldsmith/LGI (Sting and Summers), courtesy of A&M Records and Jay E. Nolan (Copeland)
Police Color Photos: Lynn Goldsmith/LGI, courtesy of A&M Records
Police B&W Photos: Jay E. Nolan and Jeff Nehrbas
Bandy Photos: Courtesy of Rogers and Cowan

MODERN RECORDING & MUSIC

Editorial and Executive Offices
Modern Recording & Music
14 Vanderventer Ave.
Port Washington, N.Y. 11050
516-883-5705

Cowan Publishing Corp., Richard A. Cowan, Chairman of the Board & President; Cary L. Cowan, Vice President; Jack N. Schneider, Vice President, Marketing; Amy C. Gilman, Secretary/Treasurer; Sanford R. Cowan, Founder & President Emeritus.

Editorial contributions should be addressed to The Editor, Modern Recording & Music, 14 Vanderventer Ave., Port Washington, N.Y. 11050. Unsolicited manuscripts will be treated with care and must be accompanied by return postage.

Letters to the Editor

Synthesizers

Could you give me the addresses and phone numbers of the following synthesizer companies? Passport Designs; SMS; Oberheim; Linn; Microtune; E-mu; Multivox. Thank you.

—Francois Ledoux
Montreal, Canada

This is what we came up with:

Passport Designs
Headquarters:
P.O. Box 478
La Honda, CA 64020
415-747-0614
Marketing:
P.O. Box 21061
Minneapolis, MN 55421

SMS
P.O. Box 40267
San Francisco, CA 94140
415-824-4837
East Coast Office:
8 Tyler
Norwell, MA 02061
617-659-2618

Oberheim
1455 19th Street
Santa Monica, CA 90404
213-829-6831

Linn Electronics, Inc.
4000 W. Magnolia Blvd.
Burbank, CA 91505
213-841-1945

Microtune
104 Charles St.
Boston, MA 02114
617-262-6267

E-mu
417 Broadway
Santa Cruz, CA 95060
408-429-9147

Multivox Corp. of America/
Sorkin Music
370 Vanderbilt Motor Parkway
Hauppauge, NY 11787
516-231-7700

.isten to Jay Graydon, Steve Lukather
and Dean Parks and You're
.istening to Ibanez.

JAY GRAYDON

STEVE LUKATHER

DEAN PARKS

E400
ULTI-EFFECTS
rack mount
npressor
aser
ercrive
reo Chorus/Flanger
ternal Effects Loop
ta-Patch patching matrix
mote FET footswitch panel

E405
ULTI-EFFECTS
rack mount
npressor/Limiter
reo Chorus
rametric EQ
alog Delay
ternal Effects Loop
ta-Patch patching matrix
mote FET footswitch panel

D202
NALOG DELAY
rack mount
lay, Doubling, Stereo Chorus
d Flanging modes
adroom indicator
mplete delay modulation
ntrols
parate dry and delay
puts

ust listen to the playing of studio greats Jay Graydon, Steve ukather and Dean Parks and chances are you're listening to e sound of Ibanez effects.

ach of these guitar monsters uses the Ibanez UE400 and E405 Multi Effects Units and the Ibanez AD202 Analog Delay in his bag of studio tricks. Why? Because they're clean, quiet, versatile and most of all, they sound great.

Wherever you play–live or in the studio–use the next generation in effects–Ibanez.

Ibanez

The Studio Effects Connection

For a full-color catalog send $2.00 to: IBANEZ Dept. MR
P.O. Box 469, Bensalem, Pa. 19020 327 Broadway, Idaho Falls, ID. 83401
In Canada-6355 Park Ave., Montreal P.Q. H2V4H5

CIRCLE 177 ON READER SERVICE CARD

In addition to contacting the companies we've listed here, may we suggest you look at the article, "An Overview of Synthesizers, Part I," by Devarahi, appearing in the April 1982 issue of MR&M. *It contains quite a bit of information that you might be able to use, plus several addresses of synthesizer manufacturers that you may want to use.*

The Needs of Small Groups

I like *Modern Recording & Music* very much. However, much of the information you broadcast is simply over my head, or does not apply to my needs.

I am the vocalist for a five piece group which works an average of 40 nights a year in honky-tonks, gin mills, V.F.W.'s and American Legions, private parties, country clubs and dance halls in a geographic area where the population of towns averages about 5,000.

When we stroll into a music store in this area the electronic devices in stock include a fuzz, a wah-wah, and maybe, just maybe, a phaser.

Everybody knows what a nightmare mail order can be and often is. With this in mind, do you suppose you could put together an article or series of articles which focus on the needs of a vocalist? Also, you could include information on how to use delays, chorus effects, compressors, etc. You could also talk about "voice-doubling." What kind of microphones are best for a voice which is often working in baritone to bass frequencies is another possible topic.

It is a pity what a bass voice can do to "Hot-Spot" type monitors. Perhaps also in an article you could mention a couple of complete systems including mikes, mixer, effects, amps, speakers, and monitors, all of which would be compatible with each other and in the 5,000 dollar range.

I'm sure by now you're wondering whether we'd also like you to come here and run everything for us, too!

We know it's asking a lot, but we are also sure the interest and response would be considerable. After all, there have got to be about 20 little groups like ours who only play on weekends for every one group which has attained the degree of technical and professional competence your magazine seems mostly oriented towards.

—Dave Young
Miles City, MO

You've given us some line-up of possible topics. Just how many of you weekend players are there out there? We may be able to do as you ask. But there is an article which appeared in the November 1978 issue of MR&M *by Bruce Swedien, called "Vocal Miking Techniques." In most of our live sound studio articles you'll find detailed listings of mikes used by the players. There you'll also find the vocal mikes outlined. We suggest also that you always read the Talkback column of* MR&M *for information on vocal miking. And we'll see what we can do about your suggestions.*

8-Track

Now that you've done articles on multi-track magic for the 4 track studio why not now turn your attention to those of us who have recently upgraded to 8 track and do a series of articles on 8 track multi-track techniques? In the summer of 1981 I got an Otari MX 5050-8SHD (8 track) and an Otari MX 5050-B2HD (2 track) to mix down to. Working with 8 track is a whole new dimension because we are working with stereo drums, greater quantities of overdubs and more complex and flexible mixes. Right now I'm making do awkwardly with an 8 in/4 out console left over from my 2 years of 4 track work (it's a Tascam model 3 with two model 1's) while I save to buy a first rate low impedance 16 in/8 out console ($14,000-$15,000 range) this April. When I did 4 track work, I concentrated exclusively on live concerts featuring guitar/bass/drums trios, recording the drums in stereo and putting bass guitar on both those tracks. Except for an occasional vocal repair I didn't really do much studio stuff until shortly before I got the 8 track. From the very start I started collecting state of the art effects units which are now housed in an anvil rack. I've got an MXR Digital Delay (2 boards), Eventide FL201 Instant Flanger, Eventide FL201/BPC101 Phaser, 4 dbx 160 Compressor/Limiters, Orban 111B reverb, Orban 622A parametric and Yamaha P-2200 amp monitored through JBL 4311B's. I opted to do without noise reduction until this next summer, so I used limiting and kept the levels up between 0 and +2 VU.

I'd like to know where Otari Corporation is currently located. The ads for the largest machines say that they're in Belmont, Calif. and the ads for their smaller machines (like the MX 5050 series) say that they're in San Carlos, Calif. I've tried writing them there about a minor problem with my

Now Technics lets you hear nothing but the sound of the source. Introducing the SV-P100 Digital Cassette Recorder.

No tape hiss. No wow and flutter. Not even head contact distortion. With Technics new SV-P100, they no longer exist. The result—now you listen to the actual music . . . the source, not the tape or the tape player.

Utilizing the Pulse Code Modulation (PCM) digital process, the SV-P100 instantaneously translates musical notes into an exact numerical code, stores them on any standard VHS cassette, then "translates" them back into music on playback. Duplicate tapes are exactly the same as the original. Thus, every recording and every copy is a "master."

The revolutionary size of the new Technics SV-P100 recorder (17"x11"x10") is the result of state-of-the-art semiconductor technology. The built-in videotape transport mechanism brings the convenience normally associated with conventional front-loading cassette decks to a digital application. Tape loading is now fully automatic. And, frequently used controls are grouped together on a slanted panel with LED's to confirm operating status.

Despite its compact size, the SV-P100 recorder offers performance beyond even professional open reel decks. Since the digital signal is recorded on the video track, the space usually available for audio can therefore be used for editing "jump" and "search" marks. The unit employs the EIAJ standard for PCM recording. And, in addition, editing and purely digital dubbing are easily accomplished with any videotape deck employing the NTSC format.

Technics new SV-P100 is available at selected audio dealers. To say that it must be heard to be appreciated is an incredible understatement.

Technics
The science of sound.

CIRCLE 92 ON READER SERVICE CARD

two track and although my letter didn't come back, they didn't write back either. Sometimes when I push "play," the pinch roller sticks and doesn't come against the capstan until I give the pinch roller a nudge with my index finger (and until I do, the reels spin wildly forward, as if the machine was in the fast forward mode).

—Phil Cohen
Bay Harbor, Fla.

The address of Otari Corporation is 2 Davis Drive, Belmont, California 94002. Their phone number is (415) 592-8311.

As far as your ideas concerning 8-track articles, we will see what we can do. If anyone out there has any ideas or information they'd like to share about 8-track techniques please send them to us.

What's Red and White, Sung, Sings, and Gathers No Moss?

Is it true that the Rolling Stones got their name from a blues tune by Muddy Waters called "Rollin' Stone"? What's the background on this, eh?

—George Adam
Hartford, CT

We wonder whether the Stones tour has stirred up this interest. It certainly has stirred up a lot already. Muddy Waters recorded a blues tune in 1954 called "Rollin' Stone," which, in addition to inspiring the name of the group, also inspired Bob Dylan to write "Like a Rolling Stone," and provided the name for Rolling Stone magazine.

Muddy Waters himself was born in Rolling Fork, Mississippi in 1915. He recorded most of his tunes in the 50s and 60s. He plays on the Chess label. In 1977 guitarist Johnny Winter and harpist James Cotton formed a blues triumvirate with Waters. Together they made an album called Hard Again.

More Tom Robinson

One very critical error in your Letters to the Editor, February 1982! Tom Robinson is still very active, and has a most interesting and provocative album that was recorded in spring of 1980.

Titled—*Tom Robinson/Sector 27*, produced by Steve Lillywhite, who by the way produces X.T.C., U2, and quite a few other new wave English bands. Released on I.R.S. Records, the 11 cut endeavor shows an excellent blend of moderate new wave and enjoyable listening, foot moving, rock.

—Doug Hilliard
Mt. Vernon, IL

Tracking Down Tubes

In the June 1981 issue of *Modern Recording & Music*, Brian Roth mentioned three brands of European tubes—Amperex, Telefunken, and Mullard. Do you have any idea where I may obtain these tubes in the United States? Thank you.

—Michael R. Kristofic
Gibsonia, PA

Brian Roth recommended looking in esoteric hi end hi-fi shops, or hunting through the classified ads of various audio magazines. He also recommended writing to or calling Audio Dimensions at 8888 Claremont Mesa Blvd., San Diego, Ca. 92123. Their phone number is 714-278-3310. Another store which carries Amperex, Telefunken and Mullard among other European brands (Hungarian, Russian) is Tube God. (They publish The Tube Bible. *They are located at 33 N. Riverside Avenue, Croton-on-Hudson, New York, 10520, and their phone number is 914-271-5141.)*

"The EX-18 could well become a classic audio tool"

June 1981 Modern Recording and Music © 1981 Cowan Publishing.

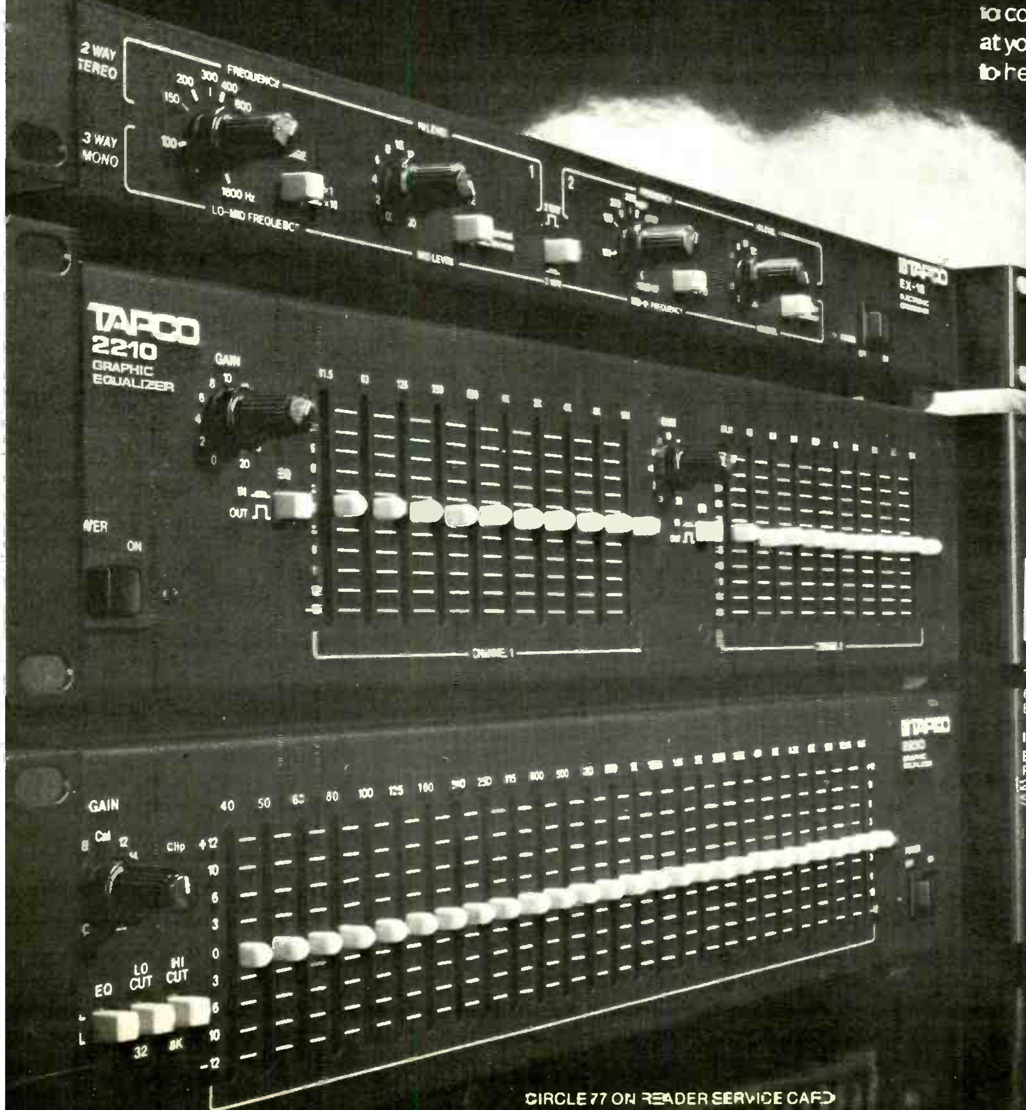
That's what Modern Recording said about the EX-18 stereo 2-way/mono 3-way electronic crossover. The same statement could very well apply to the new TAPCO 2210 and 2230 graphic equalizers as well.

The EX-18 provides all the necessary controls and functions for bi-amplifying stereo or tri-amplifying monaural speaker systems, and this can be accomplished using a unique mode switch so no external patching is required. A single knob on each channel adjusts the crossover frequencies, with a 10X multiplier available for very high frequency crossover operation. It is definitely one of the cleanest and quietest electronic crossovers available.

The same precision design and human engineering found in the EX-18 is found in the one-third octave 2230 and the dual ten-band 2210 graphic equalizers. Both are magnificent performers in recording and sound reinforcement applications. Whether you need the precision of the 2230 with its combining filter action, switchable high and low-pass filters and floating balanced outputs, or the economy and flexibility of the 2210, there are simply no better values in today's marketplace.

All three units are equipped with removable security covers to prevent accidental operation of any of the controls once your requirements have been set.

There is no need to settle for less than the best sound available. Especially when these E-V/TAPCO signal processing units give you professional sound quality for less than you'd expect professional quality to cost. These units must be auditioned at your E-V/TAPCO dealer. It's the only way to hear how good your sound can be.



EV TAPCO
a gulton company

TAPCO
600 Cecil Street
Buchanan, MI 49107

In Europe:
EVSA
Romerstrasse 3
2560 Nidau
Switzerland

In Canada:
Gulton Industries, Ltd.
Box 520, 345 Herbert Street
Gananoque, Ontario K7G 2V1

In Australia:
Electro-Voice Australia PTY-LTD
59 Waratah Street
Kirrawee, N.S.W. 2232

In Japan:
Electro-Voice, Ltd.
6F No. 2 Taro Building
2-10 Yotsuya, Shinjuku-ku
Tokyo, Japan 160

CIRCLE 77 ON READER SERVICE CARD

A Wild February

Thank you! The February issue of *MR&M* was worth what I paid for the entire year's subscription! I especially enjoyed the articles "Mic Maintenance" and "Recording Techniques—Part I." They were both clearly written and informative. I realize there must be card carrying audiophiles among your readers. Thanks for giving us beginners a chance to catch up! I look forward to the rest of the Recording Techniques series. How about some of what goes into the audio for radio stations?

—Regina R. Shulman
Teaneck, N.J.

We can't promise you that we will be able to come up with anything on audio for radio stations—it's not really part of MR&M*'s usual areas of concern. But we accept the compliments about February and hope Recording Techniques keeps on pleasing you.*

The Echo of an Interesting Chamber

Could you explain the differences between the different types of reverberation chambers for me? I'm going to be building a studio in a few months, but could use a little advice from people "who know." I figured *MR&M* would be the ones to ask!

—Stan Richards
Jersey City, NJ

There are four main types of reverberation chambers: "live," plate, spring, and electronic, or digital. A "live" chamber has a speaker at one end and a microphone at the other. The "send" signal gets played through the speaker, and after it bounces around the room, gets picked up at the other end of the room by the microphone. The "reverberant" sound is fed back into the console's "echo return" buss.

Plate echo uses a large metal plate (4'x8') which vibrates according to the signal which is fed into it.

In spring reverbs the signal is sent along a coiled spring which is suspended between two posts.

The last, digital reverb, synthesizes reverberation by using a microprocessor and does not use mechanical or acoustical means.

The most popular reverberation device is the EMT plate. ECOPLATE manufactures another plate-type device. There is a wide variety in spring reverberation devices, but digital reverb devices are a bit harder to find. You have to make "live" reverb chambers yourself.

Wrong Facts

In Part I of Bruce Bartlett's "Recording Techniques" series (February 1982 issue, pages 34-38), we inadvertently supplied you with some inaccurate information. To enlarge Mr. Bartlett's point concerning cardioid dynamic microphones that have a presence peak (page 36) we supplied a short list of examples. Therein lies the rub! The Audio Technica 811 and 812 models are condenser microphones, not cardioid dynamic; and the Electro-Voice DS-35 is a flat response mic that does not include a "presence peak" in the frequency response. Sorry for any confusion that may have resulted!

Fostex

Just so as to avoid any confusion that might have been sparked in readers' minds by an item in Talkback in the February 1982 issue (see "Multi-Track Mixdown"), we'd like to reiterate that the Fostex A-2 runs at both 7½ and 15 ips. This fact was not made clear in the item. Sorry.

Truly superior. We'll prove it!

The Orban 424A Gated Compressor/ Limiter/De-Esser

There are lots of production limiters out there. Old favorites. Pretenders to the throne. The competition is fierce. So, when Orban set out to design a new production limiter, we knew it had better be superior.

The result of our research is the "Studio Optimod"—a Gated Compressor/Limiter/De-Esser with versatile controls, simple set-up, and a natural, transparent sound that must be heard to be appreciated.

Try one and A/B it against your current favorite. You'll notice the **sound**—remarkably smooth and natural over a wide range of control settings—even at high compression ratios where apparent loudness and punch are significantly enhanced. It's no accident: The unit is a direct descendent of our super-popular, second-generation OPTIMOD-FM broadcast limiter. So it exploits our years of experience in making an AGC device sound natural on diverse program material without critical re-adjustments. Yet full versatility exists for special effects in production.

A bonus is a smooth, natural de-esser. It's independent of the compressor/limiter section so you can simultaneously compress and de-ess vocal material without compromise. You can even de-ess sibilant vocals which have been mixed with other program.

The icing on the cake is unique gating and "idle gain" functions which prevent unnatural noise-producing gain variations during pauses and abrupt gain changes when the unit is switched in.

Our new Model 424A (dual channel) and 422A (single channel) are destined to become the new industry standards in dynamic range control. Prove it to yourself. Contact your Orban dealer today.

orban

Orban Associates Inc.
645 Bryant Street
San Francisco, CA 94107
(415) 957-1067
Telex: 17-1480

CIRCLE 100 ON READER SERVICE CARD

TALK BACK

"Talkback" questions are answered by professional engineers, many of whose names you have probably seen listed on the credits of major pop albums. Their techniques are their own and might very well differ from another's. Thus, an answer in "Talkback" is certainly not necessarily the last word.

We welcome all questions on the subject of recording, although the large volume of questions received precludes our being able to answer them all. If you feel that we are skirting any issues, fire a letter off to the editor right away. "Talkback" is the Modern Recording & Music *reader's technical forum.*

Transformer Talk

My question has to do with the need for and the use of microphone step-up transformers. I operate a mobile recording service and work with a Teac A6100-Mk II, a Teac A2340SX, a Teac Model 2 mixer and an assortment of low-impedance, balanced microphones.

People—supposedly in the know—say that a transformer going into the mixer is a "must"; that they "negate any RFI problems, reduce hum and increase signal-to-noise." For the most part I have been running long mic leads with 2 conductor shielded cable, terminated at the mixer end by (assuming No. 2 pin to be hot) hooking No. 2 to tip and tying 1 and 3 together to ground. I have plugged every piece of cord that I have together (a total of about 700 feet) with no hum, RFI or high-frequency loss that I can tell by ear, without the aid of test instruments. (This was done on a test basis.)

On the other hand, I assembled a transformer setup with the transformer out of an Electro-Voice 671 mic, housed in a metal box, wired after the schematics of the Electro-Voice 502 series transformers. True, the signal was increased; however, the residue hiss of the long mic cords was also increased and hum was introduced that never existed without the transformer.

Also, I cannot understand why, if (in the old tubes vs. solid state controversy) the lack of a transformer in the output of an amplifier improves frequency response and transient characteristics, why the recording industry uses transformers for microphones. Wouldn't the same discrepancies of response be present there, as well?

The only advantage I have found of the transformer (assuming that a person has enough signal to drive his preamp, which has never been a problem for me) was when I had a long length of mic cable and the switch on the mic was turned off so that the transformer cut out the RFI (although that was not a problem so long as the mic was left on).

Does a microphone transformer have advantages that I am not aware of to warrant the extra cost? I'm operating on a limited budget.

—Paul F. Becker
ASP/Accurate Sound Productions
Bismarck, N.D.

Dealing with your last question first, I think it would be helpful to understand just what a balanced input really is and how it works.

Figure 1 shows an ideal balanced input. It is essentially two inputs with identical characteristics of impedance, frequency response and gain, differing only in that they are opposite in polarity (positive and negative). When these opposite polarity inputs are summed or added the result is equal to the algebraic *difference (a — b)* between the signals present at the two inputs. Note that if *a* and *b* are *equal* the result is zero.

Figure 2 shows a real world situation where a typical balanced output microphone feeds a standard 2-conductor shielded microphone cable which terminates in our balanced input of *Figure 1*. The jagged arrow crossing the microphone cable represents an interference signal. This could be radio frequency (RFI) or 60 Hz hum from a nearby power transformer or light dimmer, for example. This interference signal is induced onto both conductors *equally*, due to their close proximity, creating identical signals at both the positive and negative inputs, also known as a *common mode* signal. The *difference* between the two inputs is *zero* and the result is *no signal.* Therefore, what the balanced input is said to be doing is *rejecting common mode* signals.

Figure 3 shows the same circuit with the microphone generating an audio signal. Note that the audio signal is *not* common mode; that while one side goes positive the other goes negative in polarity. This *push/pull* configuration is typical of all balanced audio signals and is what enables a balanced input to discriminate between wanted (audio) and unwanted (interference) signals. The balanced input "sees" the *difference* be-

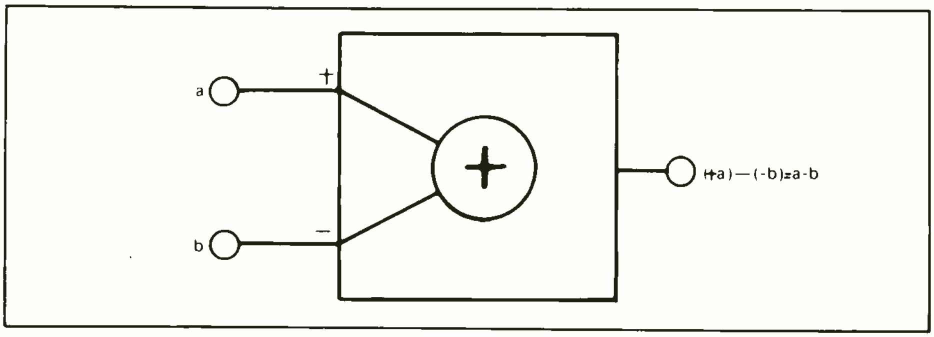


Fig. 1

tween the two inputs and the result is audio signal.

As seen in *Figure 2*, our ideal balanced input has infinite *common mode rejection*, or CMR. A single-ended input, on the other hand, has no CMR whatsoever, and will pass audio and interference signals induced on its single conductor equally.

When considering the very low voltage level of a typical microphone output and the fact that a typical microphone cable length of several hundred feet can, in severe circumstances, carry an interference-induced voltage level far exceeding that of the audio itself, it is easy to see the obvious need for a balanced input with its very high CMR.

(Hang in there, I'm going to answer your questions soon!)

Now what we need is a real world device to replace our ideal balanced input. There are several choices but it just so happens that a transformer is the easiest and least expensive way to go and has a number of other advantages. Transformers, inherent to their basic design, can easily be made to have very high CMR, in excess of 120 dB! They provide DC isolation, making phantom powering a simple matter and protecting active circuitry in the input amplifier. Very high quality input transformers are readily available at reasonable cost ($20-30) and are easy to implement in existing situations such as yours. As if all that were not enough, that same input transformer can also achieve impedance and level matching. *Figure 4* shows a simple balanced microphone input circuit with phantom power. The transformer shown is available from either Deane Jensen (#JE-115K-E) or Sescom (#MI-85) and is a high quality unit which provides an optimum load impedance to the microphone as well as providing 20 dB of voltage gain.

Finally, as I promised above, here are direct replies to your statements and questions:

1. A 700-foot run with unbalanced termination at microphone level is absolutely unthinkable and you were either conducting your test in the one interference-free location on earth (slim chance) or you didn't notice what might have been a subtle level of noise. Try loud headphones next time.

2. The impedance adapting transformer in the E-V microphone you mentioned is not designed to operate as an input transformer. The hum increase you noticed was due to improper connection of grounds and/or shields. Micro-

phone cables do not have residual hiss. The added noise was probably created in your mixer by an impedance mismatch due to the improper transformer type and wiring.

3. Input transformers are not subject to some of the design limitations which restrict the quality of practical high power output transformers. So, while the same discrepancies of response are there, they have been reduced to very low levels. For instance, the two units I have recommended have much flatter response, lower distortion, and better transient response than any *microphone* I know of.

—Peter J. Yianilos
President
Artisan Recorders, Inc.
Pompano Beach, Fl.

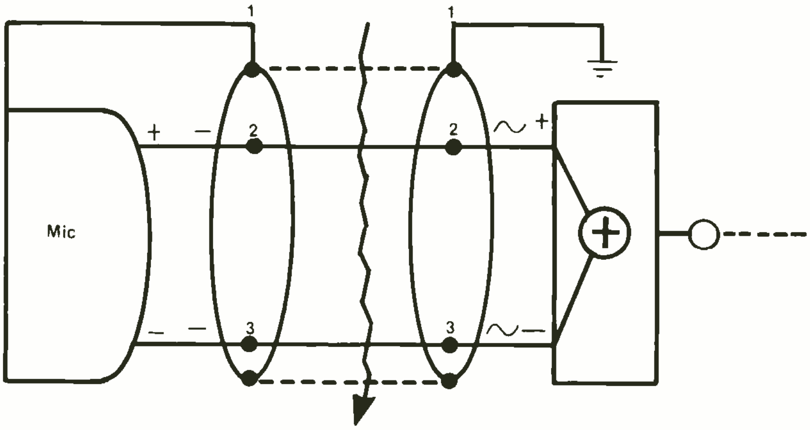


Fig. 2

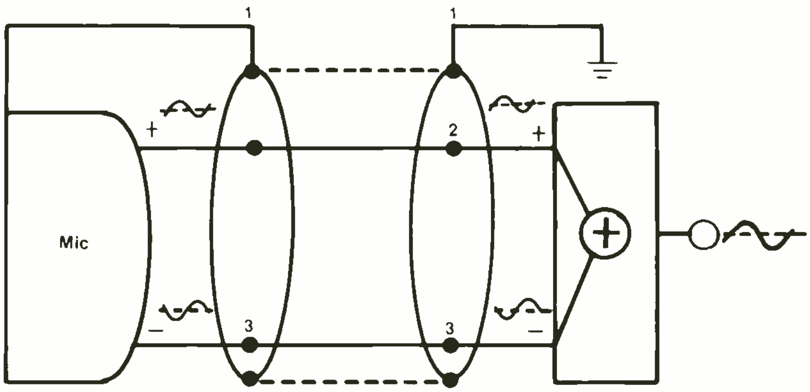


Fig. 3

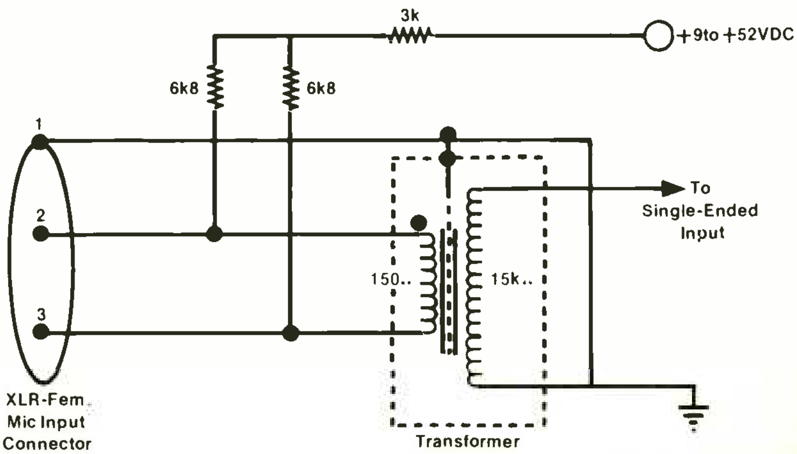


Fig. 4

YOU CAN HAVE A COLUMBIA RECORDS RECORDING CONTRACT

Columbia Records and the American Song Festival® have joined forces to offer talented performers an unprecedented opportunity to receive a fabulous contract to record and release an album.

Columbia Records is looking for new performing talent, and for original record concepts and ideas. **Columbia Records** releases and promotes *all kinds* of music: Pop, Rock, Jazz, Folk, Country, R & B, Soul, Classical, Gospel, etc. No matter what kind of music you perform, there may be a place for you on the **Columbia** label.

In its first 8 years, the **American Song Festival** has awarded over $700,000 in cash prizes to winners of its lyric and song writing competitions. Even more important, hundreds of winning songs have been recorded by major recording stars. Plus grand prize winning songwriters have recorded their own songs for the top labels, and have **#1 singles and #1 albums** to their credit. It could happen to you.

All you need is a simple home demo to display your performing talents. You define the type of music on your tape and the **American Song Festival** finds the right people in your style of music to do an honest appraisal of your artistic endeavor. The top pros in the music industry will listen to your work. Imagine being able to audition for a record company, a music publisher, an artist manager, and a record producer all at once. That's the kind of exposure you could receive.

If you want to further your career and record for **Columbia Records**, the first step on your path to success is to mail the coupon to:

American Song Festival
Recording Contract Search
5900 Wilshire Boulevard
Los Angeles, CA 90036

© 1982 American Song Festival

OR CALL TOLL FREE:
In the Continental United States (except California) **800-421-4212**
In California **800-252-2252**

Columbia is a trademark of CBS Inc.

Send to:
American Song Festival, Recording Contract Search
5900 Wilshire Boulevard, Los Angeles, California 90036

Please send me free details of the Columbia Records Recording Contract.

Name______________________________

Address______________________________

City______________ State____________ Zip________

A Presentation of Sterling Recreation Organization ©1982 American Song Festival MRRC

CIRCLE 114 ON READER SERVICE CARD

Ping Pong Tournament

I own a Teac 3440 4-track recorder and a Teac Model 2A mixer. Having learned the most basic methods of recording on this equipment, I am interested now in learning ways to record more than four tracks on a particular song. Can you give me information or refer me to sources which explain ping-ponging methods and other ways to do multi-track recording?

—Russell D. Carter
Decatur, Ga.

No one will ever believe that we didn't "plant" this question ourselves! The ultimate answer—we can hear all you readers who absolutely revelled in the article hooting out there even as we say it—is Craig Anderton's three-part "Multi-Track Magic: Creative Multi-Track Recording" series which appeared in the May, July and August 1981 issues. In the first part of the series, Craig offered six suggestions on how you can make your seemingly "limited" 4-track a veritable cornucopia of sounds. Techniques described in detail include combining three tracks into one, recording along with your premix, adding slap-back echo, adding "infinite" layering (this is precisely what you want to learn how to do, and no one can teach you better!), creating a "cheapo" echo unit (note, there's no additional equipment required; you get your effect by using only your 4-track) and adding ambience to premixes.

In Part II, Craig expands your horizons by showing what you can do simply by incorporating a standard 2-track mixdown machine into your ensemble. In this segment, you'll learn a cheap-y way to save wear and tear on your 4-track, how to make 4-track transfers to ¼-track, a method of hand-syncing (steady there, guys!), tape phasing (flanging), and how to use the second machine as an echo unit.

In Part III, Craig—feeling that he'd taught you just about all there is to know about getting the most out of a 4-track from an engineering standpoint—changes hats and shows you how to "produce" those tracks into something you'll be able to show proudly to even the pickiest professionals. From some basic strategies for approaching a session, right on through to such areas as sweetening tracks, adding effects during mixdown as opposed to adding them during recording and noise reduction, Craig truly instructs you from soup to nuts.

All three issues are still available from our Back Issues Dept., but keep in mind that this series appears to be one of the most popular and useful that we've ever printed, so don't hesitate.

Another Anderton-authored piece—this a soft-cover book—which might prove helpful is Home Recording for Musicians. *Again, his aim is to help you create professional quality material.* HRFM *will also provide information on equipment as well as technique, but it is worth your time. (It can be ordered from Guitar Player Books, P.O. Box 615, Saratoga, California 95070 or Music Sales Corporation, 33 W. 60th St., New York, NY 10023.)*

Patching In

I am interested in any help or plans you might be able to provide for constructing a small patch bay. It seems that many of the models available have too many patch points for my limited needs.

—Doug Horner
Sheperdstown, W. Va.

As the sage once wrote, everything is relative in this world, so, when you write us that many patch bays marketed today have "too many patch points for my limited needs," we must ask: How many are too *many? What are your limited needs? The one thing that sticks in our mind is that with any patch bay one does have the option of using as many or as few of the available patch points as his recording situation dictates.*

While we do have a rather extensive library of audio-related materials here, we didn't find a plethora of plans for patch bays, unfortunately. However, we did ascertain that Audio Amateur *did offer such a project in one of their issues a few years back, as well as a couple of columns and assorted letters on the various problems associated with these units. Write to* Audio Amateur, *P.O. Box 494, Peterborough, New Hampshire 03458. (These are the same people that publish* Speaker Builder Magazine, *which is something perhaps you'd like to fild away for later use.)*

You might also get in touch with Stereo Sentry Manufacturing Company which manufactures the "Audiopatch" line of patch panels. They came to our attention billboarded as made "to fit every need." Write them at 5420 Blodgett Ave., Downers Grove, Illinois 60515 or call 312-495-3535. Hope this does it for you!

Obsession with Excellence...

"The PL80 is the *only* mike I'll use!"

The consistently high quality of Crystal's standing-room-only concerts is a demonstration of her obsession with excellence and it says a lot about the equipment she chooses to use.

The PL80 was designed for entertainers. Entirely new applications of available computer-assisted technology were used to ensure that the PL80 performed exactly as it was originally designed to. Extensive field testing with groups like Journey and stars like Crystal Gayle were used to check the designs every step of the way. That is Electro-Voice's obsession with excellence.

The result is an entertainer's microphone that sounds exactly like you want it to. Its crisp high end and modest bass boost enhance a performer's voice without compromising the performer's vocal quality. And it sounds that way *on stage*, not just in a sterile test environment.

"I've used a lot of other mikes in my career and without doubt the PL80 gives my voice a truer sound than I could ever get using any other microphone." That is Crystal Gayle's obsession with excellence, and her statement says it all! Give your voice the sound it deserves. Get your voice a PL80 at your Electro-Voice PL microphone dealer!

Crystal Gayle insists on using the Electro-Voice PL80 microphone.

CIRCLE 82 ON READER SERVICE CARD

EV Electro-Voice®
a gulton company

600 Cecil Street, Buchanan, Michigan 49107
In Canada
Electro-Voice, Div. of Gulton Industries (Canada) Ltd., 345 Herbert St., Gananoque, Ontario K7G 2V1.

THE PRODUCT SCENE

By Norman Eisenberg

BASF UPDATES: ISSUES TEST TAPES

Continuing its advocacy of chrome tape, BASF has announced improvements in tape formulations, cassette mechanism and packaging design. The new Professional II series uses pure chromium dioxide and provides "better bias compatability with modern recorders..." says BASF. It also has better MOL values (−1.5 dB at 315 Hz); a low noise floor said to be unmatched by any other tape; flatter frequency response; and a more durable polyester backing. The Professional I series (IEC Type I for normal bias) has an improved binder system for greater durability and batch-to-batch consistency; a maghemite particle providing greater packing density; and lower distortion. All BASF tapes are housed in the same precision shell specified by Mobile Fidelity for its audiophile cassettes. This housing uses an asymmetric base/cover design for consistent azimuth tracking from side 1 to side 2. Other new features are sturdier hubs with spoke design; large pressure pad with a tight fiber nap; black screws for uniform identification; new labels and index cards.

BASF also has released two new cassette test tapes—one ferric oxide, the other chrome. Both conform to the new IEC calibration standard jointly developed by BASF and TEAC. Each cassette contains a reference level, and azimuth alignment and frequency response sections. The cassettes extend to 18 kHz and are said to be the only ones in the world that do so.

CIRCLE 8 ON READER SERVICE CARD

PCM CASSETTE DECK FROM JVC

Claimed to be the world's first is the new PCM cassette deck announed by JVC. Capable of one-hour PCM digital recording or playback, the system uses the high density (46.3 k BPI) recording technique. The format, explains JVC, is also designed to meet the mass-production requirements of prerecorded tapes. Not compatible with conventional analog decks and tapes, the new JVC system aims at providing the full advantages of digital sound for a one-hour stereo program combined with the familiar conveniences of the cassette medium. Principal specs announced so far include tape speed of 7.1 cm/sec; two heads (erase and combined R/P); sampling frequency of 33.6 kHz; quantization equivalent to 14 bits; bi-parity error correction system. The two-way stereo recording function includes service tracks for random access, program indication and other functions. No price has been announced yet.

CIRCLE 9 ON READER SERVICE CARD

SCHOEPS MICROPHONES LISTED

The extensive line of Schoeps microphones and accessories made in West Germany and distributed in the U.S. by Posthorn Recordings, NYC, is described in literature available from Posthorn. Studio condenser mics in the large Colette Series include units with different pickup patterns, powering methods and recording capabilities.

CIRCLE 10 ON READER SERVICE CARD

RTS OFFERS LOW POWERED AMP

To meet industry needs for a small, self-contained, reliable low-power amplifier, RTS has announced its model 410. Occupying a single rack-unit height (1¾ inches) and one-half the rack width (8 3/8 inches), the model 410 is spec'd for an "honest" 10 watts into an 8-ohm speaker. Designed with a balanced bridging input, the amp is spec'd for 0.03 percent THD at rated output, with a 90-dB signal-to-noise ratio. Slew rate is 6 volts per microsecond. Input is via a 3-pin XLR-type connector or standard ¼-inch phone jack. All electrical parts are assembled on one circuit card and all components and connections are easily accessible. The amp incorporates integral overload protection and is said to remain stable and safe when handling a highly inductive or capacitive load, or even a short circuit. Price is $288.

CIRCLE 4 ON READER SERVICE CARD

STUDER DIGITAL ENTERS SCENE

From Studer Revox comes word of its digital sampling frequency converter (SFC-16) for professional digital audio. The new system makes it possible to transfer digital audio programs between recorders and other systems with conflicting sampling frequencies. The SFC-16 accepts arbitrary sampling frequencies and, since it operates strictly under control of the "clock" signals, it does not require any programming. A two-channel version, shown last fall at the New York A.E.S. convention, fits standard rack-mounts and is said to offer applications that range from format conversion to mastering for the digital audio disc, as well as the digital transmission of audio programs. Company head Willi Studer coincidentally has called for worldwide standards for digital audio and is readying—for display at the next A.E.S. in Montreaux—a prototype of a Studer multi-channel PCM tape recorder.

CIRCLE 5 ON READER SERVICE CARD

H-K DEBUTS EQUALIZER

The EQ7 is a new ten-band stereo graphic equalizer from Harman-Kardon. The device provides ±12 dB range for each channel, over ten octaves, via separate sliders for left and right channels. Also featured are an extra tape-monitor loop, input level controls for each channel, overload indicators and EQ defeat. Overload level is spec'd as 7 volts, and S/N is listed as 105 dB. Price is $250.

CIRCLE 6 ON READER SERVICE CARD

LEXICON PROCESSORS

Lexicon has announced two signal processors for pro applications. One is the 224-X digital reverb unit, successor to the earlier 224. Original features are retained in the new version which adds to them the following: Full 15-kHz bandwidth; variable bandwidth from 15 kHz down to 170 Hz; dynamic delay whereby the reverb time depends on program level;

paging system which allows the six sliders on the remote panel to be redefined to control additional features while retaining the familiar pattern for the most-used reverb adjustments; an updated 8-voice chorus program for additional control over delays and density; a non-volatile register storage and extended ROM storage standard so that thirty-six user-defined pre-sets may be held in non-volatile register stoppage, with the ROM space providing for at least thirty-two basic program algorithms.

The Lexicon 1200B, successor to the model 1200, expands the frequency capability to a 15-kHz bandwidth to meet broadcast network standards. Like the earlier 1200, the new version is known as an Audio Time Compressor/Expander. Its purpose is to change the speed of taped material to fit critical time slots while maintaining broadcast-quality sound.

CIRCLE 7 ON READER SERVICE CARD

NEW HEATH CATALOG

Evidence of rising interest in digital technology, computers and energy-saving is seen in the current catalog from Heathkit, the world's largest manufacturer of do-it-yourself electronics and related mechanical devices. In addition to digital FM, the catalog shows a digital multimeter and a 20 MHz oscilloscope, as well as desk-top computer systems and associated hardware and software products including a printer, and self-instruction course; and various weather data and energy-control devices. The 104-page catalog also contains products covering the widest range of electronic interest. It is free on request to Heath Company, Dept. 350-435, Benton Harbor, MI 49022 (in Canada, Heath Company, 1480 Dundas St. East, Mississauga, Ontario L4X 2R7).

CIRCLE 17 ON READER SERVICE CARD

NEW ONKYO POWER AMP

Onkyo's new model M-5090 is a stereo power amp rated for minimum RMS output of 200 watts per channel, both channels driven, from 20 Hz to 20 kHz, at no more than 0.01 percent total harmonic distortion. The amp features Onkyo's "Dual super servo circuitry" which eliminates DC and extra-low frequency distortion in signal and ground lines. In the M-5090 this technique is said to produce a clean signal that is equivalent to a 100-times larger power supply. The amp also uses linear switching circuitry, designed to eliminate the distortion inherent in conventional class-B amplifiers. In effect, says Onkyo, the new amp achieves class-A performance with class-B efficiency. Featured are large dual-range peak power meters. The M-5090 is designed and may be used as two independent mono amplifiers, down to the two independent power supplies. Price is $1800.

CIRCLE 11 ON READER SERVICE CARD

MICMIX NOISE REDUCTION

Micmix, makers of the Master-Room reverberation systems, now has available a new single-ended noise reduction unit. The Dynafex model D-2B, unlike most of the compander-type systems on the market, is capable, says the manufacturer, of reducing noise present on the original material. A two-channel system, the D-2B is said to reduce noise by up to 30 dB or more, and can be adjusted for reference levels of +8, +4, 0 or −10 dB—which permits interfacing with broadcast, recording and consumer equipment. The Dynafex D-2B is available with 600-ohm transformers (along with a 115/230-volt option), and can be used in the recording studio as a noise gate/noise reduction system, as well as in a mixdown situation prior to the mastering tape machine. It may also be used in sound reinforcement work on a per channel basis, or in the master mix just prior to the power amplifiers to "greatly reduce noise build-up."

CIRCLE 12 ON READER SERVICE CARD

DIGITAL PERIPHERALS FROM TECHNICS

To provide digital recording with the "freedom and flexibility" found in modern multi-track analog recording, the Technics brand of Matsushita has introduced two signal-processing peripherals in digital form. One is a new version of a digital audio mixer, which includes equalization facilities. An 8-in, 2-out console, this system handles signals in digital form and it provides panning and level adjustment plus facilities for special effects control via its AUX terminals. The built-in digital EQ permits the recording engineer to manipulate the frequency spectrum without degrading the basic signal quality through the addition of distortion or noise associated with analog EQ circuitry.

The other new device is a digital audio reverberator with memory. It simulates six successive reflections, and provides control over initial and late reflections. Featured are extensive memory, preset hall simulations, and specific delay time control of frequency bands.

CIRCLE 13 ON READER SERVICE CARD

POWERED MIXER FROM BIAMP

A mixing console, power amplifier, equalizer and reverb unit are combined in a lightweight portable package in the new 29 series from Biamp Systems. Available in 8- or 12-channel versions, the new units have identical 9-band graphic EQ and internal convection-cooled power amps. The equalizer, which may be programmed or patched for independent operation, uses octave-wide filters centered on frequencies from 50 Hz to 12.5 kHz. The amp is designed to drive 2-ohm reactive loads and is rated for 300 watts output per channel at 2 ohms with both channels driven. Input jacks permit using the amps independently. Front panel LED ladders are calibrated to the outputs in percent power, automatically compensating for different load impedances. A stereo headphone jack for each amplifier allows the operator to listen to both main and monitor signals. Inputs for each channel include a female XLR connector and a phone jack. The jacks are factory programmed for different functions: four channels for channel patch; six channels for line inputs; two channels (in the 1229) for instrument input. The rear panel has four phone jacks for interfacing with a tape deck, and a footswitch for external control of the internal reverb.

CIRCLE 14 ON READER SERVICE CARD

LORAN INTROS NEW TAPE

Offered as a premium-quality high-bias type II tape is Loran's LHB. Using Loran's "Lexan" cassette housing, the new tape is available in 60 and 90 minute sizes, priced respectively at $5.75 and $7.95. The tape uses 70-microsecond EQ. Specs include flat frequency response with a 3 dB rolloff at 21.5 kHz; sensitivity of −0.45 dB at 330 Hz; S/N ratio of −64.5 dB with Dolby; and high-frequency headroom of −3 dB at 10 kHz maximum output level.

CIRCLE 15 ON READER SERVICE CARD

WISH THEY'D MAKE...

With microprocessors increasingly showing up in cassette decks, turntables, tuners, equalizers and analyzers, one must marvel at the ingenuity of equipment designers. At the same time one wishes that some of this know-how were applied to devices that could make work easier or more accurate for the "working sound person."

For openers, consider a compact gadget that gives you a readout of speaker impedance from a 20 Hz to 20 kHz. You feed the output of an audio generator into a speaker, patch in your computerized Z-reader, set its controls for nominal impedance and read the results on LEDs. Essentially, all the microprocessor would have to do is compare the voltage drop across the speaker with that across a series resistor of nominal speaker impedance and translate that data into a number of ohms for each tested frequency.

What about a microprocessor for turntables that monitors extra-heavily cut groove portions, or surface flaws that could cause the stylus to stop and cause repeats? In this system the microprocessor would automatically vary the vertical tracking force of the stylus and possibly too the anti-skating force to compensate (up or down, as the case may be) and then it would restore the original values once the obstacle has been passed.

More modest would be a device that warns you when your tape has, say, one minute of time left, just in case your attention happens to be occupied away from the deck at that moment. A warning buzz or bell could alert you, not unlike the little bell you hear near the end of a line of typewriting.

More ambitious would be a temperature and humidity compensator for speaker systems that would adjust for optimum response with respect to a predetermined, or even a variable, set of atmospheric conditions. For that matter, why not extend this concept to include compensation for the number of persons present, and possibly too for the texture of the clothes they are wearing? It can be awfully frustrating to "voice" a room that is empty, only to discover that the frequency balance changes when the room is filled with an audience, or even when the heating or air-conditioning is turned on, or when the weather changes.

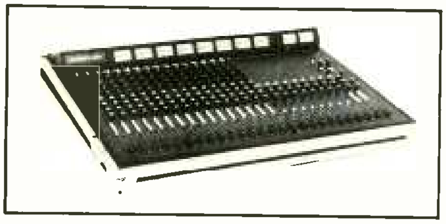
I don't know how much market research or drawing-board effort these suggestions will stimulate, but I have a feeling a lot of audio people would welcome devices such as these. Any other ideas from anyone out there?

NEWSICALS

The most recent NAMM (National Association of Music Merchants) convention was held this past February 1982 in Anaheim, California. As usual there were many, many new products for the musician, recordist and soundmixer to pant over. The following products were showcased at the convention, and there will be more updating on the other new offerings in future issues.

MIXING BOARDS

European Audio Distributors are the exclusive U.S. distributors for the new Audiotrack 16:8:2 mixer. As the model designation implies the unit has sixteen input channels routing to a stereo main



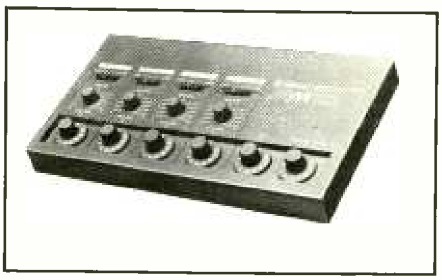
output through eight subgroups which also double as an 8-track monitoring section for recording applications. The input channels feature active balanced mic inputs, three-band EQ, three auxiliary sends, two pre-fader and one post-fader, for foldback, effects and/or monitor use. The mixer is available in two forms: built into a heavy-duty road case for portable use or with walnut side rails for fixed installations.

CIRCLE 18 ON READER SERVICE CARD

Biamp Systems has introduced two new mixer models, one an unpowered rack-mount unit and the other a powered unit boasting over 220 watts of power into a 4-ohm load (300 watts into 2 ohms). The 683 is the latest addition to the company's 83 series, and is a full-feature, six-input, stereo/mono, rack-mount mixer. Each input channel features an active balanced mic input with gain trim, three band EQ, effects send, monitor send, sub1/sub2 pan and main level. The two submixes may be used as stereo outputs or combined through the main output channel. Two LED meters are provided which are switchable sub1/main and sub2/monitor. The Model 619 is a powered, rack-mount mixer incorporating many of the design features of Biamp's 29 series of powered mixers. The 619 features balanced low-Z mic inputs, unbalanced high-Z mic/line inputs, input pad, two frequency EQ, monitor bus, an effects bus feeding an internal spring reverb, auxiliary inputs, RIAA magnetic phono preamp inputs, tape recorder inputs and outputs and a nine-band graphic equalizer on the main output. The power amp section features autolimit circuitry with indicator light to yield higher levels of loudness without clipping or amplifier overload.

CIRCLE 19 ON READER SERVICE CARD

As part of its Producer Series of miniaturized sound system components, Yamaha Combo Products division has introduced the MM10 mic/line mixer. The MM10 is a 17 ounce, four-input stereo mixer measuring only 8 3/4



x 5 x 2 1/16 inches. The unit is powered by 9-volt batteries or an AC adaptor for true portability and freedom from AC hum. Each of the four input channels has a mic/line switch, a panpot and a level control, and an auxiliary input and a master level control are also provided. Also in Yamaha's Producer Series is the MA10 headphone amplifier which is physically the same size as the MM10 mixer, is also battery powered and will drive two pairs of stereo headphones.

CIRCLE 20 ON READER SERVICE CARD

POWER AMPLIFIERS

Ashly Audio has been best known for its signal processors—professional limiters and equalizers and low-level active crossovers—but has branched out into the power amplifier market with the introduction of the FET-200 amplifier *[see this issue's "Lab Report" section].* The new Ashly power amp is not just another "me too" power amp, as it was designed from the ground up to take advantage of the desirable characteristics of the MOS-FET (metal-oxide semiconductor, field-effect transistor) output devices it uses. Besides having certain operational and sonic similarities to vacuum tubes, MOS-FETs are immune to an effect known as thermal runaway, which is the great destroyer of conventional solid-state amplifiers and which designers must spend considerable effort and expense designing around. The FET-200 is a two-channel amp which may be operated in the bridged mono mode for high power applications with only the flip of a switch. Each channel has a pair of ¼-inch phone jacks and a male and female XLR-type connector for ease in paralleling multiple amps; the input circuit is bridging, active balanced for use with balanced or unbalanced sources. Level controls are provided on the rear panel and a pair of 10-segment LED level displays facilitate level setting. The unit is housed in a 3½-inch package with steel chassis and aluminum panel, and is fan-cooled with a rear panel fan drawing air in through front panel slots. Output power for the unit is specified as 100 watts per channel into 8 ohms; 160 watts/channel into 4 ohms; 320 watts into an 8-ohm load in the bridged mono mode.

CIRCLE 21 ON READER SERVICE CARD

QSC Audio Products has introduced a new line of professional power amplifiers known as Series II. In designing these new amps, QSC emphasized improving the electrical efficiency of the amplifiers, particularly at low to medium power levels. To achieve high efficiency, which carries with it the advantage of much lower heat production, QSC came up with a unique design which uses one group of output transistors operating off a low voltage power supply to handle power needs at lower power levels while a second set of transistors connected to the full supply voltage remain biased off until moderate power levels are reached. Three models are available: the 2200 packages 135 watts/channel at 4 ohms in a 1¾-inch unit; the 2350, rated at 265 watts/channel at 4 ohms; and the 2500 puts out 400 watts/channel at 4 ohms. The 2350 and 2500 are 3½-inch rack mount packages which are also capable of driving 2-ohm loads for greater power outputs. All three have separate channels and power supplies so a failure of one channel can't affect the operation of the other.

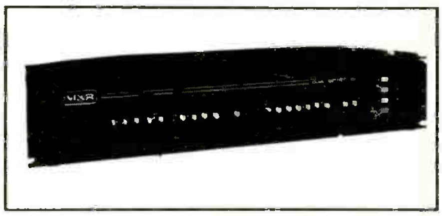
CIRCLE 22 ON READER SERVICE CARD

SIGNAL PROCESSORS AND EFFECTS UNITS

Audio Matrix manufactures a range of tube preamps and overdrive units plus a range of hybrid tube/transistor amp heads in rack-mount configurations. The Audio Matrix line includes the Mini Matrix, a foot-pedal-style tube-overdrive unit featuring a series preamp configuration to produce a true tube amp overdrive sound with any amplifier, tube or solid state, and the Tube Loop, a rack-mount, tube-overdrive preamp designed to be used in the external effects loop of conventional amps or as a separate tube preamp. Also featured are the Lead Rack and Duo Rack, both of which are available as preamp only or as powered units with 100 watts RMS onboard. The Lead Rack is a two channel preamp with an FET rhythm preamp channel and a tube lead channel noiselessly switched via J-FET circuitry; traditional four-knob EQ; a reverb unit; and an effects loop. The Duo Rack is an enhanced Lead Rack with the addition of a five-band graphic EQ to supplement the three-knob conventional EQ controls; the two EQ sections are normally used separately but may be cascaded in the tube mode for extreme tonal control. The Bass Rack is basically a Duo Rack with the circuitry optimized for bass guitar rather than optimized for a lead or rhythm guitar, and with the addition of a fully controllable, low distortion compressor circuit.

CIRCLE 23 ON READER SERVICE CARD

MXR Innovations has revamped its family of popular graphic equalizers. The first of the new models to be available is the Model 170 Dual Octave Equalizer, which will be followed this summer by a Dual Two-Thirds Octave to replace the current Dual 15-band Equalizer and a One-Third Octave unit to replace the current 31-Band Equalizer. All three units utilize minimum-phase, active combining filters for minimum distortion and



feature overall level controls and a switchable 18 dB/octave filter to roll off undesired low-frequency content. Inputs and outputs are via ¼-inch phone connectors, with the input being active balanced to interface with balanced or unbalanced sources, and a signal present LED indicator is provided to aid in set-up and troubleshooting. The new Model 170 features ten filter sections on ISO preferred octave frequencies from 31.5 Hz to 16 kHz, each section providing ±12 dB of control via center-detented slide controls. All the three new models also feature revised graphics with blue and white markings on a black panel, and they conform to EIA rack-mount dimensions.

CIRCLE 24 ON READER SERVICE CARD

A dual channel, stereo/mono compressor-limiter mounting in 1¾ inches of rack space has been introduced by Pulsar Labs. The new unit is called the CL2000, and utilizes several exclusive design features to provide a high level of performance. The unit uses a special gain-control element Pulsar calls a "transimpedance VDCA," the letters standing for Voltage & Duration Controlled Amplifier, indicating that the element is controlled by peak duration as well as voltage (amplitude). The CL2000 may be used as two independent limiters, or the two channels may be couplec stereo tracking with a high degree accuracy; their exclusive Digitr TM circuitry supposedly reduce stortion to ten times less than their n est competitor. Each channel of the it has a three-section, tensegme LED meter for simultaneous meteri of input and output levels and amou gain reduction. Threshold for each nnel is selected via a pair of pushb ons operating digitally for exact re tability, and controls are provided compression ratio (2:1 minimum 25:1 maximum), attack and release es and output level. The unit uses ecial circuit called Dynalease™ (pate pending) for dynamically-controll ttack and release characteristics eve the manual modes minimize pun ng and breathing; a fully aut tic mode is also provided. Other feat s include side-chain patching for fre ncy-dependent limiting or de-ess , and a limiter bypass mode.

CIRCLE 25 ON READER SERVICE CARD

SPEAKER SYSTEMS

Biamp Systems has announced two new speaker systems for P.A. applications. The 210T is a compact two-way column speaker suitable for small P.A. applications such as churches, schools, clubhouses, etc. The 210T is unlike many other column type systems in that it is a full-range system rather than being for voice range only. Frequencies from 80 Hz to the 5 kHz crossover point are handled by a pair of high efficiency 10-inch drivers, while the treble end of things from 5 kHz to over 20 kHz is reproduced by an array of four tweeters. The system is housed in a tolex-covered enclosure (36"h x 16"w x 12"d), weighs 64 pounds and is rated at 98 dB SPL at 1 meter for a 1 watt input.

Another new speaker from Biamp is the 1545T, a medium-sized, full-range, 3-way system suitable for keyboard amplification, stage monitoring or any short- to medium-throw P.A. application. The system comprises a woofer in a Thiele-Small aligned vented enclosure, an array of four midrange drivers and an array of six 2½-inch tweeters. The use of the particular midrange and treble drivers in the system is said to achieve an order of magnitude improvement in harmonic and intermodulation distortions and stored energy over conventional compression driver/horn combinations. The system is rated at 97 dB SPL output at 1 meter for a 1 watt input. The enclosure measures 32"h x 24"w x 16"d and weighs 91 pounds.

CIRCLE 26 ON READER SERVICE CARD

New from the folks at Bose and designed specifically for vocals and guitar are the 402 and 402-W Articulated Array™ Speakers; the two models are acoustically identical, but the 402 is intended for portable or outdoor use while the 402-W is covered with walnut-grain vinyl for permanent installation. Both models are compact column-type speakers using four 4½-inch Bose D-22A drivers on a uniquely angled baffle; the top and bottom drivers are angled outward (up and down) while the center two drivers are angled toward each other and have an acoustic diffractor to broaden the horizontal radiation pattern. The cabinets utilize Tuned Reactance Radiator (bass reflex) slots to improve bass response down to 90 Hz. The entire 402 enclosure and the complexly-shaped baffle of the 402-W are manufactured from polyethylene copolymer structural foam reinforced with mica for low weight, high rigidity and environmental resistance. The systems are rated at 8 ohms impedance, 120 watts continuous power handling and 98 dB SPL output at 1 meter from a 1 watt input (for midrange frequencies; the broadband efficiency is 93 dB). Both models are intended for use with the 402-E active equalizer before the power amp to achieve flat bass response. Size is 23" x 8 1/8" x 7¼".

CIRCLE 27 ON READER SERVICE CARD

SYNTHESIZER EQUIPMENT

New from Sequential Circuits is the Prophet-T8, a completely touch-sensitive, programmable, 8-voice polyphonic synthesizer. The six-octave keyboard of the T8 has both velocity and pressure sensitivity which can be used to control various synthesizer functions. The adjustable velocity sensitivity may be used to control any or all envelope parameters of the ADSR envelope generators, and for additional versatility these envelopes may be routed to other destinations, such as oscillator frequency or pulsewidth, via the Poly-Mod section. The keyboard pressure sensors are activated by additional pressure after a key is depressed—"second touch"—and can be used to control oscillator frequency, filter cutoff, LFO amount or various other parameters. The Prophet-T8, like the other Prophets, assigns the eight voices to the keyboard according to four keyboard modes: Normal, Double, Split and Unison/Track. Other features include polyphonic glide, a programmable initial amount of modulation for the mod wheel, two programmable, switchable release values for the ADSR generators and a new envelope mode called ADR (zero sustain) for more natural percussive envelope.

CIRCLE 28 ON READER SERVICE CARD

The Yamaha Combo Products division of Yamaha International introduced a new, high-performance micro-synthesizer known as the CS01. This highly portable, battery-operated monophonic synthesizer is the central

component of Yamaha's Producer Series, which also includes a mini-mixer and a headphone amp. The CS01 packs a 32-note keyboard and a lot of features into a package only 19 1/4 inches wide and 6 3/8 inches deep. Controls include pitch and modulation wheels, variable glissando, a keyboard range selector and full controls for VCO (pitch, waveform and pulse-width modulation), VCF, VCA and envelope generator. An unusual feature is an optional breath controller for modulation of VCF or VCA parameters with your breath.

CIRCLE 29 ON READER SERVICE CARD

New from Moog Music is a comprehensive microprocessor-controlled sequencer they call the Digital Sequential Controller, or DSC for short. The DSC is capable of storing up to four sequences of non-fixed length totalling approximately 800 notes. The unit has three distinct recording modes: a Normal mode in which all keyboard techniques are recorded exactly as played; a Metronomic mode in which timing is ignored and each event is given the same duration; and Expanded mode which stores and replays anything that can be played on the synthesizer including pitch bend, vibrato and glide. Other unusual functions include storage of a transpose chain of keyboard intervals to produce automatic key changes, the ability to add glide to any sequence, or to convert recorded glide to quantized glissando, and the programming of output pulses to trigger external devices. The DSC has all the typical functions for stopping and restarting a sequence, single-stepping, editing and correcting sequence and storing sequences on cassette for later retrieval, but in addition the DSC generates a SMPTE-compatible sync signal for ease in synchronizing the sequencer with tape machines or additional controllers.

CIRCLE 30 ON READER SERVICE CARD

MUSICAL INSTRUMENT AMPLIFIERS

Legend Amplifiers has announced several additions to its range of amps. Newly added to the bottom of the Legend line is the Model A 30, a 30-watt RMS combo amp with a single 12-inch Jensen special design speaker. The preamp section of the A 30 is a tube design (as are all Legend preamps) featuring two gain sections which may be selected by footswitch for instant changes between two different gain settings. The A 30 also features an external effects patching loop, a Hammond long spring reverb tank and four bands of equalization in the preamp. Legend also announced that all its various configurations of amp heads and combo amps will be available in two versions, the traditional Legend style Rock'n'Roll Series II products, and the new Legend Super Lead series. Both series feature the same hybrid design philosophy, using a tube preamp section for the warmth and distortion characteristics of a tube amp but with a solid-state power amp stage producing either 50 or 100 watts RMS of

clean, cool power. The basic difference between the two series is that the Super Lead amps were designed for maximum versatility and range of sound to make them more suitable for country and jazz applications than the Rock'n'Roll Series which duplicate the classic "British" tube amp sound. The Super Lead series uses an Active Bass Enhancement circuit which provides a parallel signal path for the frequency range from 15 Hz to 230 Hz; this secondary, parallel signal is derived before any overdrive stages and is combined with the main signal after the EQ section to allow up to 40 dB of distortion-free boost in the low-frequency range to produce the warmer "American" sound rather than the classic "British" character. Also added to the Super Lead series is a Hammond long spring reverb unit with footswitch.

CIRCLE 33 ON READER SERVICE CARD

GUITARS AND BASSES

New from St. Louis Music are three new models which compromise the Silver Maple Series of "all wood" dreadnaughts. All three models feature solid spruce tops with scalloped parabolic bracing, double reinforced mahogany necks with ebony fingerboards, genuine maple binding, and a distinctive grey, black and turquoise herringbone inlay around the soundhole. The DY52 Silver Lark features sides and back of solid walnut, has pearl snowflake inlays in the fingerboard and comes with a separate wood pickguard with adhesive backing for optional installation. On the DY53 Silver Harp, rare burled mahogany with a satin finish is used for back and sides plus a headpiece facing and separated pickguard of the same material. For the DY54 Silver Fawn, the material used for the back, sides, headpiece facing, and separate pickguard is satin-finished oboncol wood, and this model also features a special inlay of rosewood on the lower bout.

CIRCLE 36 ON READER SERVICE CARD

One of the most respected of the specialty guitar makers is Steve Hegelson, the proprietor of Moonstone Guitars. A highly distinctive feature of Moonstone's guitars is the use of American hardwood burls as a primary material or as a facing material. (Burls are knobby growths forming on the trunk of certain hardwood species such as bigleaf maple, myrtle and black walnut in response to certain peculiarities of soil chemistry and/or terrain. Burl is reknowned for the beauty of its very pronounced and highly irregular grain patterns and its high density, both of which characteristics make it a desirable wood for use in electric guitars.) Moonstone's top of the line model is the Vulcan, whose body is carved from a single piece of the finest burl. Coupled to this exotic body is a 5-piece laminated neck of hardrock maple and African padauk wood, and an electronic package that includes a preamped Bartolini ES1 humbucking pickup in the bridge position and a Bartolini Beast II with a 5-position voice selector switch in the rhythm position. For those who prefer a neck-through-body design, Moonstone makes the Eclipse, which combines a laminated maple neck with a body constructed of bookmatched burl facings over a mahogany core. The Eclipse is available in Standard and Deluxe versions as well as a bass guitar. An unusual offering is the M-80, a semi-hollow design whose top and back are carved from one-inch-thick blocks of solid or bookmatched maple. Other offerings from Moonstone include the Eagle, a burl-bodied guitar intricately carved in an eagle motif (complete with 126 detailed feathers!); Eclipse-style doubleneck guitars; "V" and "Explorer" series guitars; and an electric mandolin.

CIRCLE 31 ON READER SERVICE CARD

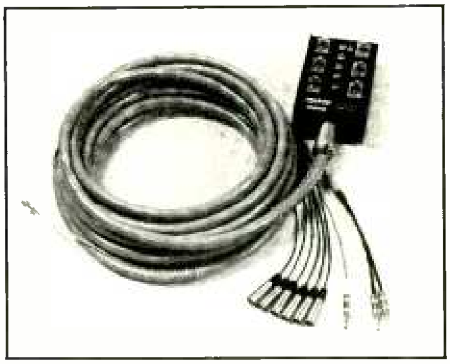
For bass players only comes the news from Ken Smith Basses, Ltd. The top model in the Smith line is the IIG Electric Bass, which features an exclusive active/passive electronics package combining passive controls for master volume, pickup balance, tone and phase switch with active circuitry including a 4-position state variable filter with frequency and Q controls, and an envelope follower with up-drive and down-drive capabilities. The instrument itself is a neck-through-body design with a figured maple body and laminated maple neck with graphite carbon-fiber inlay. Components include Bill Lawrence pickups with rosewood covers, gold-plated Badass II™ Bridge, an engraved goldplated brass control plate and gold-plated hardware. On a somewhat more modest scale is the PAS-II, which is the same basic design executed with mahogany trim rather than rosewood, chrome plated components and hardware rather than gold and conventional passive electronics. Smith also makes the S-II Series of Basses which feature a joined neck construction; the three-piece maple and rosewood neck is joined to the two-piece maple body with a dovetail joint for strength. The latest variation offered by Ken Smith Basses is the B.T. Series, which puts active electronics (15 dB boost/cut bass & treble controls, separate volume controls for two pickups, phase switch and three-way pickup selector) into standard Smith woodwork.

CIRCLE 32 ON READER SERVICE CARD

AUDIO CABLING

Whirlwind Music, Inc. has developed a new series of multicable systems known as the Medusa Power Series designed specifically for use with the new, high-performance powered P.A. mixers. The new system is the first to combine both microphone lines and high-power speaker leads into one custom-manufactured multicable (patent applied for). The Medusa Power Series is available with 6, 8 or 12 microphone input lines and 14-gauge speaker wires which come in a variety of configurations.

Whirlwind has also announced that it will introduce redesigned versions of its two common configurations of the ¼-inch phone plug. The #280 connector, which has been the standard on Whirlwind's Snake series guitar cables, has been redesigned to provide adequate



cable strain relief internally, rather than requiring an outboard strain relief arrangement. The #70, large barrel ¼-inch connector, which Whirlwind has been using on its SK series of speaker cables has been plagued by physical failure of the connector where the separate shaft and body were joined by a rivet; in the redesigned version, the shaft and body of the connector are now tooled from the same piece of metal to strengthen that high-strain area.

CIRCLE 34 ON READER SERVICE CARD

Recording Techniques Part 3

by Bruce Bartlett

Good microphones can make the difference between a mediocre recording and a professional-sounding one. If you're unclear on the characteristics of the various types of microphones, choosing the right one can be difficult. But by learning a few definitions you can select and use microphones with better results.

A microphone is a *transducer*, a device that converts one form of energy into another. Specifically, a microphone converts *acoustical energy* (sound) into electrical energy (the *signal*). This conversion is done because electrical energy is relatively easy to control, manipulate and record. By controlling the signal, you also control the recorded sound of the music.

Transducer Type: Condenser or Dynamic

Microphones for recording can be divided into two types depending on their operating principle: *condenser* or *dynamic*. In a condenser microphone (*Figure 1*), a conductive diaphragm and an adjacent metallic disk are charged to form two plates of a capacitor. Sound waves vibrate the diaphragm, thereby varying the capacitance and producing an electrical signal. With dynamic microphones (*Figure 2*), the signal is generated when sound waves cause a conductive coil or metal ribbon to vibrate in a magnetic field.

A condenser (or *electret condenser*) microphone generally provides a more extended frequency response, faster transient response and a higher output than an equally priced dynamic microphone. However, condensers require a power supply to operate, e.g., a battery. The battery must be checked and periodically replaced. Many mixing consoles have built-in *simplex* or *phantom powering* for condenser microphones, so that the microphone simply plugs into the console for power. Low-cost condenser microphones are more likely to distort than dynamics when picking up loud sound sources, and they are also more susceptible to degradation from high temperature and humidity.

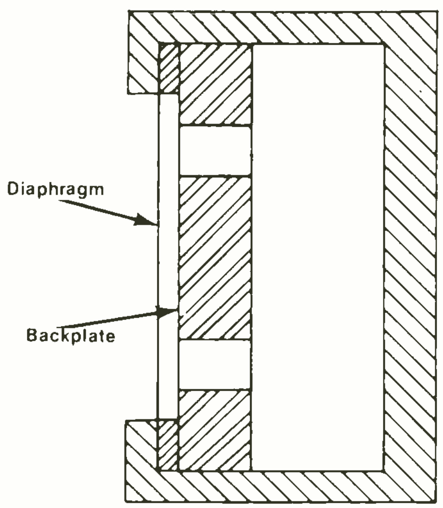


Figure 1: Condenser transducer.

Dynamic microphones, on the other hand, work without any power supply and provide a reliable signal under a wide range of environmental conditions. It is almost impossible to overload a well-designed dynamic with a loud sound source.

Two divisions of dynamic microphones are the *moving-coil* type and the *ribbon* type. Quality moving-coil microphones are very rugged. They generally have a "rougher" response than ribbons or condensers, although moving-coil mics of excellent quality are available. Ribbon microphones, while more delicate than the moving-coil variety, are often prized for their warm, smooth tone quality.

The sound that condenser microphones provide is generally clear, smooth and detailed, making them suitable for pickup of cymbals, snare drum, acoustic instruments and studio vocals. Flat-response moving-coil mics are often seen on woodwinds and brass. Moving-coil microphones emphasizing the "presence" range around 5 kHz are typically used on amplified instruments, drums and stage vocals. The mellow sound of a ribbon is often preferred for brass instruments. Of course, any microphone can be used on any instrument; use whatever sounds good to you.

Experiment with as many microphones as you can to become familiar with their distinctive "sounds."

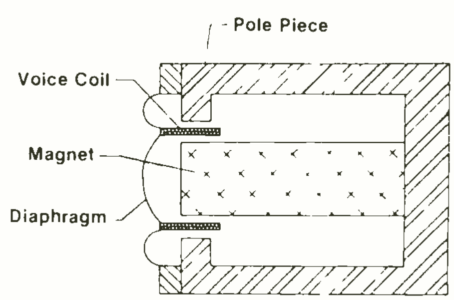


Figure 2: Dynamic transducer.

Directional Patterns (Polar Patterns)

Microphones also differ in the way they respond to sounds coming from different directions. An *omnidirectional* microphone is equally sensitive to sounds arriving from all directions. A *unidirectional* microphone is most sensitive to sounds arriving from one direction—in front of the microphone—but discriminates against sounds entering the sides or rear of the microphone. A *bidirectional* microphone is most sensitive to sounds arriving from two directions—in front of and behind the microphone—but rejects sounds entering the sides. *Figure 3* shows various polar patterns.

The unidirectional classifications can be further divided into *cardioid, supercardioid* and *hypercardioid* pickup characteristics. Probably the most popular of the three is the cardioid or heart-shaped pattern. A microphone with this characteristic is most sensitive to sounds arriving from a broad angle in front of the microphone. It is about 6 dB less sensitive at the sides, and about 15 to 20 dB less sensitive at the rear. You can hear how this pattern works by talking into a cardioid microphone from all sides while listening to its output. Your reproduced voice will be loudest when you talk into the front of the microphone and softest when you talk into the rear of the microphone.

A cardioid's rejection of sound from the rear can be used to discriminate against unwanted sounds such as leakage, poor room acoustics (reverberation) or feedback. A pair of cardioid mics can be angled apart to record large ensembles in stereo.

Despite its name, a supercardioid microphone is not necessarily superior to a cardioid microphone. Supercardioids do not reject sound from the rear as well as cardioids. However, supercardioids are less sensitive to sounds from the sides than cardioids, and so have a "tighter" pickup pattern. The same is true to a greater extent for hypercardioid microphones.

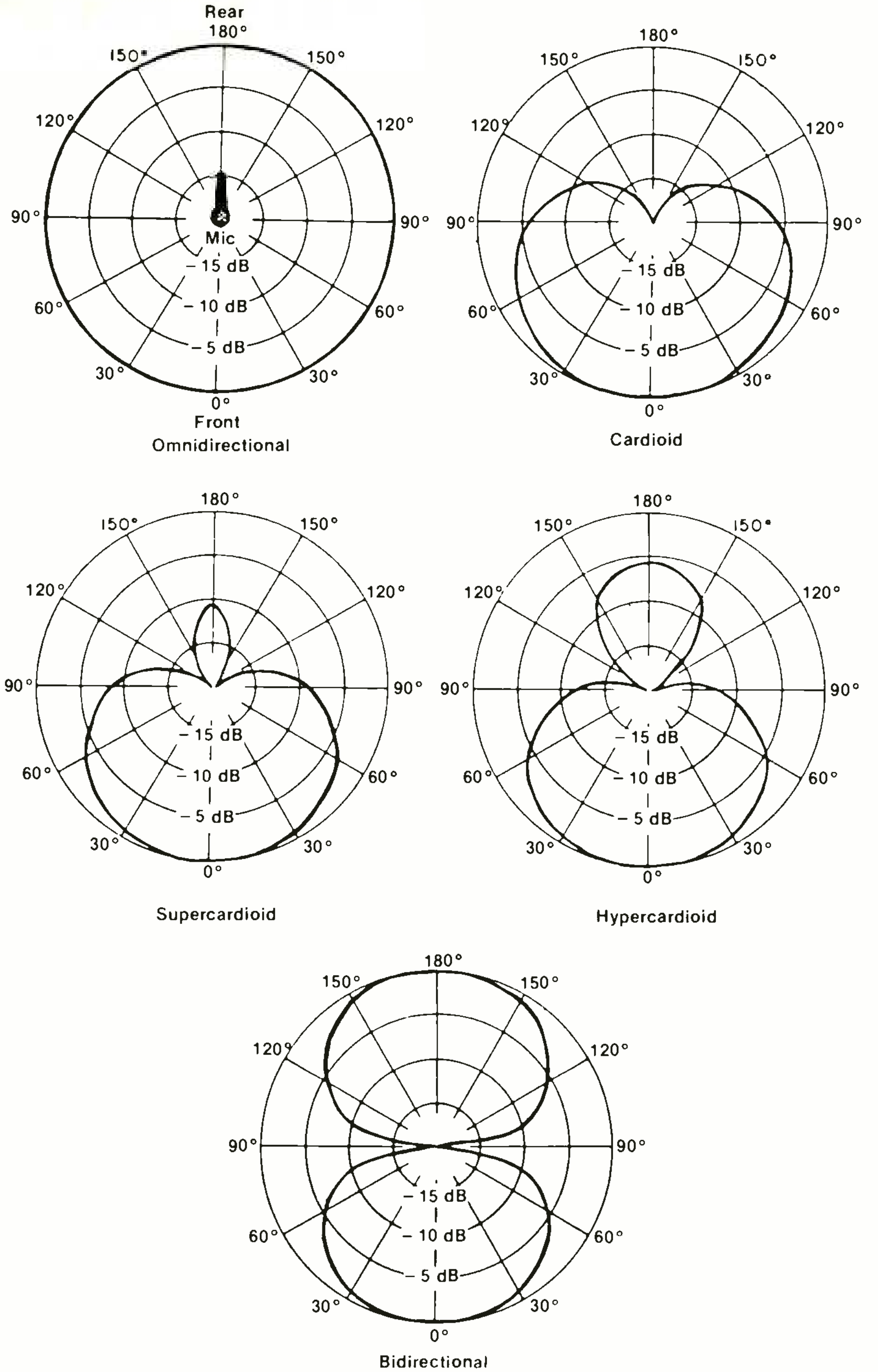


Figure 3: Various polar patterns. Sensitivity is plotted vs. angle of sound incidence.

Most unidirectional and bidirectional microphones boost the bass progressively when placed closer and closer to a sound source (say, from less than 2' to close-up). This bass boost related to close mic placement is called *proximity effect*. The "warmth" created by proximity effect can add robustness to a thin-voiced singer or fullness to a tom-tom, and gives a popular vocal sound for sound-reinforcement systems. In some recording situations, however, proximity effect lends an unnatural boomy or bassy sound to the instrument or voice the microphone is picking up. To reduce it, some microphones are specially designed to minimize proximity effect; other have a bass-rolloff switch to compensate for the bass boost. You also can compensate for proximity effect by rolling off the bass on your mixer's equalizers. By doing so, you also reduce low-frequency leakage and feedback picked up by the microphone.

An omnidirectional microphone has some characteristics that make it especially useful for certain applications. Omni's are much less sensitive to "pop" (explosive breath sounds) and handling noise than are uni's. Omni's have no proximity effect, so they can be used up close with no resulting bass boost. Their all-around pickup pattern helps to add ambiance to a recording made in an overly "dry" or "dead" concert hall. In general, omni's cost less than uni's of comparable frequency response. Omnidirectional microphones pick up more reverberation than unidirectional microphones when both types are placed the same distance from the sound source. Conversely, uni's can be placed farther from the sound source than omni's to pick up the same direct-to-reverberant sound ratio. For example, a cardioid microphone can be placed 1.7 times the distance from a sound source that an omni is placed in order to pick up the same ratio of direct-to-reverberant sound.

We stated earlier that omni's are equally sensitive to sounds coming from all directions. Actually, that is true only for frequencies below about 1 kHz. At higher frequencies, an omni tends to become unidirectional because the microphone itself acts as a baffle, blocking high frequencies. You can hear this effect by recording yourself repeating the word "seven" into an omni mic while rotating it around the front grille. In front of the microphone the sound is clear, but to the side or rear the sound is duller. This effect is minimized if the diaphragm is small or if the mic aims straight up.

The shape of a microphone has little to do with its polar pattern. For example, a microphone with a ball-shaped grille is not necessarily omnidirectional; a long probe-shaped microphone is not necessarily unidirectional. Talk tests and manufacturers' data sheets best indicate the directional characteristics of a microphone. Note that either the condenser or dynamic types mentioned earlier can be obtained with any kind of directional pattern.

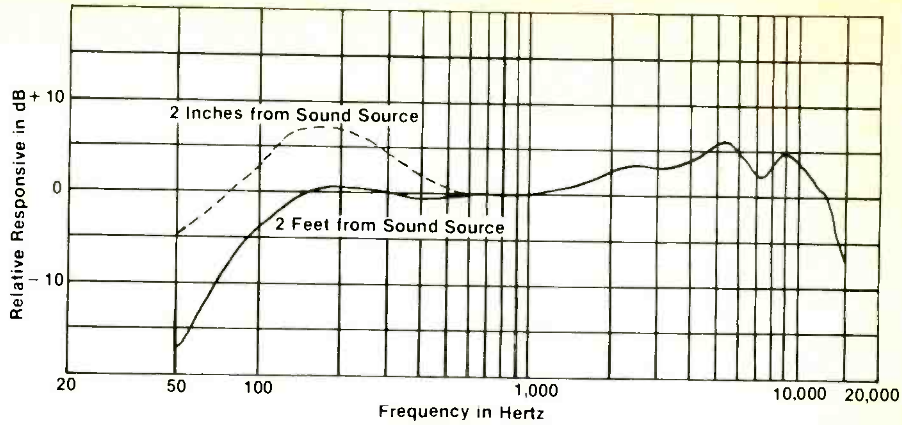


Figure 4: Example of frequency response of a microphone with proximity effect and a presence peak around 5 kHz.

Frequency Response

Frequency response is a measure of the tonal coloration a microphone will impart to the instrument it is used on. The frequency-response curves shown in manufacturers' data sheets give a general idea of how that microphone should sound compared to other microphones at a miking distance of 1 to 3'. For example, a microphone with a rising high end or a "presence peak" around 5 kHz tends to make instruments and voices sound crisp and articulate. A microphone with a limited-range response, say 200 to 7,000 Hz, does not reproduce all the high-frequency harmonics or the deep fundamentals. Such a microphone used to record a marching band, for example, would make the cymbals sound dull; the bass drum, thin. On the other hand, a high-quality microphone with a wide, flat frequency response has little or no coloration of its own. It usually records voices or instruments with high fidelity.

There are situations, however, where a non-flat microphone gives the most natural sound. Suppose you mic an acoustic guitar very close to the sound hole. It will sound bassy, even if you use a flat-response omni-directional microphone. That is because the sound hole resonates at low frequencies, giving a boomy tone quality which the microphone picks up and emphasizes. A more natural tonal balance for that microphone placement can be achieved if the microphone has a low-frequency rolloff. Some manufacturers make special clip-on mics that have such a rolloff for the acoustic guitar.

Here's another application where a non-flat response gives the best fidelity. Suppose you record a very loud instrument, such as a guitar amp or drums, with a flat-response microphone. When you play back the recording at a lower volume level than the "live" instrument played at, you will hear less bass and treble in the reproduction than you heard from the original "live" instrument. The sound may be thin and lacking in punch or presence. The reason is that the ear is less sensitive to low and high frequencies at low volume levels than at high volume levels (this is called the Fletcher-Munson effect). So, to make a loud instrument sound full and clear when reproduced at a less-than-"live" loudness, it may help to use a microphone with proximity effect (for bass boost) and a presence peak (for treble boost) to compensate for hearing phenomena occurring with a lower-level playback. *Fig. 4*, shows a typical frequency response of such a microphone—one with proximity effect and a presence peak around 5 kHz.

The low-frequency response of the microphone you use on a particular instrument should be limited to the lowest fundamental frequency of that instrument, if possible. For example, the lowest note on an acoustic guitar is 82 Hz. A microphone used on the acoustic guitar should roll off below that frequency to avoid picking up low-frequency noises and room rumble. Some microphones provide a low frequency cutoff switch for this purpose, or you can try to filter out the unneeded lows on your mixing console.

Unlike speakers, the size of a microphone is not necessarily related to its frequency response. For example, some very small microphones have excellent bass response.

Off-Axis Coloration

While a microphone may have an acceptable response for on-axis sounds (those arriving from in front of the microphone), it may have a different response to sounds reaching it from other directions. Even if a microphone measures "flat" on axis, at other angles it may have a high-frequency rolloff or an uneven response which produces a dull or colored tonal quality. Some examples of off-axis pickup are leakage from other instruments, multiple sound sources being picked up by one microphone (as in drum-set miking or ensemble recording) and moving sound sources (such as theatrical performers). To minimize off-axis coloration:

- Place cymbals and percussion as on-axis to the microphone as possible.
- In overdubbing situations, place the recording microphones relatively far from the instruments to pick them up within a smaller angle, more on axis.
- Use microphones with small diaphragms (less than 1" diameter).
- When recording several in-

Introducing TDK AD-X. The normal bias tape with Super Avilyn technology.

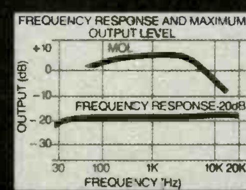
New TDK AD-X is the first normal bias audio cassette to use TDK's Avilyn magnetic particle—based on the renowned Super Avilyn formulation that has kept TDK the leader in audio and videotape technology.

The Avilyn advantage offered in AD-X is demonstrably clear. You now can record and play back—in the normal bias/EQ position with complete compatibility for any cassette deck over a wider dynamic range and with far less distortion. Even at higher recording levels, the increased headroom in new AD-X can easily handle strong signal input without oversaturation.

Record at higher levels with far less distortion.

When you hear the brilliant playback resulting from the higher MOL and lower bias noise you won't believe that your deck can "improve" so much.

The new AD-X has truly versatile applications. Its higher sensitivity makes it ideal for all-round home entertainment use and also suitable for any cassette player.



Avilyn magnetic particle achieves higher saturation and lower noise.

To ensure years of reliable use, AD-X is housed in TDK's Laboratory Standard Mechanism, and protected by TDK's lifetime warranty. With its distinctive packaging, you won't miss it.

So for high quality recordings in the normal bias/EQ position, snap in the new TDK AD-X. You'll discover that the Avilyn advantage means superior overall performance for you.

TDK
THE MACHINE FOR YOUR MACHINE

CIRCLE 67 ON READER SERVICE CARD

struments or vocalists surrounding a single microphone, aim the mic down above the center of the group or aim it up below the center of the group.

Sensitivity is a measure of the efficiency of a microphone. A very sensitive microphone produces a relatively high output voltage for a sound source of a given loudness. High sensitivity is an asset when you're recording quiet sound sources. For such sources, much gain is required on the mixer for an adequate record level (resulting in increased amplifier noise) unless the microphone sensitivity is high.

One way that microphone sensitivity is specified is in "dB re 1 volt per microbar." That figure tells what voltage the microphone produces (in dB relative to 1 volt) when picking up a 1,000 Hz tone at 74 dB/SPL (normal conversational level). A sensitivity specification of −60 dB is considered high (typical for condenser microphones); −75 dB is considered medium (typical of moving-coil microphones); and −85 dB is considered low sensitivity (typical of ribbon or small moving-coil microphones). Differences of a few dB among microphones are not critical.

The louder the sound source, the higher the signal voltage the microphone produces. For example, if a microphone produces −75 dB (0.18 mV) for a 74 dB/SPL source, it will produce −9 dB (0.35 V) for a 140 dB/SPL source (like a kick drum). Such a voltage can easily overload a microphone preamp, unless it is reduced before it reaches the preamp by a resistive network called a *pad* or *attenuator*. Your microphone dealer can supply an external pad to insert in mic lines or you can use the "input attenuator" switches on your mixer when recording very loud instruments.

Impedance is a measure of the a.c. electrical resistance of a microphone. Microphone impedances from about 150 ohms to 600 ohms are considered low; impedances around 40 kilohms are considered high. Low-impedance microphones are preferred for recording because they pick up less hum than high-impedance mics, and they allow long cable runs without serious high-frequency losses.

If you plug a low-impedance mic directly into a high-impedance input (such as a guitar amp), the microphone will work but will require a lot of gain for an adequate signal level—resulting in audible amplifier hiss. To match a low-impedance mic to a high-impedance input, you need an in-line transformer, available from your microphone dealer.

If you plug a high-impedance microphone into a low-impedance input, you may lose level and temporarily alter the frequency response of the microphone.

Hum

Microphones are sensitive to other "inputs" besides sound, such as hum fields, pop and handling noises. Minimizing pickup of these noises is important wherever there is concern for high-quality audio.

Alternating currents in power wiring, transformers and silicon-controlled-rectifier dimmers radiate magnetic and electrostatic fields which are picked up by microphone wiring and cables. When amplified, the induced alternating voltages are audible as a low tone or buzz called *hum*. To minimize hum pickup:

- Keep microphone lines away from power lines and cross them at right angles.
- Use condenser microphones.
- Use dynamic microphones with humbucking coils.
- Use low-impedance, balanced-line microphones.
- Avoid SCR dimmers and fluorescent lights.
- Aim the microphone for minimum hum pickup.
- Be sure microphone-connector plugs are securely screwed or locked into microphone handles.

Pop

When a vocalist sings a word emphasizing "p" or "t" sounds, a turbulent puff of air is forced from his or her mouth. A microphone placed close to the mouth is hit by this air puff, resulting in a thump or little explosion called a "pop." Even high-hat cymbals force out a puff of air that can pop a snare-drum mic. To minimize pop pickup:

- Place the microphone at least 1' away from the mouth.
- If you must mic close, place the microphone less than 3" from the mouth.
- Use microphones with built-in *pop filters* (also called *windscreens* or *ball grilles*).
- Put an open-cell foam pop filter on the microphone, keeping an air space between the foam and the microphone grille, if possible.
- Place the microphone out of the path of pop travel—above, below, or to the side of the mouth.
- Use omnidirectional microphones instead of unidirectionals if feedback and leakage are not severe.

Handling Noise

Microphones pick up thumps or scraping-handle noises (mechanical vibration) if they are on a microphone stand that is kicked or handled, or if they are hand-held. To minimize pickup of handling noises:

- Place microphones in stands, rather than letting vocalists hold them.
- Use microphones with built-in shock mounts.
- Place microphones in external shock mounts on stands.
- Use condensers instead of dynamics.
- Use omnidirectional mics instead of cardioids. Note, however, that some cardioid mics with effective internal shock mounts are comparable to omnidirectional mics in handling noise rejection.

Cables and Connectors

A microphone cable carries the electrical signal from the microphone to the mixing console or tape recorder. With low impedance microphones, you can use hundreds of feet of cable with little or no signal degradation. Some microphones have a permanently attached cable for convenience and low cost; others have a connector in the handle to accept a separate microphone cable. The second method is preferred for serious recording because if the cable breaks, you have to repair or replace only the cable, not the whole microphone.

Microphone cables are made of one or two insulated conductors surrounded by a metallic shield to keep out electrostatic hum. If you hear a loud buzz when you plug in a microphone, check that the shield is securely soldered in place. To wire a 2-connector shielded, balanced-line cable to a 3-pin connector: solder the shield to pin 1; solder one conductor lead to pin 2; and solder the other lead to pin 3. If the microphone output is 3-pin balanced, but your mixer or tape deck input is unbalanced, solder the shield and one conductor to the long lug (or sleeve) and solder the other conductor to the short lug (the "hot" or "tip" terminal) of the phone plug. Use the same color code for wiring all cables.

Phase

Some microphones produce a positive voltage at pin 2 with respect to pin 3 at the instant the sound pressure pushes the diaphragm in (positive sound pressure). Other microphones may be wired in opposite polarity. Microphones that are wired out-of-phase, when combined to the same channel, can cancel low frequencies in the sound pickup. To prevent this from happening, check that all your microphones are wired in-phase as follows:

Choose one microphone as a phase or polarity reference. Plug it into your mixer. Talk into it from about 3" away and set the meter to peak around 0 VU. Now, do the same with a second microphone. With both microphones feeding the same channel, hold both mics together and talk into them again from a distance of 3". If the meter reading peaks around 0 VU or higher, the mics are in-phase. If the meter reading is lower, the second mic is out-of-phase with the reference. In that case, remove the plug from the second microphone and proceed to reverse the connections to pins 2 and 3 inside the microphone.

Junction Boxes and Snakes

It is messy and time-consuming to run several individual cables from many microphones to a mixer. Instead, you can plug all the microphones in the studio into a junction box with multiple connectors. A single, thick, multiconductor cable (which is called a *snake*) carries the signals to the mixer. At the mixer end, the cable divides or *fans out* into several separate mic connectors to plug into the mixer.

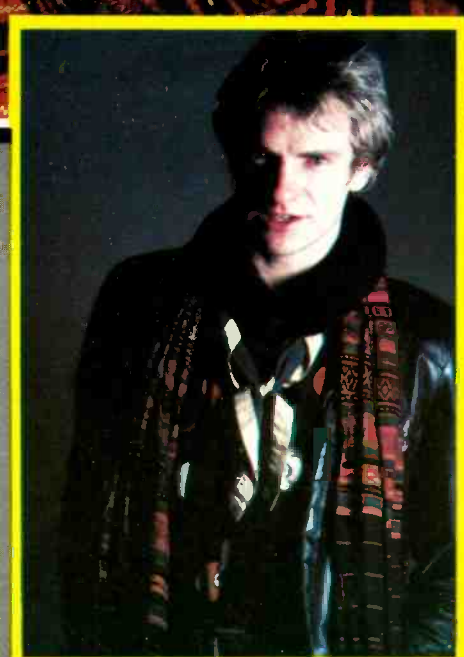
Splitters

If you are recording a "live" band that is using a sound-reinforcement system, you need each mic to feed a signal simultaneously to your recording mixer and the band's sound-reinforcement mixer. A *microphone splitter* is required. It has one input for each microphone and two isolated outputs per microphone to feed each mixer. Splitters can be obtained from your sound dealer.

Conclusion

We've discussed some microphone types, characteristics, problems and accessories. You should have a better idea about what kind of microphone to choose for your own application.

Keep in mind that there is no one "correct" microphone to use on any particular instrument, because you choose the microphone that gives you the sound you want. Most recordings, however, start with an accurate, natural-sounding pickup and in these cases the quality of the microphone is always important. But even an excellent microphone cannot sound good without the proper *placement*. We'll begin to explore microphone placement in an upcoming issue.



THE POLICE "LIVE!"

by Ellen Zoe Golden

"I don't think this bus is comfortable at all and I'm not coming back in it. Kim (Turner, the Police's road manager) is getting me a limo," guitarist Andy Summers says adamantly as he shifts a bit on the couch of the luxury bus transporting the Police, some crew and a few journalists from New York City to the Police's show at Philadelphia's Spectrum Arena. "I'm sitting here getting sick from being batted all over the place," he rants. "This is for groups like Molly Hatchet who want to get boozed out and lay some scrubbers in the back. It's not for English gentlemen like us."

Wait a minute, hold that thought. Is this the same guitarist who, along with singer/bassist Sting and drummer Stewart Copeland, made his first sojourn around the United States in a small, rented van? Yes, but things have changed a bit in the four years since the fledgling trio self-financed a U.S. tour on the successful heels of "Fall Out" (an independent import single on Miles Copeland's Illegal Records), and "Roxanne" (a cut from a domestically unreleased A&M album). Fortunately, the discount tour introduced enough Americans to a unique brand of reggae-threaded pop, that the generated excitement had A&M releasing Outlandos d'Amour *stateside in mid-tour. It sold over 500,000 copies.*

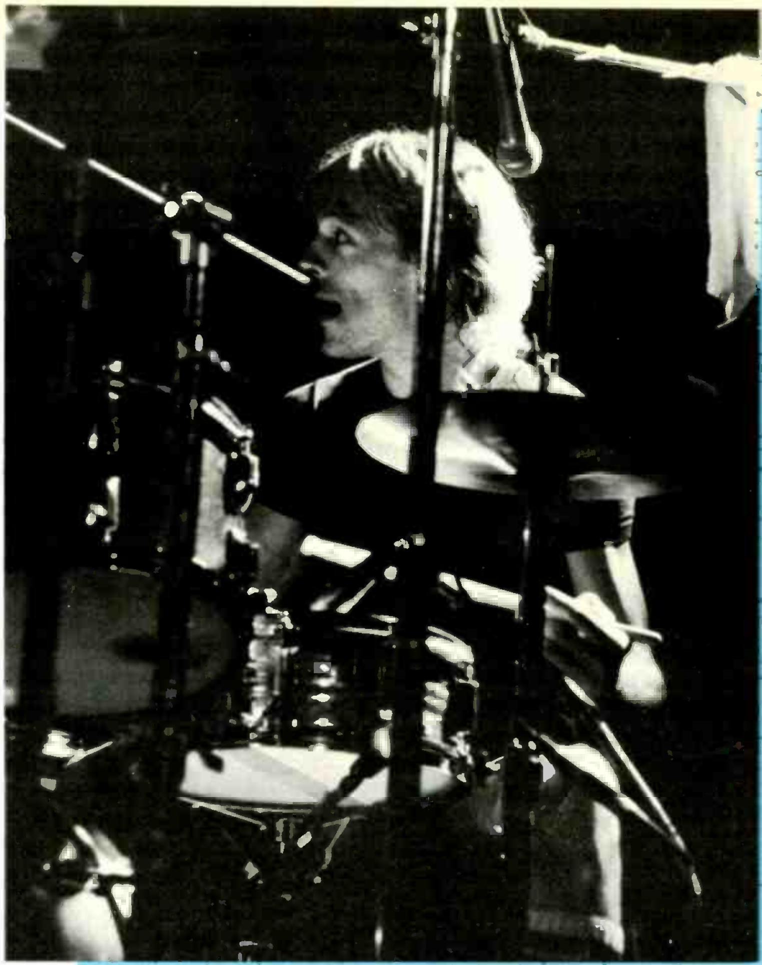
Today as the Police sit in their tour bus conducting interviews and planning song lists for the evening's sold-out performance, it's easy to applaud the band's strategy for success. And strategy it is, Copeland contends. He should know, having conceived the idea for a sensible rock threesome in England during the late 70s. The country was being torn musically between punk groups and the excessive progressive musicians, and Copeland slated the Police to fall somewhere logically between the two camps. Copeland's brother—and the band's manager—Miles Copeland, once said that "the philosophy of the Police has been a three-piece, condensed [unit that] recorded cheaply, keeping everything basically as simple as possible and capturing that element of what made rock music great in the first place." About the only variation on that theme has been the freedom of *choice that success has afforded. Where the Police once scoured the States in a rented van to trim costs, they now reflect on recent global excursions. And though the debut,* Outlandos d'Amour *cost a mere $6,000 to make, the Police have generated enough capital from* Regatta de Blanc *and* Zenyatta Mondatta *to choose, if they like, more elaborate studios in more exotic locations.*

Such a choice was AIR Studios on the Caribbean isle of Montserrat, where the Police recently capped off the diversified recording sessions of their fourth LP, Ghost In The Machine. *Part of the fresh sound on the album is due to the innovation of co-producer Hugh Padgham, who sent the musicians up to record in different rooms of the facility, thus maintaining enough individual space for a very "live" sound.*

In addition to the ambience, Ghost *explores a myriad of new styles. There's a mesh of jazz, funk, calypso, rock and latin sharpness backed by a variety of new beats in addition to a refined trademark reggae shuffle. Sting has solidified as a lyricist, as he makes clear, political statements without condemnation on songs like "Spirits In The Material World," "Invisible Sun" and "One World (Not Three)."*

Of the album's decidely funky flavor. Sting maintains that his saxophone playing seasoned it that way. Summers broke loose at times to add an acidic vengeance and Copeland complemented his famous drum styles by handling much of the keyboard chores—with the exception of "Every Little Thing She Does Is Magic," a Caribbean flavored rock love song that features Jean Roussel on piano.

Though Ghost *hugged the top of the charts for many weeks, that is a hard, cold mathematical conclusion to a story that doesn't even begin to explore the warmth, intelligence and sensitivity each of the individual members of the Police retains.* Modern Recording and Music *spent time on the road with the trio trying to unleash personalities and preferred musical styles, while discovering how, in a variety of ways, they handle the pressures (and pleasures) of success, including outside projects to nurture their creative egos.*



Modern Recording & Music: It sounds like the Police used more instruments on *Ghost In The Machine*...

Stewart Copeland: It's not more instruments, it's the same instruments overdubbed a lot more times.

MR&M: Was that an idea of your new co-producer, Hugh Padgham?

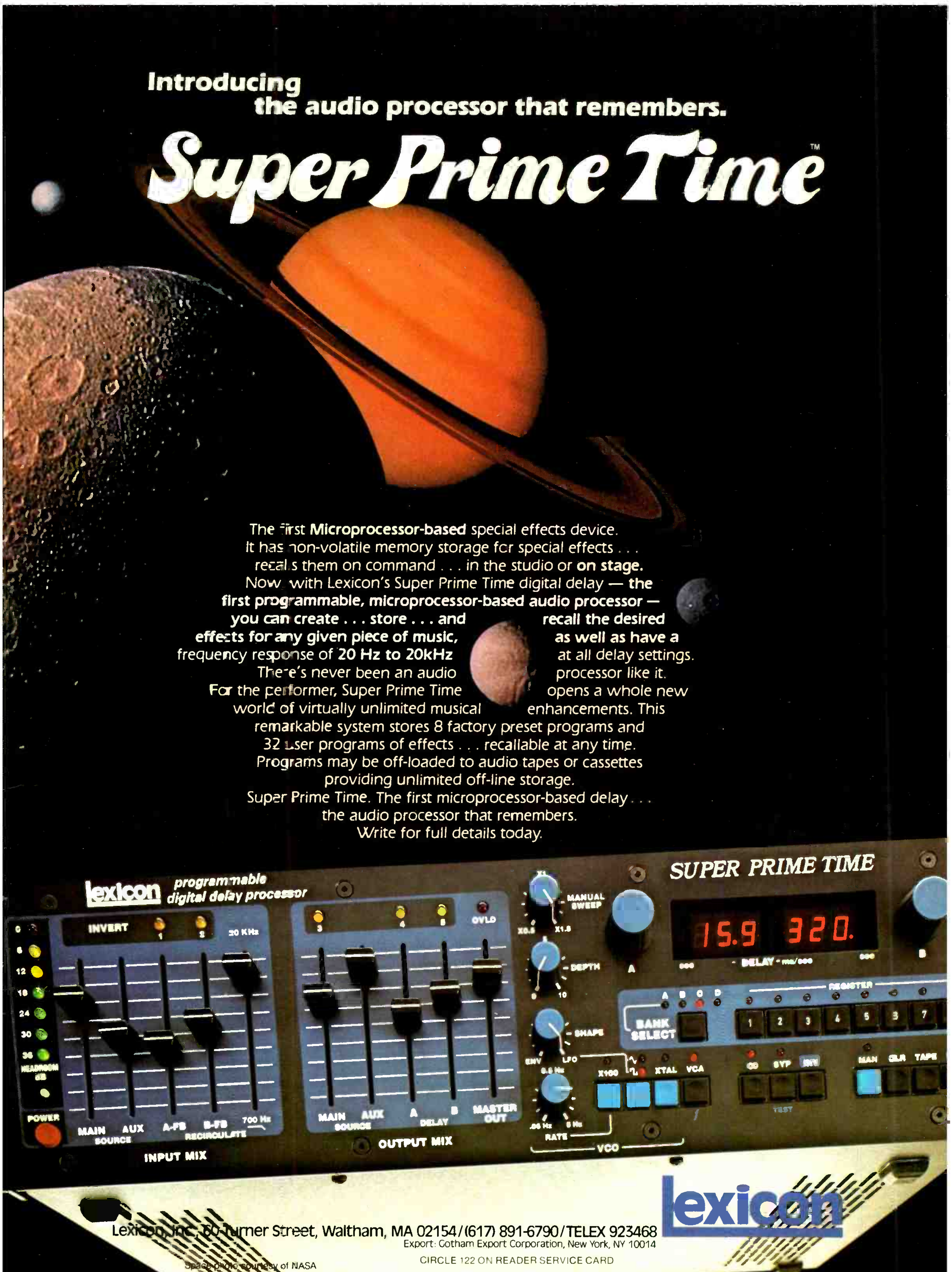
SC: No. It was just that we had more time to do that sort of thing. By the way, we've never had a producer, per se; we've just changed engineers. We had used Nigel Gray for three albums, and we figured that we could do well to use somebody with some new tricks. I was particularly impressed by the drum sound that Padgham's gotten on the Genesis album. He knows how to get a good sound, and that's what an engineer is all about.

We play the music and I can make my drum sound terrific in the room, but it's like what Kim [Turner, road manager and "live" sound engineer] does at shows: I can get my drums to sound great, but to get them to sound great through a microphone into an amplifier and coming out of speakers, there's got to be a lot of faking going on. Not so much faking...but a lot of technology has to be mastered and controlled and that's not my job; that's not what I'm good at. On stage, Kim takes care of that. And in the studio, we need somebody really clever with microphone placement. In fact, Padgham initiated a technique on this album where we were all split up. Andy was in the recording room, Sting was in the mixing room and I was in the dining room. What that means is that I was in a room all by myself—in a different building, even—and the drums sound very "live" because it's a clattery, echoey room. Sting was plugged straight into the [mixing] desk—he didn't use an amplifier—and Andy had the whole recording room to himself with his amplifiers turned up full blast.

Most bands record in the same room with partitions in-between the different musicians. The drums are all muffled up and close-miked. Since I was in a room all by myself, the microphones could be placed at the other end of the room. The other way you can't have microphones at the other end of the room because there is a guitarist blasting away and a bass player blasting away.

MR&M: Why did you pick AIR Studios?

SC: Because it was 12 hours from the



Introducing
the audio processor that remembers.
Super Prime Time™
The first Microprocessor-based special effects device.
It has non-volatile memory storage for special effects . . .
recalls them on command . . . in the studio or on stage.
Now, with Lexicon's Super Prime Time digital delay — the
first programmable, microprocessor-based audio processor —
you can create . . . store . . . and recall the desired
effects for any given piece of music, as well as have a
frequency response of 20 Hz to 20kHz at all delay settings.
There's never been an audio processor like it.
For the performer, Super Prime Time opens a whole new
world of virtually unlimited musical enhancements. This
remarkable system stores 8 factory preset programs and
32 user programs of effects . . . recallable at any time.
Programs may be off-loaded to audio tapes or cassettes
providing unlimited off-line storage.
Super Prime Time. The first microprocessor-based delay . . .
the audio processor that remembers.
Write for full details today.
lexicon
programmable digital delay processor
SUPER PRIME TIME
INVERT
INPUT MIX
OUTPUT MIX
POWER
MANUAL SWEEP
DEPTH
SHAPE
RATE
VCO
BANK SELECT
REGISTER
DELAY
15.9
320.
lexicon
Lexicon, Inc., 60 Turner Street, Waltham, MA 02154 / (617) 891-6790 / TELEX 923468
Export: Gotham Export Corporation, New York, NY 10014
CIRCLE 122 ON READER SERVICE CARD
Space photo courtesy of NASA

"I'm used to working with two guys who seem able to do anything."

nearest industrial distraction, and they had the best equipment there, too, except that they didn't have a computer for mixdown. I've gotten spoiled, I like computer mixdown.

MR&M: Do you have a studio in your home?

SC: Yeah, I do. I have an Allen & Heath 28-channel desk, 24-track Otari tape recorder and all sorts of auxiliary equipment. I enjoy recording; I'm a recording enthusiast.

MR&M: Indeed. The first Police album only cost $6,000 to make. Now that the band makes more money, do you spend it?

SC: Well, the last album cost significantly more. Not because we were any more indulgent, but because the studio was more expensive and transporting all our equipment half way around the world, in addition to our other expenses, all added up. So next time we're probably going to record in London again.

We did have more time to spend in the studio this time—we had six weeks instead of four, and we've had lots of breathing space between our last album and this one to consider what we were going to do in the studio...

MR&M: Does that account for the growth obvious on *Ghost*?

SC: Well, maybe. With every album we attempt to take a chance and every album seems to be a departure from the previous one. But we've accomplished more of a departure this time. It's not a new philosophy or anything, but we've just been successful.

MR&M: The record did make a clearer political statement this time. Why is that?

SC: The way the whole group is involved in the answer of that question is that all of us have been subject to the same kind of background. I grew up in what is politely called the Third World [as son of Miles Copeland, co-founder of the CIA, Copeland spent much of his early days living in Egypt and other parts of the Middle East], which Sting has recently discovered as a source of songwriting. That's always been something that has been a part of my life, but I've never been able to think of a way to put it into music. It isn't really a musical subject for me. I think Sting's actually accomplished something by being able to do that. But as far as the effect on all three of us goes, I suppose it's a natural result of having traveled to those places to play and having seen them with our own eyes, smelled them with our own noses and heard those cries of anguish with our own ears.

MR&M: Would you like to do more of the writing for the band?

SC: I do lots of writing, but a lot of it isn't appropriate for this group. Needless to say, I'm green with envy over Sting. But, you know, I have to be philosophical about it. I mean, if I were more of a singer, I'd probably end up writing more of the songs. I did write "Darkness" and "Rehumanize Yourself," but I think it's really important for a singer to sing a song that he can identify with.

MR&M: What was your inspiration for those songs? Are you a philosophical person?

SC: I am, but I'm into prose, discourse and argument; fitting it into a few sleek lines of verse is not really my style. I like to argue at length, and that's not what it takes to write songs. You have to generalize and be very economical in your statements when you write, and I'm specific in my thinking. On political issues, instead of thinking, "I hate the Commies," I think, "In this country this system works and in that country it doesn't work." Instead of saying, "Reagan is an asshole," I think, "Reagan has accomplished things in certain areas and in certain other areas he is totally fucked up." For writing songs, that's not the right kind of mind you need.

MR&M: Was there anything special you remember about the recent sessions?

SC: I remember at one point our accountant actually came down there because there was some paper work that absolutely had to be done so things could progress. At one point, while I was playing keyboards on "Darkness," the accountant was literally tripping over me. Something I'll always remember is having the accountant tripping over me while I was trying to record. It could have been anybody else—like the tea boy—but the fact that it was the accountant and I was actually playing the take at the time, seems to be a kind of poetry.

MR&M: Stewart, I meant was there anything the band got to try in the recording process that hadn't been done before?

SC: I didn't get to play any guitar this time. There weren't any gaps for me to do that. In the past, for one reason or another, I've ended up with an opportunity to play the guitar. You know, I'm a frustrated guitarist and a frustrated piano player, a frustrated everything.

MR&M: Is that the frustration that makes you "green with envy" over Sting? He seems to get most of the attention...

SC: Yeah, he certainly does, but I'm not sure if that means we've got lopsided power within the group. He doesn't have more of a say and I can categorically tell you that there's no band leader. But Sting does write more of the material and is the more visible member of the group as the singer. That's a natural progression, and, as I said, I'm green with envy. But that's life. The point is that the instrument that really turns me on and that I enjoy playing the most—even though I'm a frustrated guitarist—is the drums.

The drums are like tennis: They might not seem like the most intellectual things in the world—I've got other things to occupy the higher areas of my mind—but I enjoy playing them. It actually takes a lot of concentration and it's a very deep subject that I find absorbing. On the other hand, the drums are not the primary focus of an audience or the world's attention. Humans breathe oxygen and when listening to music, humans tend to listen to the voice. The other instruments just fit around it. Take the other instruments away and they'd notice something missing, but they're usually focused in on the one element of singing.

MR&M: Has the Police become what you intended it to be?

SC: Yeah, it has. In fact, it sounds a bit Machiavellian, but it has actually gone according to plan. It sounds terrible to use words like plan and strategy with artistic concepts, but I say that quite innocently, really. We just decided we wanted to have a group that was free to develop in its own way and come up with our own goods, without the pressures of the industry. Now those pressures have to be scientifically avoided—you have to use your brain,

"The performance and technology sold me. The sound of the '90 sold my clients."

Guy Costa, Vice President and Managing Director, Motown/Hitsville Studios

"When we first got our hands on the Otari MTR-90 we were impressed. The tape handling is superb. The production features are all there. The electronics and logic are to the highest standard we've come across in all the years of Motown's recording history.

But, as I've learned over the past nineteen years in this music recording business, it isn't just specs that count. The producers and artists have to like the way it sounds. A mastering multitrack machine has got to have a "musical" sound. Transparent. Clean. Performance that has to deliver everything possible — right up to the limitations of the tape.

I'd say that judging from the reactions of the creative people who record at Motown/Hitsville,

CIRCLE 156 ON READER SERVICE CARD

Otari's got a platinum record coming up for their New Workhorse.

And one added thing. We bought two because alot of the music product is going 48 track. The '90 synchronizes beautifully through the AudioKinetics controller and interface. Now, all we have to do is figure out how we can juggle this year's budget to get four more!

If you use your ears for a living, use your head too. Listen to the Otari MTR-90. I did."

OTARI

The New Workhorse

Otari Corporation, 2 Davis Drive
Belmont, California 94002
(415) 592-8311 Telex: 910-376-4890

not your art or your talent, not those mystical artistic things that lurk beneath the skin in your heart and soul. You have to use your brain and plot a course that will keep your art free of all distractions. And that's the part where we've been clever and Machiavellian. As far as the actual development of our music, that's where we follow our own instincts.

MR&M: How did the Police get together?

SC: Oh, nooooo!

MR&M: Wait a minute, I only asked that because the band has been known to tell conflicting stories.

SC: That's what happens in every interview. *[He laughs]* Somebody will ask a question and Andy will give his answer and I'll look at him in utter astonishment and say, "What?!" I mean, you saw it here when I was telling an interviewer something and Andy perked his ears up and said, "That's not true." We all have different ideas of the why, wherefore and how of the whole thing.

MR&M: Okay, so how'd you find Sting?

SC: He was playing in a jazz group in Newcastle. I was in a quite successful, established group [Curved Air] at the time and I was passing through that town when I saw Sting singing there. I took note and later, when I wanted to form a group, I called him up out of the blue.

MR&M: How did you think a jazz singer would fit into your group?

SC: I didn't regard Sting as a jazz singer. I regarded him as a singer who was playing in a jazz band. I don't think that band fit him well at all. He'll tell you something different, of course, but I thought he was conspicuous in that band and capable of something much more interesting.

MR&M: And what about Andy?

SC: A real creep you know. No, seriously, Andy actually discovered us. He saw us playing with the wrong guitarist doing the wrong thing and he joined us...

MR&M: Can you describe your feelings about Andy and Sting?

SC: Can we slip off into a quiet room somewhere so I can do that? If they said something nice about me, then I can say something nice about them...*[Laughs]* I'll tell you honestly and frankly: I have the highest regard for my colleagues, who are very talented musicians...

MR&M: Yes, Dr. Copeland...

SC: Well, obviously my feelings are too complicated. I'd have to spend a whole day talking about the feelings I have about Andy and another day talking about the feelings I have about Sting, because we've lived with each other for five years now and our feelings about each other are very complex and they vary from day to day.

MR&M: Okay then, let's talk about the next album. How are you going to top yourselves next time?

SC: Same way we topped ourselves last time, which is, well, you tell me...

MR&M: I saw more expansion and more refinement...

SC: No, not more expansion and more refinement, just different expansion and different refinement, with different avenues. Hopefully, between now and the time we go into the studio, I'll have new ideas and so will all of us.

MR&M: You seem a mite irritated?

SC: Irritated? No, no, not at all. It is a difficult question to answer. When asked, "How do you feel that your music is different from all the others?" it's hard to put your finger on it. There're different chords, different songs and different lyrics.

MR&M: Do you arrange and compose your songs before the whole band meets you in the studio?

SC: Yeah, note for note, but when we arrive in the studio it all gets changed. I sit at home in my studio and figure out the bass part and then record it. I work at it and work out how it fits the drum parts. I work out the guitar chords, and how it all fits together rhythmically. But, all that work is irrelevant because that process is just to sell my ideas to the other members of the group.

MR&M: Did they ever say, "That's horrible"?

SC: Oh yeah, frequently. In fact, more often than not. When we get into the studio it's a whole different story. Andy's got fifty chords in his [musical] vocabulary to every one of mine. And though Sting may have a very clear idea of what the rhythm should be on a song he has written, of course, I've got an even clearer idea of what kind of rhythm I'm going to play in the studio and that's the rhythm we end up with.

MR&M: A quick sidelight question: What about that chewing gum commercial that the Police made back in the early days. How come it's never been seen?

SC: Because the Wrigley's Chewing Gum commercial is still sitting somewhere and the people who have it don't even know what they've got in their possession. It's in England right now. The commercial was about a punk group in an agent's office stealing his cigars and pinching his secretary, and stuff like that. Finally, when the agent can't handle it anymore, he pulls out a stick of chewing gum. And that's the whole commercial. But, it was a bit too wild for Wrigley's, which is a very straight company. They decided not to use it.

MR&M: Would you like to get your hands on it?

SC: Yes, I would; I'd like to see it. I mean, I don't take my image that seriously. Certain other members of the group would not be caught dead...it'd be the same as if Marilyn Monroe's early porno flicks were discovered. I don't actually give a damn; I'd like to see it because it tickles my sense of humor.

MR&M: Speaking of maintaining an image, can you tell me about Klark Kent?

SC: Sure, if you'd like, I'll tell you what I know and we'll explore the mysteries that lie there. Plumb the depths...

MR&M: Stewart, what is Klark Kent and why are you putting out records under that name?

SC: I first became a member of the Church of Kinetic Ritual at the age of 14. I was a member of that religious cult and I met the leader of the church—Dr. Klark Kent—who was in a different form at that time...

MR&M: Are you putting me on?

SC: ...And he led me along the path of Kinetic Ritual, which is tied up with why music works and why church services work to bind people together. The same thing happens in a church that happens at a rock concert or a football game, when thousands of people act as one and all of their energy is focused onto a particular spot. In church it's very obvious what they are doing as they act together doing incantations and feeling the presence of God coming down on them. Dr. Kent would tell us that something else is happening: this presence is actually being created by those people. That's Kinetic Ritual—they create a higher consciousness by joint participation. And, um, it's a very long and involved subject...

Klark Kent's involvement is widespread. The Kent Foundation has a lot of political functions, some of which are quite shaky. He's involved in the religious things, as well as ballet works...and he's even acted in Egyptian movies.

MR&M: What about the records?

SC: That is only one of his many activities. He has made rock'n'roll albums and his works of poetry have been published in nineteen different languages. He's also done artistic works of many kinds under different names.

MR&M: How are you involved in the Klark Kent records?

SC: I am merely a humble devotee of this great man.

MR&M: But you play all the music on the records...

SC: Ha, I'm honored that you could suggest such a preposterous thing. Of course that is beyond my humble capabilities. I'm just a humble drummer in a rock'n'roll band...

MR&M: One with an active imagination...What kinds of music do you listen to for your own enjoyment?

SC: There are some kinds of music that seem to stick with me forever—Pieces by Ravel, and some things from Leo Kottke's six and twelve-string guitar. These are records that have stuck with me over the years, but I only play them once every ten years and that's it.

What I actually listen to on my [Sony] Walkman are the latest releases. Those change all the time. I mean, one week I'll get a Grace Jones album [Jones' *Nightclubbing* album featured the Police's "Demolition Man"] and I'll play nothing else for a week. I'll memorize it and then I'll get something new. I go through all the latest releases and I listen to them again and again until I get sick of them; then I never listen to them again.

MR&M: Do you ever find bands out on the road to bring back to brother Miles for his IRS Records?

SC: Occasionally, yes, but it all comes down to whether or not Miles is as enthusiastic as I am.

MR&M: How involved are you in IRS Records and FBI Booking Agency?

SC: Not really at all anymore. I started Illegal [one of a few British independent labels that merged to become IRS Records] to put out the first Police record, "Fall Out." As soon as I had done that, I wasn't really interested in making anybody else's records. Don't get me wrong, I do care about other groups and I would like to help them...

MR&M: Would you like to produce other groups?

SC: I don't think producing is my bag at all. I would get too frustrated. I wouldn't be able to put up with some drummer banging away; I'd rather do it than show him how to do it. Even when I do sessions with other musicians, I get frustrated because I'm used to working with Andy and Sting—two guys who seem able to do anything. It's not just that these are the guys whom I like best, it's just that I get frustrated by other people's limitations. As a result, I don't think I would be a good producer—I'd end up in pitch battles with the artist.

MR&M: I would think that some of that attitude would carry over into the Police's recording sessions...

SC: Yeah, it does, but then again, I'm up against some pretty stiff characters. So it equalizes itself out.

MR&M: That's exactly the stern image the Police have often given out.

SC: We have gotten this sort of grim atmosphere about us. It's because of the heaviness of the last album. Everyone thinks we are all into a morbid depression over the state of the world. I'm concerned over the state of the world and I think it's fine that Sting is expressing his concern through his songwriting; I'm happy that this subject is the lyrical content of the songs that I'm playing. But, my sense of humor is still intact. As is the sense of humor of the people who are living in utter deprivation over in India. Even though they are living under a car somewhere, they haven't lost their sense of humor.

The whole image thing comes down to this: As long as it looks good and people like it, fine. It doesn't bother me in the slightest.

MR&M: Aren't you perpetrating that image?

SC: Yeah, sure I am because it [the image] has a job to do. On the other hand, this *[he holds up* Ghost *LP]* is really our only contribution to our image, and this is about how important the blond hair, handsome faces and photogenic smiles are. Being sex symbols is quite fun, but it has nothing to do with selling records. That has to do with enjoying myself when I'm stuck in Minneapolis...

MR&M: How much has Stewart Copeland changed from the days of stuffing singles into sleeves to the days of platinum records?

SC: All three of us have always had a hell of a swagger—we've always been swollen headed since we started off. Even for our first gig in the States, we were just as egotistical as we are today. And I'm proud of that fact. So we don't have anymore self-confidence than we had in the beginning. In fact, we probably had even more then, because it was still in front of us to prove. Now we can relax, having proved it.

MR&M: What happens if you lose it?

SC: Well, there you go. That's what keeps us on our toes. I'm sure I have changed—I mean...I've grown up. When we started out I was 24, now I'm 29. The world changes, people change and I've changed.

It was depressing when I ran into some old school friends of mine who got their first job at an insurance company the minute they graduated from college. They put on their business suits and became 40 years old from that moment until the point when they will become 60 or so, and then retire.

MR&M: Are you always going to play rock'n'roll?

SC: No, there are other things that interest me. I'm into Super 8 photography and I've got a studio for it at home. I'd like to do a *Star Wars*, but obviously I don't have the time or the credentials. But there're lot of things that I can do. For example, I shot a film starring Andy Summers as the detective, Nat Hunt. And I'm getting a screenplay together for another film like that right now.

MR&M: Do you ever plan to use any of your films on stage?

SC: No. The reason we work together is because we have the music in common. I have no interest in trying to sell my film to the other members of the Police. It's enough of a hassle selling my songs to them.

Born in 1951 as Gordon Sumner in the ship-building town of Newcastle, England, singer/bassist Sting started out playing copy rock but shifted, at the age of fourteen, to improvisational jazz. Breeding jazz into his vocal styles as well, Sting was first spotted by Stewart Copeland while playing with a band called Last Exit. When Last Exit disbanded, Sting remembered his encounter with Copeland and joined Copeland and original guitarist Henri Padovani in the earliest version of the Police.

Immediately mastering most of songwriting and singing chores, Sting helped gel the embryonic trio (which now featured Andy Summers instead of Padovani on guitar) with hit after hit. No one could deny that his visibility as lead singer also enhanced the band's popularity, and it wasn't long before Sting wanted to try some outside activities. His wife, actress Frances Tomelty, persuaded him to try for the role of Ace Face in Quadrophenia. *He got the part. "I'm a budding Laurence Olivier," he remarks about his performance in the soon-to-be-released eccentric English film,* Brimstone and Treacle, *written by Dennis Potter* (Pennies From Heaven).

Sting's role backstage at the Spectrum is somewhat amusing: he dresses as a gangster, complete with a futuristic Steinberger "gun" bass guitar.

Modern Recording & Music: Why'd you decide to do the sax parts on *Ghost In The Machine* yourself?

Sting: I bought a saxophone in Japan in February [1981]. I had always wanted one, but until recently I could never afford one. I started to play and ended up doing some very basic, James Brown-type riffs, using the sax rhythmically and building it up as a section. I had a tenor and an alto, and I overdubbed about four tracks—two tracks of each—and it ended up sounding like a brass section. When it came to trying to do it on stage and trying to sing at the same time, even *I* found that difficult. So I thought, "We'll have to bring in some experts." But we couldn't get them *[he laughs]*, so we got the Chops [a three-man horn section].

I didn't try anything fancy in the studio with the saxophone. I just did very basic riffings. But the Chops have obviously adapted to the stage, and they are much less subtle.

MR&M: Can you tell us a bit more about your horn parts?

Sting: No, I just played the parts in-between the singing. I'd sing a bit, then play the saxophone and then sing a bit more. *Every* track had the saxophone on it. It did, actually. It was the first riff I learned to play—"Da Did Did Did Da"—and I laid it in in every key in every number and they had to spend a whole week wiping it off! *[Laughs]* By the end of the sessions, I had convinced them that it sounded good.

MR&M: *Ghost* is much funkier than the group's earlier works. Why is that?

Sting: I think the saxophones had such a strong color that they just changed the whole thing. It's that kind of injection, the addition of one color, that made the band sound terribly different. It set a tone and it set a mood. Also, the material was different.

I used more choral effects in my voice; I overdubbed my voice a lot more than I ever did before. I just got into doing that a lot. It's interesting, maybe next time we'll just have one voice. It's just the development; I find harmonies very easy and natural for me to do.

MR&M: How would you describe your vocal style?

Sting: I've really been influenced more by women singers than men, because of my range. Ella Fitzgerald, Billie Holiday, Cleo Lane and Joni Mitchell have influenced me more than Frank Sinatra. There aren't many rock singers that I appreciate, except maybe Little Richard, James Brown and Stevie Wonder.

MR&M: What about the political influences obvious on *Ghost*, where did they come from?

Sting: The album is not so much political as it is socially or world aware. The songs are perhaps more serious than ever, but then again I think it's the time we're at at the moment where we have to get serious, because the problems that we face are very serious. But even humble musicians have to be aware of them.

MR&M: How much do the Police get involved in the production of their records?

Sting: Producing is a very, very ambiguous thing. What does it mean? It certainly depends on the group. A band can go into the studio for the first time without a clue as to how to get a drum sound or a guitar sound and they need somebody to show them how to get it. We're slightly different because we know how to get our sounds. We've always produced ourselves, but then we were a knowledgeable group. I was 25 when we made our first album, and had a bit more experience [than most musicians]. Basically, a producer works the machinery. We have the ideas and we ask if such an idea is possible. We've been lucky in the past, we've had very pliable, good-thinking engineers.

MR&M: But what about co-producer, Hugh Padgham?

Sting: There's no such thing as a producer in this group. We have enlightened engineers.

MR&M: As a result, are the chores evenly divided between the three of you in the studio?

Sting: That's such a delicate area—personal egos. Since I write the songs and I sing them and I play the bass and I write the chords, obviously I have a lot to say. But, that doesn't mean that Andy's contribution is meaningless. It's just that in volume I do more. This is such a delicate area that to talk about it is damaging.

MR&M: Do you write your songs before you get to the studio?

Sting: Oh, absolutely. There's no question about it. I don't write on the road. I spend a few months at home in private and then I go into the studio and play demos to the group. It's the only way. I can't sit down with a guitar and play it to them; it has to be a fairly polished product.

MR&M: Have you any ideas yet for the next LP?

Sting: For the next album we are going to change our image totally. We are all going to shave our heads, wear fishermen's sweaters and sing songs about real ale. And we're going to play banjos like Pete Seeger and do folk stuff. That's the next thing we're going to conquer the world with. It's going to be called *Blowing In The Wind*.

MR&M: Sorry Sting, that's already been done.

Sting: Darn...

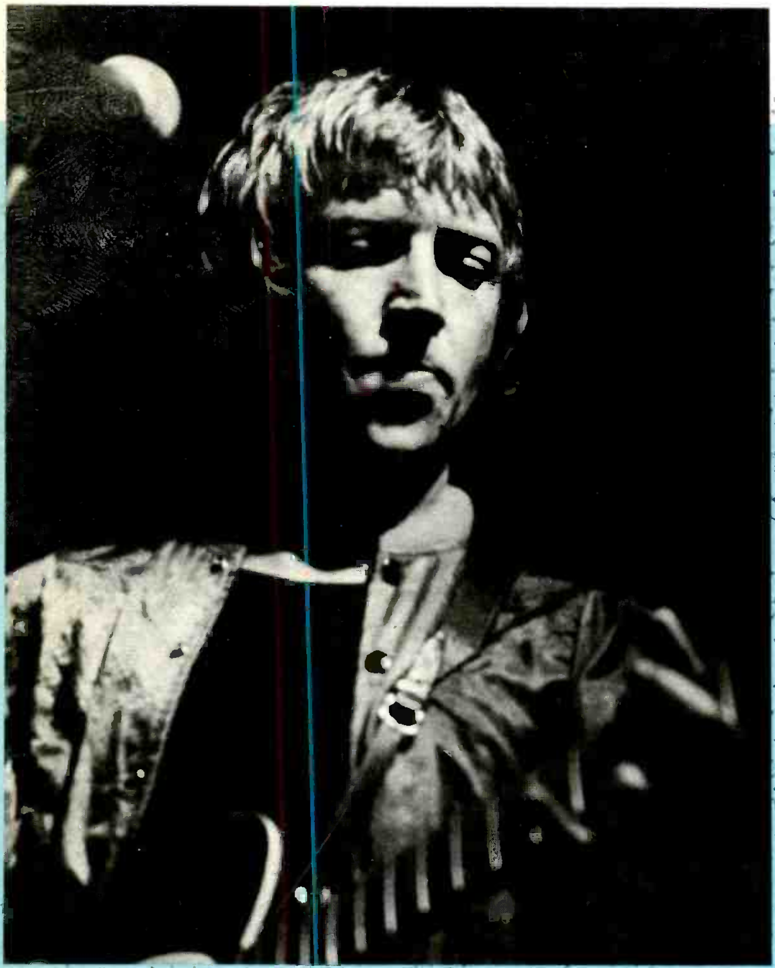
At 38, Andy Summers is the oldest member of the Police and a seasoned veteran of the progressive rock circuit. His stints with the Animals, Zoot Money, Kevin Coyne and Kevin Ayers established him as one of Britain's foremost guitarists. Yet, when the new music philosophies of punk started to take root, Summers shelved his "excessive" guitar styles and mastered the lean approach for his work in the Police.

At times, Summers maintains a dual personality. On the bus and backstage at the Spectrum, he gives the impression that he is a proper English gentleman, though he's been known to exhibit bold humor and his stage mannerisms are anything but tame. Before the Philadelphia show, however, Summers ignores the backstage antics of his colleagues, opting instead to sit in a corner and flip through his notebook. He has a lot on his mind right now: with the Police tour slated to run through March, he needs time to find a book deal for his photographs, and it'll be Spring before Summers can resume the recording sessions he began last year with Robert Fripp.

Modern Recording & Music: What's the status of the project you are doing with Robert Fripp?

Andy Summers: We'll probably finish the album off in May. We had two weeks recording together in September, at which time we started totally from scratch, learning how to play together. We had never played together before then.

MR&M: It seems like the two of you would make a good combination. How did you two get together?



AS: Robert and I are both from the same background—we're both Irish and grew up in the same town [Bournemouth, though Summers was born in Blackpool, Lancashire], and we're both from the English rock scene. We're both progressive rock musicians and both of us have listened to a very wide, eclectic range of music. When I was 15 and playing in Bournemouth, he was doing the same thing and we knew about each other. We were both young, contending guitar players. Now we're old, contending guitar players.

I thought there was enough of a range between us to find a lot of compatible work areas. Also, we did have dissimilar approaches to music. I thought that would bring on a bit of tension so we would have to work hard to bring the whole project together. We pooled our resources and found areas that were common ground.

We didn't use any "Frippertronics," except for a little bit on one track. I was looking forward to trying that, but using tape machines isn't as exciting as two guitars playing together. We spent the first six days, seven or eight hours a day, facing each other with our guitars figuring out what to work on.

MR&M: Did you write the whole album together?

AS: Yes. We did that for the first six days and we recorded for the remaining five days or so. And that was it. We got down about twelve tracks, which I have since mixed.

MR&M: Is the record going to have a lot of guitar solos?

AS: Oh, you think it will be loaded with guitar solos. No, it's not anything like that at all. All the music is pretty and accessible, and very melodic. There is some very far out, avant-garde stuff, but it's very powerful and scary. The melodic, attractive pieces are heady sounding and intoxicating. There are no lyrics; it's either two guitars alone or two guitars with overdubs. And it's just Robert and I.

MR&M: Where did you record this?

AS: We did it in a tiny little studio in our hometown of Bournemouth. It is owned by an old friend of ours and it's a funky little studio.

MR&M: Did you both produce and engineer the record?

AS: Yeah. Well, I mixed it all down later in London, but Robert's very happy with the mixes. We'll probably finish up in London in May, but before we go back into the studio, we'll do some rehearsing at my house as well as trying to do some "live" playing.

MR&M: Did your work with someone other than Sting and Stewart Copeland give you a chance to do more of what you want to do with the guitar?

AS: I think, in all creative situations, it's good to impose a fairly tight framework on yourself. You say, "I'm going to work within these elements and I'm not going to step outside of them." By restricting yourself, it makes you try harder and dig deeper into the resources you've chosen to work with. It's like giving a painter three colors and saying, "Okay, that's it." That's exactly the reason why Robert and I are not making an indulgent guitar album. It's stupid to put in every single lick we ever knew. We were trying to make a guitar duet album that's very 80s, using guitar synthesizers and contemporary sounds.

MR&M: It's really a coincidence that you should make a solo record after you really cut loose on the guitar parts of *Ghost In The Machine*. "Demolition Man," for example...

AS: I've always been capable of playing that way, but I had to put it in the right context. In the early days of the Police, we were looking to create a new, three piece—guitar, bass, drum—style. So, there were certain things we obviously had to get rid of, like the very long, indulgent lead guitar solos which were standard rock cliches. I changed the sound of the guitar because there was a change in the bass and drums. Instead of the guitar wailing over and being supported by the bass and drums, what we had was three soloists. The guitar was very harmonic and orchestral, and that has become its role for me in the Police. I obviously do play lead solos, but only when it actually fits the song, as on "Omegaman."

MR&M: You wrote that song; how'd you come up with it?

AS: I think the song is about feeling hope in times of depression, experiencing truth and being reminded of truth as a form of a memory-like revelation. But, you need something to jar you into that state, and to me that's what the song is about.

MR&M: Since much of the Police's music borrows from reggae, did any of you actually study reggae recording techniques?

AS: Yeah, but I don't mean "study" like sitting down with a note pad and taking notes. No, we listened to reggae and got familiar with how dub is achieved. We never actually used dub techniques in the studio before, but we did on "One World (Not Three)" last time. But that was done after the track was recorded. It's just a question of listening and remixing the song through the board.

MR&M: How do the Police approach recording sessions?

AS: We usually reach a sort of tense compromise. There are some things that we will be unanimous about—we have a certain approach that we came to fairly quickly, which is once we've got the hang of a song, we put the backing track down, which is guitar, bass and drums, then we usually get all the vocals on.

Sting sings all of the vocals. His voice sounds much better when he sings with himself [overdubs, harmonies], and Sting is a brilliant harmony singer. He just knocks them out one after another. Engineers are always blown away by it. After the vocals, we decide about doing

Kim Turner, road manager and "live" sound engineer for the Police, offered Modern Recording & Music *some technical insight into working a "live" Police show.*

Modern Recording & Music: What is your musical and technical background?

Kim Turner: I left school at 15. I met the Copelands through my oldest brother—Miles managed the group my brother was in, Wishbone Ash. Eventually, Miles and Stewart Copeland managed my group—Cat Iron.

Miles had this bright idea one day of buying these big amplifiers—I think he had just seen Alice Cooper's show—and so we had these great stacks of gear. Stewart would take us down to rehearsal and say, "Alright, you guitarists have to move a little more and get out there in front when you do a solo," and other crazy stuff like that. Though I was the drummer, I was basically the only person into the whole thing. The personalities in the band were not into staging. They wanted to be serious musicians—so the group folded. From there I went off into various bands and Stewart tour-managed Joan Armatrading.

In 1977, I got depressed being in bands, you know with personality changes and all that, and I just wasn't enjoying being a musician anymore. I eventually went back to England, went to London and looked up Miles, whom I hadn't seen in a few years. I showed up at Miles' house and he and Stewart were packing Police 45s into sleeves. That was "Fall Out" on Illegal Records. A friend of Stewart's, Paul Mulligan, put up the money for the label, and they made this record. Anyway, Miles asked me what I was up to and if I wanted to be a drummer or what. I said, "I'm kind of depressed, y'know." He said he needed people to work for him and look after these punk bands because they didn't know shit. They don't know how to put out a gig on the road, and that I should get a van, P.A., lights and find the cheapest way of doing it. I said it sounded interesting and I tried it. I'd never done that before, but I knew what a musician required and I knew what had to be at the gig, so I got to his office the next day and put it all together.

So, I put these tours together and came out making a little money. Before that time you couldn't make money because you had to have big P.A.'s and record company financing to tour.

After that I managed one of Miles' bands (Cortina). Before that I had never managed a group, and I didn't think I was capable of doing it, but I said I would do it if Miles was involved. Miles at the time was handling his new record company (IRS), but he needed someone to take the group Cortina out on the road to promote records. We did an album for CBS, but unfortunately the kids, who were 16 and 17 at the time, had earned all this money and had no worries and decided they wanted to go off into the hills and rehearse for a while. All of a sudden, the band just completely changed and split up. They were kids—no responsibility at all.

At that time, the Police had been out with Henri Padovani on guitar. Then Andy joined the group and they were making their first LP. Miles had 'em in Surrey Sound Studios. They had "Roxanne," which Miles was doing for A&M; they were going to release it as a single. The single did okay, but it was at that point that Miles said the Police was going to need more management. Miles offered me 25% of the management and I became the road manager, the tour manager, the accountant. We went to America—the three guys and myself. At that time, I had been hearing the band with a few sound engineers in these small U.S. clubs while I was wandering around setting up gear and setting up with the promoter. When I couldn't hear the drums through the P.A., or if the guitar was too loud and blocking out the vocal, I'd go up and look at the board and say, "Hey, that drum sounds like a cardboard box, how do you get some tone out of that thing?" And he would show me. There was a certain sound I wanted to come out of Stewart's drum kit, and the same goes for the bass and the guitar. Every now and again the guitar would get too bassy, it wasn't clear and clean. So I started getting involved in telling these sound engineers to get a different sound. Eventually they asked me if I wanted to do it, and they showed me what to do. So I learned how a board works on the first U.S. Police tour. Then I started doing sound regularly for the Police.

When Andy got an echo unit, I thought, "Wow, that's great, I'm going to throw one of those on Sting's vocal to get that sort of reggae dub—"hello, hello, hello [he makes an echo sound]." And that was another toy.

overdubs or some other coloring of the track. The song is the thing and it should stand up without all kinds of overdubs.

MR&M: What about input?

AS: Stewart says he has complete control over rhythm. That's not true. I tell him what to play and he tells me, "Don't play that." We all tell each other what to play, and if one guy goes off and does all the work himself, the other two will stand there and produce him. It gets heavy sometimes: Stewart has to please Sting; I have to please Stewart and Sting; Sting has to please Stewart and me. That's the way it goes. And that's the way it should be because better things are reached that way. We're less likely to get away with a cliché. Everybody starts with familiar things and the others will say, "No, no, I've heard that before." Then you start working toward something that hasn't been played before and it sounds fresh and more spontaneous. You keep putting on sounds and keep working until you've got the shape of the song right with the right order of sounds. It's got to have attack the first time you hear it.

MR&M: How does the band recreate its sound "live"?

AS: I don't know. The sound takes on a different shape "live." It presents some problems, but the problems aren't so much in terms of getting the big sound—it's just getting real good at playing the number and finding all the nuances. Generally, after I record a number and go out and start playing it and playing it, I get much better at doing the song. And I think, "If only I'd done that at the time we'd recorded."

MR&M: At one time the band was doing very long jams...

AS: Well, there are still jams, but they are not as long now. The show has gotten tighter. We had to jam in the beginning because we didn't have enough material; jamming was really very strengthening for us. In general, we shape numbers a little bit and rework them until we get them sounding right "live." I think they sound better "live" than on the record because the songs have more impact and, as I said, we are usually playing them better.

MR&M: How do you feel about working with Stewart Copeland and Sting?

AS: I think—and hopefully I'm not bragging—we're all virtuosos, technically accomplished musicians. It obviously goes beyond just that to make original music. Technique is one thing, the way you apply it is something else. That's the creative part. Both Sting and Stewart are masters of their instruments, and Stewart is probably the most technically accomplished drummer I've ever played with. He's really fortunate to be able to play in medium tempo, and he does that better than when he plays hard rock. We've made a real thing out of medium tempos and his creativity has come out of that kind of playing.

It's hard to fault Sting with anything. He's a real good bass player—not the type of virtuoso bass player that Stanley Clarke is, yet brilliant at playing the right thing at the right time. He also has a fantastic independence against his vocals. I've never heard anybody sing the way he does and play so independently and solidly on the bass. Not that many people can play that far away from what they are singing; that's really difficult to do.

And then as we were touring and I'd come out to the different boards, I'd see things like Harmonizers™ and delay units and dbx and all that stuff. I was really learning as I went along.

MR&M: What do you feel your role is as a sound mixer?

KT: A lot of kids will come up to me after a show and say, "Wow, that was a great guitar solo you got." Now, in the case of Andy Summers, lots of the time I hit the EQ button and have the EQ out. So his sound is coming flat off the stage. All the effects you hear from the guitar during a Police show—his flanger, compression, echo, analog delay—that's all coming from Summers' pedal-board which he uses onstage. Andy's got a lot of toys on his pedalboard, so basically all I do on the guitar is separate his two cabinets—left and right, put them in stereo.

MR&M: What about Stewart?

KT: Stewart has the top heads on his skins quite tight. He always uses a bottom head and I always use a mic on the top of the skin that he hits. A lot of engineers say you pick up too much spill and you get too much boom and they prefer to cut a hole in the skin and put the mic up inside the drum. I think that sucks. When you listen to a drum, your ear is two feet away and that's the tone you hear. We place the mics one to two inches away from the skin. Okay, so I am picking up a little bit of cymbal, but I am getting that tone I want. If you put the mic up into the drum you've got a lot of messing around to do with EQ and things to get it to sound good. I just generally throw a mic on the top.

There are various mics you can use, too. I'm constantly trying different mics: AKG, Shure, Sennheisers...but it all comes down to the sound coming out of the drum.

MR&M: Okay, what about Sting's vocals and bass?

KT: Let's first deal with his bass. He uses two basses, which I always put a direction "injection" on. That comes straight from his guitar and before it goes into his equipment, it comes to me. I put a little compression on his bass to even it out a little—I think compression is important on a bass. The difficult part of mixing a bass is getting all the notes even. The bass itself has got to be set up right, too. I find that a little bit of compression on the bass evens it out. It doesn't make a lot of difference in a big hall if you take the compression out, no one will notice, but *I* know a little bit of compression makes a difference. Sometimes I can't even hear it [the compression] unless I turn the compression up and hear it sucking.

Sting's got an upright double bass; it looks like the sides have been sawed off. That thing is just tremendous, you get such a big bass sound out of that, because the strings are so thick. That, again, uses a little compression. I do use [boost] a lot of bass frequencies on the bass guitars—both the electric and the upright. I think a bass guitar should have bass on it. I don't like the trebly bass sound, it sounds like a lead guitar bass to me. That's okay if a particular band wants that sound, but in Sting's case, it's a bass sound, and therefore he gets those frequencies on his guitar.

Also Sting uses two sets of bass pedals. They're tied together in a box, so they both come down the same line. He also has an OB-Xa, so there are four instruments coming out of Sting. Not necessarily all at the same time.

Now Sting's vocals...I play with them and I do a lot of things with them. I use effects, the Harmonizer™, a Lexicon digital reverb, Lexicon "Prime Time." On some of the numbers I leave his voice flat and don't put any effect on it, but I like to use reverb on the vocal, I think it fattens it out a little. I also like to use echo repeat with Sting and for that I use a Delta Lab DL-4. It's a nice-looking unit, and you get up to a 520 millisecond delay on that. So, I play around with his vocals a lot.

MR&M: Do you have a preferred style of working sound?

KT: I'll tell you what I do—which I think is very important in a big auditorium—is put everything up in mono. The P.A. is in stereo, but I always think that if I've got something in stereo, the kids on stage left or those on stage right aren't going to hear everything. I always think of that poor kid out there who spent lots of money on tickets. Now and again, I'll use the stereo effect for something, but I figure in a big auditorium it's better to use mono because everybody gets to hear everything. When you've got kids behind the stage you can play the stereo a little more, but the majority of kids are in front of the stage, so that's where the emphasis is.

THE POLICE ARSENAL—EQUIPMENT LIST

Stewart Copeland: Stewart uses Tama drums, with octabands and two heads, surrounded by high-hat, mini-splash, splosh and splish, ice-bell, ride and crash cymbals. He uses a Roland Space Echo on his drums, usually putting the high-hat—and sometimes the snare and bass drums—through the unit via footswitch. Among other gadgets are a repeat/hold switch, several digital timing devices and a Syndrum set-up that enhances the bass drum.

Sting: Steinberger bass; Fender Precision bass; fretless Fender; and Ibanez. For recording, Oberheim OB-Xa's. Sting's prized custom-built doublebass has been fitted with a pick-up between the bridge and the body. Essentially an amplified instrument, the body has been streamlined. Amps: the speakers are two 12-inch folded horns in cabinets for the low midrange; six 12-inch speakers front-loaded for high midrange; two Gauss HF4000 drivers in Gauss diffraction horns for the highs. Driving all that are Crown amps. There's a PSA-II for the low bass and low midrange, a DC300 mono amp for the high midrange and a D-75 mono amp for the highs. For effects: Ashly single-input preamp (for the doublebass); an Ashly four-input preamp; two Klark-Teknik equalizers; two dbx 160 compressor/limiters; and a Roland Space Echo. Harmonizer™ and echo color his vocals onstage.

Andy Summers: Guitars (onstage and in studio) customized '53 Telecaster with a Gibson pickup and a little pre-amp on the back, and a phase switch; several Hamer guitars, including a customized instrument with a Sunburst body and two '58 Gibson PAF pickups; a Stratocaster; a Les Paul; a '58 black-inlayed Gibson ES 335. Plus Roland guitar synthesizers. Two souped-up, 100-watt Marshalls onstage. In the studio, Roland Bolt amps. His effects board is loaded with a phaser, flanger, analog delay, fuzz and a compressor. All the effects go out to a Roland Space Echo before coming back into the pedal board, which has an overall power switch and can be turned on or off while Summers is playing. (Summers can cue the effects he wants to use and then bring them all in at once by hitting that one switch.)

MR&M: You've been labeled by some of the press as an "optimistic cynic." Can you be optimistic and cynical at the same time?

AS: Oh, sure you can. I think I am. You get cynical about this business and anybody with any intelligence would. You see the business for what it is; play the game and cooperate. You know that it's real bullshit, but you know you have to do it because you believe in the music and cooperation is part of success.

Okay...you come to America and there's a huge business machine that you've got to turn on. You've got to get all the radio stations working; you've got to shake a lot of hands; you've got to be a nice guy, do a lot of interviews when you don't always feel like it. That's not the fault of record companies and journalists. I mean, they're doing their jobs and we're doing ours, but you do get sick of it. The more successful you get, the higher your asshole quotient.

I think because of the background of the three of us, we are probably in-built cynics, which is okay. I don't mind cynicism, because it's a sign of understanding reality. If you make really good music, and you believe in it, you do what's needed to get your music across. The alternative is musicians who think they are too good to fool with those things and they go and bury their heads in a hole. A lot of musicians are even more cynical [than we are], and that's really stupid as well.

MR&M: Do you think the band has "hardened" because of all the attention it's gotten?

AS: No. Try and see behind it and try and see what got us the attention. It's not because we're no good and don't have feelings, but because we do have feelings and we have the drive to succeed. Most people lack understanding and don't really think far enough. They just look at surface things and they don't really think about what's gone on.

MR&M: What are the band's plans for the future?

AS: It's very hard to say. Our plans at the moment keep changing. I think what we're ultimately going to do is wait until the end of this year to make another studio album and bring it out at a different time. I think we're getting to the point now where we don't need to rush out right on the dot one year later with another album. I think we really need to step back a bit, so when we go back into the studio, we'll be ready for it. I think probably what we'll do is take time off and do a few festivals in the summer and then record at the end of the year.

MR&M: Are you going to do more writing?

AS: Oh yeah, I've got lots of things I want to do. I want to finish the album with Robert, and obviously that takes quite a lot of commitment, energy and work. It's got to be put together with an album cover. Then I'll probably come over to the States and promote it a bit. Hopefully, the nicest thing will be for Robert and me to actually go out and play together. That is our aim, but, of course, that takes rehearsal.

MR&M: What do the other members of the Police think about that?

AS: I don't think they mind. I mean, I haven't done it yet or talked about it that much, but they both know I'm doing the album. I think it's so far away from what the Police is doing that there's no threat at all. At this point, the Police is the prime thing in our lives that fuels everything else. It is now possible for us as individuals to have satellite projects because we're all good enough to go off and do that. I also think it's very necessary—certainly for me—to get outside the group. Musicians that get locked inside a group lose sight of themselves as players. You've got to be able to function in more than one framework, and not throw all your eggs into one basket.

"THE FIRST NEW APPROACH IN OVER 10 YEARS"

Our revolutionary new Tri-Flex™ (patent applied for) speaker system represents what we feel is the best performing compact speaker system ever offered. This system is based on two accepted acoustic principles...First, that *low frequencies* below 250 Hz are essentially *nondirectional* with directional information occurring in the range from 250 Hz up. Secondly, it is an accepted fact that transducers designed for a *selected bandwidth* can be *optimized* for *maximum efficiency.*

With these factors in mind, the Tri-Flex™ was conceived and designed to produce the remarkable sound pressure level of 102 dB with 1 watt input at 1 meter. Those who have heard the system have been astounded by its compact dimensions and phenomenal performance.

The low end is provided by two 12" drivers operated into a special "slot loaded" bass enclosure, the size of the slots being specially configured to *accept the two extension speakers* **within** for transport. The outboard mid/high range cabinets consist of a specially designed 8" mid range speaker and a new type of driver/horn for the highs.

With this new and unique system, it is no longer necessary to use small, inefficient loudspeakers that have excessive power requirements, nor is it necessary to use the "crutch" of special electronic equalizers. The complete system is available at a suggested list price of $549.50 *including stands.* We feel that the Tri-Flex™ represents the very best performance available from a compact system, regardless of price.

Power Handling:	150W Continuous, 300W Program
Frequency Response:	60 Hz to 14 KHz, +-3 dB
Efficiency:	102 dB (SPL) @ 1 watt, 1 meter
Dimensions:	Bass Encl. 31 1/2" W x 17 3/4" H x 18 1/2" D
	Outboard Encl. 9 1/2" W x 14" H x 6 1/4" D

Chrome plated folding stands included

PEAVEY

PEAVEY ELECTRONICS CORPORATION, *711 A Street, Meridian, MS 39301, U.S.A.*

CIRCLE 89 ON READER SERVICE CARD

Ambient Sound

By Len Feldman

Getting Into Digital Through The Back Door

I returned from the Winter Consumer Electronics Show some weeks ago, but it's taken me that long to sort out the heaps of literature, press releases and mental impressions which manage to cram my luggage (and my head) every time I go to one of these affairs. The fact that the Winter CES is held in Las Vegas doesn't help matters either, as you might have guessed.

For this column, I tried hard to compartmentalize the products—and the impressions—in an effort to make them relevant to the reader of *MR&M*. It isn't easy. The lines between the professional recordist, the semi-pro and the dedicated hobbyist are ever more blurred. Last October, when I returned from the Tokyo Audio Fair, I was excited at the prospect of stationary-head digital (PCM) recorders that used audio-sized cassette tapes, though I was a bit worried over the fact that each of several manufacturers showing prototypes of such machines had gone off in its own direction and come up with machines that were totally incompatible with each other I'd like to think that my editorial comments had something to do with the fact that most of these prototypes never showed up here in the U.S. at the CES, but being a realist, I suspect that there are other reasons for their absence. The truth is that many major consumer audio equipment manufacturers have all they can do to interest customers in their regular line of analog audio products without having to worry about far-out PCM tape decks that can record using compact cassette packages. For while sales of pro audio equipment are holding up *fairly* well in this recessionary economy, the same cannot be said of the consumer audio segment of the market.

All of which leads me to the discussion of a pair of products, one of which is reviewed by Norman Eisenberg and myself elsewhere in this issue. I refer to the PCM Audio processor—a product which, though operating in what has to be called the "video domain," is actually an audio product. As early as the mid-1970s, not long after Sony introduced the Betamax home video recorder, its engineers (and, no doubt, engineers from other high-tech companies) realized that the information-density capacity of a video cassette tape (which, after all, had to have reasonably good response out to several Megahertz, if it was to record color video signals faithfully) made it a perfect storage medium for digital audio information as well. Digital, or PCM audio, after all, also involves bandwidths extending into the Megahertz regions.

So, almost as an experiment in leading-edge technology, Sony came up with a "big black box" PCM processor that was only slightly smaller than the proverbial breadbox, weighed more than most of us would like to lug around on field recording dates, and cost in excess of \$5000. It was called, appropriately enough, the PCM-1. You needed a video recorder to go with it, which added about another \$1000 to your digital recording equipment bill. Most of us were convinced, at the time, that Sony was simply "showing off" and was not serious about selling the PCM-1 to consumers—professional or otherwise. Well, we were wrong. It actually sold some. What was even *more* surprising was the fact that several Japanese companies then got together to create uniform "standards" for using video-type signals as the basis of a PCM digital recording system. The resulting standards were issued under the auspices of the EIAJ (Electronic Industries Association of Japan), and, because they differed in some respects from the format that had been used by Sony in its original PCM-1, that company decided to come out with a successor model, the PCM-10, which conformed in every way with the new standards. This unit was also bulky, heavy and still cost around \$5000! By this time, people were beginning to take the idea of digital recording on video cassette tape seriously, and before you could say "14-bit word rate and 44.1 kHz sampling rate" most of the major electronic firms of Japan had come up with *their* own PCM processors. But it was at around this time (1979 or 1980) that two separate approaches to PCM video-tape recording evolved. There were those who chose to follow the Sony approach, with its separate PCM processors (which required a VCR to complete the setup) and there were those, such as Technics (the Matsushita hi-fi audio division) who felt that *everything* needed to produce digital audio on video tape should be contained in one package. Technics, therefore, has recently come up with *its* version of a PCM recorder that uses video cassette tape as its storage medium. This unit, known as the SV-P100, resembles a very large, front loading cassette deck. Actually, it looks a bit like an Elcaset machine which appeared on the market a few years ago as a product that came midway between a cassette deck and an open-reel deck in terms of performance. The resemblance is, of course, only physical, for the technology incorporated into the SV-P100 is of the same order of sophistication as that found in the Sony PCM-F1 (its model number for the processor we tested for this issue of *MR&M*). Since the SV-P100 is made by a company that subscribes to the VHS video tape format (rather than Beta), it uses VHS-type tape cassettes; there's no choice in the matter. In the case of the Sony PCM-F1, while that company would, of course, like you to use one of its Beta-format video recorders, either a VHS or a Beta-type VCR can be coupled to it. The Sony machine is also a very lightweight unit that, when combined with Sony's new, lightweight SL-2000 portable Betamax, makes for a total portable recording system that weighs less than 18 pounds and can be operated in the field on batteries or from a car battery.

So, that leaves you with a clear advantage only if you are in the business (or the hobby) of recording "*live*" music or vocal performances. But if that's what you are up to, chances are that you also need to be able to edit. And, as of now, at least, that's where the problems start. As anyone who has used a video cassette recorder knows, there is no ready access to the tape inside the video cassette package. In fact, there are standard warnings about even touching the tape inside the video cassette package, for fear that oily fingers, dirt, etc. will impair video reproduction or even damage tape and/or tape heads. In the case of PCM digital audio recording things are even worse. Even if you were allowed to get at the tape, it is not possible to do "razor-blade" editing of a digitally recorded tape without introducing a loud "glitch" during playback at the very least, and a total muting of sound for an instant at the worst.

A form of electronic editing using either the dedicated Technics machine or the Sony VCR-related PCM processor is possible *if* you are willing to work with two machines (in the case of the Technics approach) or two VCRs and a single Sony PCM-F1, in the case of the PCM processor approach. That, by the way, is *one* advantage of the separate processor approach over the all-in-one machine approach. You'd only have to spend enough to buy a second VCR (perhaps another $1000 or so) rather than another $3000 for a whole new "dedicated" PCM tape deck. You'd be dealing with a rather crude form of editing, not the sophisticated, perfectly synchronized kind of editing to which you've become accustomed if you work with a good analog reel-to-reel deck as most of us do.

"It's not fair to imply...that digital mastering facilities are now available at no greater cost than a good analog mastering system..."

By this time, if you are involved in semi-pro or professional recordings, you probably have visions of scrapping your present analog portable recording equipment or even the AC-operated analog mastering equipment you use in your studio and buying either the Sony PCM-F1 or the Technics SV-P100 (or any of the several other video-format, PCM digital recording processors or complete recorders that are sure to follow from the likes of Sanyo, Toshiba, et al). Well, far be it for me to discourage you. No one was more impressed by the excellent performance of the PCM-F1, as is evidenced from my comments in the test report concerning that unit. But hold on just a moment. Let's think the thing through. What are you actually going to *record* once you plunk down the $1900 or so for the PCM-F1 (plus the $1000 or thereabouts for the matching VCR, if you don't already own one) or the roughly $3000 that Technics has projected as its suggested retail price for the SV-P100 all-in-one dedicated PCM audio recorder?

Clearly, if you're going to transcribe your existing tapes to digital format, there's really not much point in dubbing a recording that exhibits fairly high distortion levels, modulation noise, background noise and wow-and-flutter onto a medium that is so much better in its capabilities than the program material being transcribed. The same holds true if your plan is to transcribe your favorite discs onto digital video cassette tapes. The only advantage in transcribing discs to digital form is that if you then play the digital-tape versions, you won't degrade or wear out the originals. Then again, that would also be true if you transcribed your discs to analog tape.

It seems to me, therefore, that the idea of PCM digital recording on video tape cassettes is not going to catch on in a serious way until the manufacturers of these processors or complete decks also give us true electronic editors (such as those which are now showing up in high-end professional equipment circles for use with stationary-head, multiple-track digital mastering systems) with which we can perform all of the audio editing tricks that we need. Judging by the costs of professional digital editors, such editing facilities are not going to be cheap, even if they are "toned down" for semi-pro, small studio or serious audiophile use. On that basis, amazing as the new PCM processors are, it's really not fair for their producers to imply that digital mastering facilities are now available at no greater cost than a good analog open-reel mastering system. At least that's how I see the current status of things. Of course, Sony, or any one of the other companies deeply involved in digital audio could surprise me at any moment. After all, I never expected to see a PCM processor selling for roughly one third of previous models barely a year and a half after its predecessor was first introduced either!

Profile: MOE BANDY

By Rob Patterson

If perhaps "success story" seems an exhausted phrase, one probably has yet to meet country singer Moe Bandy, rightly dubbed by the Texas Legislature as "The King of Honky-Tonk Music." But less than ten years ago, Bandy was serving his musical serfdom in San Antonio's night-spots. He probably didn't even feel like the king of his castle at home; after all, the furniture was all mortgaged to cut another single: "I Just Started Hatin' Cheatin' Songs Today."

With that classic slice of barroom blues, released on a small label in Atlanta, Bandy topped the country charts in 1973. Everything that's happened since—twenty-eight Top-10 country records to date—has proven that Bandy's gamble was a wise investment. Yet for all his newfound fame and acclaim, Moe Bandy remains very much as he was, living in San Antonio and hanging around with the same friends he had back in the days when he was a sheet-metal worker.

Yet his goal to be a country singer was not far-fetched. His mother's father had worked on the railroads in Mississippi with Jimmie Rodgers. Bandy's father, like himself, was a sheet-metal worker with a country band on the side. As a local act, Bandy opened for Loretta Lynn, Charlie Pride and Webb Pierce, and got one of the biggest thrills of his life when he backed Bob Wills.

Bandy also cut some singles locally, but it wasn't until he met another native San Antonian, Ray Baker, a Nashville music publisher and producer, that he threw his Stetson into the national arena. Baker was in town for a hunting trip. Bandy heard about him through one of Baker's relatives and brought him a tape. Baker heard a surprisingly good voice, and offered to produce for Bandy gratis, *if Bandy paid the session costs. Today Baker is still Bandy's producer and manager.*

Baker was obviously the right counterpart for Bandy, having himself played in country bands while in high school around San Antonio. Through his work on-air radio stations in Uvalde, Texas (KVOU) and San Antonio (KMAC), Baker met country stars like Jim Reeves and eventually landed a job on Nashville's WMAK. Later becoming the head of Reeves' publishing company, Baker discovered country chestnuts like "As Usual" for Brenda Lee and "The Race Is On" for George Jones. A year after Reeves died in 1964, Baker started his own company, Blue Crest Music, and signed up writers like Dallas Frazier, author of this year's CMA "Song Of The Year": "Elvira."

Baker has also produced Connie Smith, David Houston, Sammi Smith and ex-Texas Playboy Leon Rausch, as well as the hit-making duo of Moe Bandy and Joe Stampley, whose two albums cast them as the rollicking Heckyl and Jeckyl of honky-tonkin', and advanced both artists' careers with hits like "Good Old Boys."

On songs like "Barstool Mountain," "Cheatin' Situation" and a host of others, Moe Bandy has displayed a much-lauded purity in his approach to country music. Among the tributes he's earned is a song called "Bandy, The Rodeo Clown" by Lefty Frizzell, inspired by Bandy's three or four years on the rodeo circuit (where his brother has been a Top-10 bullrider for almost a decade). Bandy also co-owns a successful talent agency—Encore Talent—based in his hometown. Bandy's latest album, Rodeo Romeo*—accurately links the singer's rodeo roots with his position as a master of the cheating song. In New York for a stand at the Sundown club (recorded for DIR Broadcasting's Silver Eagle radio show), Bandy spoke with* Modern Recording & Music *about making his "hard country" records, while Ray Baker explained how his end of their fruitful partnership works.*

Modern Recording & Music: You're known for a pretty straight, traditional country sound. How have advances in studio technology affected that kind of music?

Moe Bandy: I think it's been great for it, because we can go in there and get that straight, pure country music sound, and have the real good quality too. To me that's the only difference between what we're doing now and the country that was recorded in the Fifties and early Sixties. We're getting such a great stereo sound, and we can go in and do overdubs because we have so many tracks. There're just so many things we can do now.

MR&M: Do you think that studio technology might also have contributed to some of the overproduced country records we've heard lately? It seems that there's a real special quality to some of those old country discs—say, Hank Williams records—that sometimes gets lost in the studio today.

MB: Well, we have the opportunity with so many tracks that you can go in and put so much on that a lot of people have dressed it up too much. But we try to keep that same old sound, a similar sound, but with better quality. We also have more musicians and possibly better musicians today. It's just great to be able to do that.

I think if Hank Williams was alive today, and going in and recording on today's equipment, it'd sound just great!

MR&M: Nashville has almost too many good studios. Which ones do you prefer?

MB: We used to use Columbia B, which we still use some. But in the last few years we've moved over to Jack Clement [now known as Sound Emporium, although Nashville regulars still call it Clement Studios, in deference, one supposes, to Cowboy Jack Clement himself]. The one reason I like the studio we use there is that I like a small studio, because I like to be real tight with the musicians. I like to be right in the middle of them so I can hear really well everything that they are doing.

MR&M: I assume you like to cut "live," singing with the musicians?

MB: Yeah, I try to as much as possible. We overdub some, but we've always felt that we get the best possible feel if I can sing with the band right there. You can see the pickers kind of groovin' a little bit, and it's just a better overall feel.

MR&M: Which musicians do you generally use?

MB: We use generally the same people: Weldon Myrick on steel, whom we use on almost everything; Bob Moore on bass, who is just super (we use Henry Strzelecki some on bass, if we can't get Bob); Pig Robbins on piano; Charlie McCoy on harp; Ray Edenton and Leo Brooks on rhythm guitars; and Johnny Gimble on fiddle. And there're so many more good musicians...You usually can't get all the guys you want at the same time, but we try to when we can. We also use the Jordanaires on vocals.

MR&M: Do you get involved in the sound and arrangements?

MB: Ray Baker, who I think is a fantastic producer, is responsible for that. I really don't get into that too much. I concentrate on singing and let him handle the music to it. My feeling is that I need to go in there and really feel the song and put my best into singing it. Of course I have a suggestion now and again, but Ray has created a Moe Bandy sound with the musicians, and that's fantastic, because if you've got your own sound, that's what it's all about.

MR&M: Do you pay attention to how your vocals are recorded?

MB: I like the voice to be up enough, but again, Ray handles that very well. Ray and I work together so well it's unbelievable. He almost reads my mind sometimes. He'll do something where, if I was going to do it, I'd do it that way. We never have any problem mixing or anything. Ray will send me a mix and ask me if I like it, and I hardly ever hear anything we have to change.

MR&M: Do you do many "live" recordings, and if you do, have you any special techniques for "live" recording?

MB: Yeah, we recently did a show for the Silver Eagle, a syndicated radio show—I've done that one twice. I've also done the show "live" from Gilley's...I've done a bunch of them. I just try to do my regular show and let them worry about recording it. I used to be concerned: "Well, they're recording tonight; I'd better do something different." But when you do that you just mess up your whole show.

MR&M: I notice you don't amplify your acoustic guitar. It seems that a lot of country singers do that. Why is that?

MB: Well, I play an Alvarez guitar, and...they are making me one with a pickup. I don't usually amplify it. When I began my career, I was always playing the guitar, but now we have such good

Texas State Senator Glenn Kothmann dubbing Bandy "The King of Honky Tonk Music."

bands, they can do without my rhythm guitar. So it's mainly just the habit of being used to holding it and playing it.

A lot of songs I'll kick off cold, then hit the chord, then the band comes in. And for a long time, when I was without my guitar, I didn't know what to do, how to handle it. But now I'm comfortable enough on stage that I can do part of my show without the guitar.

* * *

MR&M: Where do you feel you get the best sound for your "live" shows?

MB: I think the best sound we usually get is in old theaters, which we do a lot of. A lot of them were built for acoustic instruments, and you get some great sounds. All you have to do is turn the volume *down* some, and you can get incredible sound. I will say that one club that really surprised me that it had great sound was Billy Bob's, which just opened in Fort Worth, Texas. It is the biggest nightclub in the world. It has just great sound.

MR&M: What was it like at the semi-professional recording sessions you did in San Antonio before you met Ray Baker?

MB: Well, we didn't have the producers, didn't have the musicians—I was using the guys in my local group at the time, who were good, but not real studio players—and we didn't have the engineers or the equipment. The mixes—whoever mixed them, I don't even know—were terrible. So it was no comparison to Nashville, which is really the best place to record country. Of course, they've been working a long time to get it that way.

As a matter of fact, the guy...the crook who had me in San Antonio re-released all my old stuff. And it's terrible, really bad stuff.

MR&M: When you were recording with Joe Stampley, did you do anything different?

MB: The only really different thing Joe and I did was more uptempo songs. And we did a bit of acting silly, playing around. My records are more serious, so I get into it in a different way. It was great doing it and it really helped both of our careers, but neither of us wanted to be half an act.

I wouldn't mind going in and cutting some things as a duo. The other day I was talking with George Jones about cutting some things together. But mostly I'm going to concentrate on my career, and I'm really excited about my new album, *Rodeo Romeo*.

MR&M: It seems country music is going back to the "hard country" style you play. Why do you think that is?

MB: Well, our music has gone so far in so many ways that a lot of people are now going back and doing it the way we used to. George Jones, for one, said that two years ago: "I wanna go back and start cuttin' *country*!" There are several country musicians that have done that. It's just really paying off for them, too. George has had the best years he's ever had in the last two years.

SELECTED DISCOGRAPHY

I Just Started Hatin' Cheatin Songs Today	(1973)	GRC
It Was Always So Easy To Find An Unhappy Woman	(1974)	GRC
Bandy The Rodeo Clown	(1975)	GRC
Hank Williams You Wrote My Life	(1976)	Columbia
Here I Am, I'm Drunk Again	(1976)	Columbia
I'm Sorry For You My Friend	(1977)	Columbia
The Best of Moe Bandy, Vol. I	(1977)	Columbia
Cowboys Ain't Supposed To Cry	(1977)	Columbia
Soft Lights And Hard Country Music	(1978)	Columbia
Love Is What Life's All About	(1978)	Columbia
It's A Cheating Situation	(1979)	Columbia
One Of A Kind	(1979)	Columbia
Just Good Ol' Boys (with Joe Stampley)	(1979)	Columbia
The Champ	(1980)	Columbia
Following The Feeling	(1980)	Columbia
Hey Joe—Hey Moe (with Joe Stampley)	(1981)	Columbia
Rodeo Romeo	(1981)	Columbia

A Conversation with Ray Baker

Modern Recording & Music: How do you think the advances in studio technology have affected the recording of traditional country music?

Ray Baker: I think that the technology in studios now is so advanced from what it was 20 or 30 years ago that now you can capture everything that's there. Whatever the musicians and singer do, you've got it. After they're gone, it's a question of being able to do any number of things through the magic of EQ, echo, etc.

MR&M: Would you agree that there's a certain something on those old country discs, like Hank Williams' records, that sometimes gets lost today?

RB: A lot of those old records were cut direct-to-disc. The quality of direct-to-disc is always much better, but back then they didn't have the mics we have today. I think the quality is much better today than it was back then, although the music was different. A lot of Hank Williams' records don't have drums. Now if you took the drums off of a record today, that's six tracks you're automatically cutting out. But of course, you can really get any type of sounds you want today.

MR&M: You seem to have fashioned what's called a "hard country" sound for Moe Bandy. How do you do it?

RB: The reason they call it hard country is more the material we record then the actual sound of the records. It's the old fashioned type songs, yet there's an obvious demand for that now, or Moe wouldn't be selling the numbers that he does. But we just record with a basic rhythm track—two rhythm guitars—regular string and a high string, which gives me a nice balance; piano; steel guitar; drums; lead guitar; harmonica; and fiddle, usually. Then I'll overdub the voices after that.

MR&M: What's your technique for mixing?

RB: Well, once I get everybody on the multi-track, the engineer and I will sit in the studio together and try several different mixes, listen, and then later I may go back and listen again. I may do the same song two or three times.

One of the the things that has really helped us lately is using the computer board over at CBS Studios for mixing. It really is a plus, because it saves time. I'll do a mix that feels really good in the studio; then I'll play it in the studio. I'll play it in my office—where I have these big Klipsch speakers—and then I'll play it in my car, where I have a really good cassette system. I'll get a pretty good idea of what I'm looking for, and if I feel that, say, the guitar break is a little too hot in the turnaround, but everything else is fine, I can go back and plug the computer in and make that adjustment. So often before, I'd go in to get a different feel, and then I'd correct what I wanted to, but I'd have to sacrifice something else. So the computer gives us an edge in that respect.

MR&M: What about miking?

RB: The only thing I go direct on, which is in an overdub situation, is like a Fender Rhodes—electric piano. Everything else is cut with mics up against the amps for the electric instruments. For the [acoustic] rhythm guitars they have these small mics that fit inside—right up under the strings by the sound hole. I don't even know what they are, these small mics like you put on your lapel. They really pick up the sound great with almost no leakage. The other thing I really listen to is how the drums are miked, so I can get plenty of separation when I need it, and get the presence I need on the toms, snare and bass.

As a matter of fact, some of the best records we've cut have a little leakage. As long as nobody makes mistakes, the leakage doesn't bother anything. The one studio that brings that point to bear more than any other is the old Columbia B, where they haven't changed the room in years, same piano, same funky floor and everything else. The room has a lot of leakage, but it has such presence and such a lovely feel to it. Some of the best records that have ever been made have been cut in that studio. As long as you know that you've got a track that's really there, and you know you're going to have to overdub the singer, I'll just get him to sing it several times, and then I'll have him back off and get a take. That way I don't have any vocal leakage in the mics, and I still have that great "live" sound.

MR&M: What special attention is paid to Moe's voice?

RB: Now I'm not all that engineering oriented, so I can't even tell you what mic we use. We just make sure that we get the voice down really well. Then, with EQ, there are so many things you can do. I usually add a little mid-range and a little top to his voice, depending on the song. If it's a ballad, a lot of times I'll use a natural echo, then I'll run a little slapback on his voice to give it some depth. I just try to capture the quality of his voice that's there and not disguise it too much.

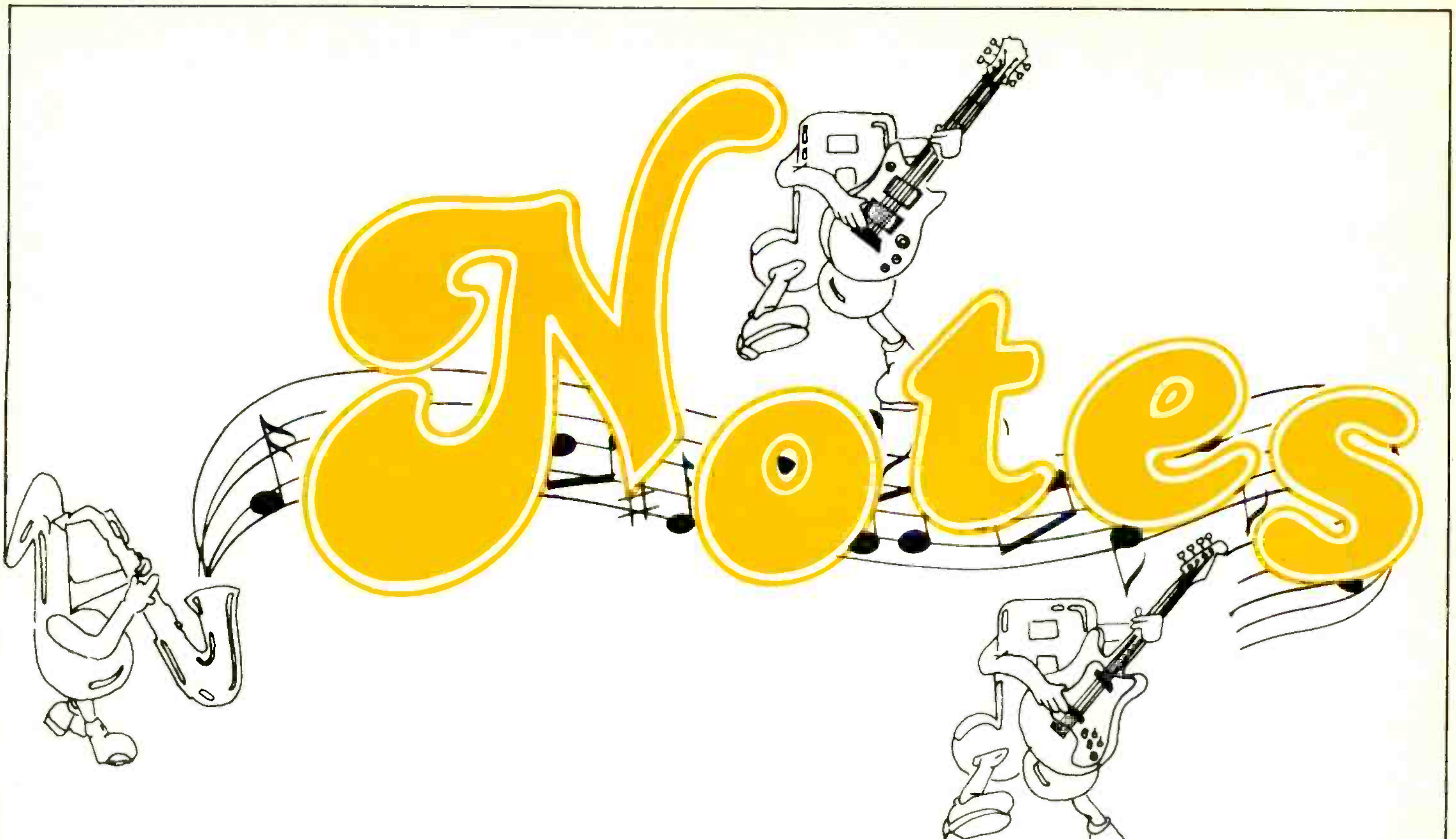
MR&M: Do you approach recording Moe Bandy and Joe Stampley as a duo in any different ways?

RB: Well, I played it by ear in one respect. If both of them were in there singing and grooving, and I knew that I wasn't going to have to overdub anything, I'd just let it go. But if I felt that I was going to have to get them to re-sing it, I'd let them work until they got the feel of the track, then have them back off and do the track. Then we'd overdub the singing later. We did it about half one way—"live"—and half the other way—vocals overdubbed later.

MR&M: What first impressed you about Moe Bandy?

RB: I was down in San Antonio on a hunting trip, and he brought me some tapes of some records he had made locally. When I got back to Nashville I listened, and really, I was surprised he sang so well. Really, there're so many guys that sort of think they can sing, but can't. But there was something in Moe's voice. I called him, and though I told him that he was a real good singer, I also told him that this is a tough business; you have a very slim chance. I didn't want to build his hopes up: you come to Nashville, make a record and become a star. I gave him all the honest facts I could, then said that I'd produce a session for free if he would pay the musicians. It had to be his venture.

The best thing about working with Moe is that he puts his complete faith in me. He allows me to choose material... work up arrangements. He trusts my judgements in putting out songs. We work together, but he really has trust and faith in me, which if anything, tends to make me more responsible.



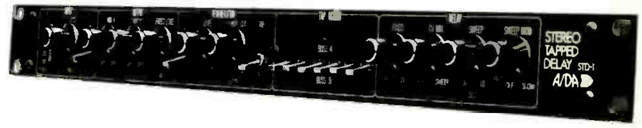
A/DA STD-1 Stereo Tapped Delay

By Craig Anderton

I recently received a call from A/DA's [Analog/Digital Associates, Inc.] marketing manager, saying that *Modern Recording & Music* had selected the STD-1 as my next product for review in "Notes," and could I please give him my address so that A/DA could ship an evaluation unit to me. Little did he know that upon first laying eyes on the STD-1 a short time ago, I had immediately placed an order for the unit (I got one of the first ones manufactured), and had already been using it extensively in my studio. Now, you might ask, what would make a do-it-yourself enthusiast like myself willing to plunk down hard-earned cash for a *manufactured* product? Read on for the answer.

WHAT is IT? First, let's discuss what it isn't. The STD-1 is like no other delay you've ever seen, for reasons which will become clear as we go along. It is *not* an echo unit, and while it flanges, it's not really a flanger. It can't give delays longer than 55 milliseconds, and what's more, if you're only interested in mono you'll be wasting your money on an STD-1; this is truly a *stereo* device (and I don't mean synthesized stereo) that can only be used to its full potential in a stereo setup. The STD-1 is superficially similar to other delay lines—it has delay time controls, modulation controls, and so on—but inside its 1¾" rack mount metal heart, the STD-1 is something altogether different.

So what is it? The STD-1 is based around the MN3011 delay line, a chip made by Matsushita. I have followed the story of this chip for quite some time, because I had been chomping at the bit to use it in a do-it-yourself project. From what I understand, the MN3011 was intended to be a solid-state reverb chip. Unlike most delay lines, which have a single



input and output, the MN3011 is a 3000+ bucket brigade delay line with six output taps spaced at *non-harmonic* intervals. Matsushita's concept was that you would sum these six outputs together, with the longest delays receiving the greatest amount of low-pass filtering (a convenient way to bypass noise problems, by the way), and feed a portion of the output back to the input. The result would be a sound that approximated room reverb. The key to this was the non-harmonic tap spacing, because for reverb you need to simulate the random reflections that occur in a room. As each tap feeds back into the input, it becomes an output at each of the six taps, which then feed back into the input, and so on. If you're used to what happens with standard analog delay lines when you turn up the regeneration, you've probably noted a repetitive, "machine-gun"-like reverb effect. This is because the delay time and feedback time are always harmonically related. By making all taps non-harmonically related, you never build up the electronic equivalent of "standing waves" in the chip itself, which yields a reverb sound that is random-sounding and more closely approximates room reverb characteristics.

So why isn't the world being flooded with analog delay reverb using the MN3011? I don't have hard facts, but the rumor goes as follows: The MN3011, already an expensive chip, was proving difficult to economically manufacture. This drove the price up even further, and with little hope of having a profitable product, Matsushita eventually put the MN3011 into corporate limbo.

Before this happened, I had managed to purchase two chips for my own use. I wasn't too impressed by the reverb sound; the "Hot Springs," *[see MR&M's October 1980 issue]* for example, costs less and sounds better (I'm not just being biased; I doubt if you could find anyone who would dispute that statement), but I was mighty intrigued by what happened when you fooled around with putting the various taps into different stereo channels. It seemed like this was a chip with lots of potential. Unfortunately, taming it took some work, and since my efforts could never turn into an article (why publish something where no one can get the parts?), I wistfully put the MN3011s aside and hoped that one day Matsushita (or someone else) would figure out how to make the things in big quantities at small prices.

In the meantime, I heard that A/DA had brought up virtually the entire existing supply of MN3011s. I figured that they were going to market some kind of solid-state reverb, but like me, their engineers were seduced by all those taps and the possibilities they offered for stereo flanging and chorusing. Some time later, I was talking to a friend who happened to be writing the manual for the STD-1. I pumped him for information, found out what the thing did, realized that A/DA was thinking about the chip the same way that I had thought about the chip, and shortly thereafter placed my order. I was a little concerned about committing myself to a product that used an "obsolete" part, but was assured that A/DA had bought enough MN3011s to take care of servicing should the need arise.

Hmmm...I still haven't explained what the thing is. Well, I thought that you might find the above "inside story" interesting (and if the editors don't find it interesting, they'll cut it out anyway so I have nothing to worry about). On with the rest of the review.

Figure 1 shows the STD-1's block diagram. There are two STD-1 models, the "instrument" and "studio" versions. Both versions have a switchable input; the "instrument" model switches between 40 k Ohms or 1 MegOhm unbalanced, while the "studio" model switches between 40 k Ohms unbalanced or 600 Ohms balanced with XLR connector. The input stage includes a *level* control and associated eight-stage LED headroom indicator. The headroom indicator shows exactly how much headroom you have left before clipping occurs, which makes level setting a snap. There's also an in/out switch, with associated LED, for cutting the effect in and out.

From the input, the signal progresses through the MN3011. Note the six taps; each tap connects to a three-position switch. In the center position the tap is off, while the other two positions send the tapped sound to either the A or B output buss. You can assign all six taps to either channel, however, you cannot send a tap to both channels. Because of the way in which the STD-1 creates stereo effects, this does not turn out to be a problem.

The busses then go to individual balance (*Mix A* and *Mix B*) controls, which let you select the proportion of straight and delayed sounds in channels A and B. There is an *output level* control common to both channels. The "instrument" model outputs (one for each channel) are unbalanced and can drive 600 Ohms, while the "studio" model outputs have

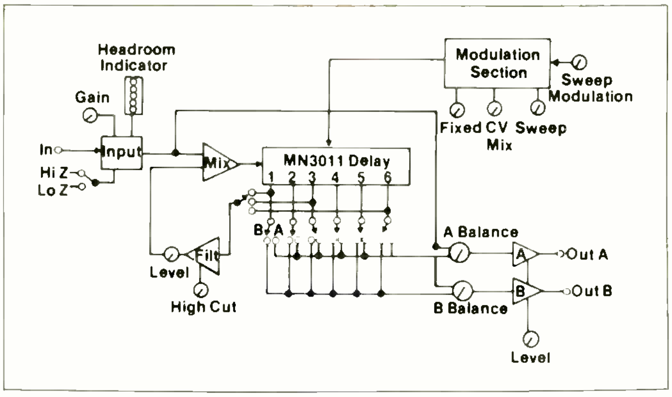


Fig.1

quarter-inch phone jack unbalanced outputs and 600 Ohms balanced, transformer coupled, XLR connectors.

The regeneration section taps off either tap 1, 3 or 6, and also includes, along with a *regeneration level* control, a *hi-cut* control with an adjustable high cut from 12 kHz to 800 Hz. This lets you simulate the loss of highs that occurs in acoustic environments with repeated echoes.

That covers the audio part of things, except that there are also two noise reduction sections based on the NE572 compander (an improved NE570). One noise reduction system is for the delay line, while the other is for the regeneration section. The use of two noise reduction systems improves the quality of regenerated sound, and helps contribute to the STD-1's exceptionally low noise (and unlike many companding delay systems, modulation noise—where the hiss comes up with the signal—is virtually non-existent with the STD-1).

The modulation section is fairly straightforward. The *fixed* control sets the initial delay time over a 5:1 range. Here are the delays available at minimum and maximum delay time settings for various taps:

Tap 1	**1.3 ms to 6.5 ms**
Tap 2	**2.2 ms to 11 ms**
Tap 3	**4.6 ms to 20 ms**
Tap 4	**5.8 ms to 29 ms**
Tap 5	**8.3 ms to 46.5 ms**
Tap 6	**11.1 ms to 55.5 ms**

Note that with the top three taps, you've got the flanging range covered, while taps 3, 4 and 5 cover the chorusing range. Taps 4, 5 and 6 are suitable for slapback and other tight echoes.

For automatic sweep effects, there's a *sweep* rate control that covers the range of 0.04 Hz (25 seconds/cycle) to 10 Hz (0.1 seconds/cycle); the delay *C.V. mix* control superimposes the desired amount of sweep onto the initial delay setting. There's also an enigmatic control called delay *sweep modulation*. This superimposes a variable sweep pattern onto the regular sweep, thus producing a randomizing effect that breaks up the monotony I find objectionable in traditional periodic LFOs (low-frequency oscillators). This effect may be disabled by turning the sweep mod control fully counterclockwise.

That's it for controls, but there are also jacks on the back for the wire freaks in the crowd. One jack allows 0 to +5 V voltage control of the time delay, while another allows 0 to +5 V voltage control of the sweep rate. Both stereo jacks are A/DA pedal-compatible, but by adding a suitable adapter cord, you can use conventional volume pedals and industry-standard synthesizer modules. There's also a CV-out jack for slaving two units together (or you can use it to send the STD-1's "randomized" control voltage to other voltage-controllable units lacking this feature), and there's a stereo jack that accepts a standard guitar amp dual footswitch. Using this allows you to cut both the delay effect and regeneration in or out. Finally, there is an AC on-off switch, and fuse post, on the back of the unit.

PRE-FLIGHT for the STD-1: You can plug an instrument such as a guitar directly into the STD-1 input. However, with my guitar there was *just* enough gain at maximum, so if you're a light picker or use low-output pickups you would probably want to add some kind of preamp prior to the STD-1. Guitars with hot-rod pickups or active electronics should pose no problem.

You should definitely go for stereo when hooking up the STD-1 outputs, either through a mixing console or by using two amps. You can use two channels of a single guitar amp, but this really doesn't do justice to the stereo capabilities. Set the input control so that the green LED right below the red LED on the headroom indicator lights only on the peaks of your playing, then use the output level control to match the levels of the delayed and straight sounds. Now it's time to fool around with the other controls.

APPLYING the STD-1: With all these controls and switches, there are so many available applications we can't possibly cover them all. So, we'll concentrate on some of the more popular variations.

First, you have to re-orient your thinking of what a delay line can do. Most people think a delay serves a single specific function or combination of functions, such as flanging, chorusing, slapback echo or long echo. However, there is a whole class of effects that fall under the category of "ambience generation." Ambience effects are somewhat more diffuse and subtle than what we normally associate with delay effects. In other words, while the STD-1 will give flanging, it is of a different nature from traditional flanging.

As a flanger, the STD-1 is in some respects less than adequate, but in other respects downright spectacular. The sweep range (5 to 1) isn't all that great; you can't, for example, sweep continuously from 500 microseconds to 25 milliseconds, which means you can't obtain real wide range effects when you're only listening to one tap. But when you switch in two taps and put them in stereo, the story changes. You can be sweeping from 1.3 to 6.5 ms. in one channel and 2.2 to 11 ms. or 4.6 to 20 ms. in the other channel. The effect is dramatic and liquid-sounding. While the polarity of the effect can't be altered (which I feel is important for a flanger), the regeneration is effective and well-behaved. Although you generally tend to feed back one of the taps you're listening to, you can achieve other interesting effects by feeding back a tap you're *not* listening to.

Now is a good time to note that the way the STD-1 creates stereo outputs is quite different from the synthesized stereo outputs available on most effects. With synthesized stereo, when both channels are played back in mono the effect (flanging, chorusing or whatever) cancels. The STD-1 actually creates two truly separate signals, not just a mirror image of the same signal, so any effect is preserved when playing back processed material in mono. If you're playing "live" this isn't such an important feature, but for anything that gets played over the radio true stereo is vitally important.

As a chorusing unit, the STD-1 is *breathtaking*. Putting different taps in different channels and lightly sweeping the modulation makes for incredibly lush and full-sounding chorus effects. The sweep modulation is a nice touch, since it helps break up the sweep pattern and produces a closer-to-random chorus effect. For mixdown processing, one of the best favors you can do for any instrument is to run it through an STD-1 and spread it in stereo. The thickening and chorusing effects on voice are amazing, in fact, just about everything I put through the STD-1 sounded fuller, bigger and acquired well-defined stereo effects. By the way, if you want to turn one channel of reverb into stereo, I can't think of a better way to do it than processing through the STD-1.

For slapback echo, the STD-1 again beats anything I've heard. Having slightly different echo times in the two channels fools the ear into placing the sound closer to one ear than to the other, almost as if you could feel the presence of walls in a room. One weakness of the STD-1 is that regeneration can only be picked up from one tap, which means that you get a repetitive echo sound when you turn up regeneration—the exact opposite of what the chip was trying to accomplish in the first place. However, I suppose you can't have everything and considering how strong the slapback sound is, with or without regeneration, I shouldn't complain. The regeneration filter also comes in handy with slapback echo, when trying to define room "sizes" and density of the "room."

"The STD-1 is one of the more creatively designed special effects units I've had the pleasure to use."

However, many of the STD-1's most interesting and novel effects don't fall under traditional categories such as flanging or slapback. With certain combinations of tap, regeneration and modulation settings, you can not only shift the perceived center of the sound from left to right, but create unusual psycho-acoustic illusions such as placing the instrument further "in front" or further "behind" the other instruments. The first time I heard this type of effect was on Carlos' *Sonic Seasonings* album, where some sounds almost seemed to come from *behind* the listener. Apparently, this is a phase-related phenomenon and with all those non-harmonic delays going on in the STD-1, the opportunities for strange phase relationships are many.

I should also say something about the quality of the sound: it's absolutely amazing, with a high end that I can only characterize as the "Sony sound"—clean, bright, with a glossy kind of sheen (like a Walkman, but without the hiss). Having gotten used to the high frequency dulling and fuzziness usually associated with analog delay lines, the clarity and precision of the STD-1's high end was most unexpected and dramatic. One reason for the exceptional quality of sound is that A/DA doesn't "push" the chip at all. The MN3011 could easily be taken out to a 100 ms. or greater delay, but A/DA wisely avoided this and operated the MN3011 at clock speeds that virtually guarantee reasonable fidelity. When you add in the compansion, pre-emphasis/de-emphasis, filtering and other circuit tricks that enhance the signal going through the delay line, the sound is astonishingly clean and has presence that just won't quit. Another contributing factor in the cleanliness of sound is the circuit board, which is one of the better laid-out circuit boards I've seen in an effect: extensive use of ground shields and proper layout of ground lines helps to keep things quiet. A/DA has taken a very conservative design approach to the STD-1, being content to maximize usefulness in a small delay range rather than try to be all things to all people and offer short, medium and long delays in one unit—a far more difficult task.

OVERALL EVALUATION: Despite all this glowing praise, there were a few things I didn't like. One thing, for example, was the limited sweep range. One of the great advantages of analog delay lines is that you can sweep them over a wide range, and most units have at least a 10:1 sweep range. With a 5:1 sweep range, the STD-1 doesn't really qualify for wide-range flanging, although for chorusing and slapback a 5:1 range is more than adequate. So, the STD-1 is more suited to "ambience" flanging than "classical" flanging.

Also, with several taps selected, adding regeneration can tend to make the sound somewhat "boxy." This occurs because you have lots of coincident signal peaks going on at once, thus creating occasional peaks that are very strong. It occurred to me that perhaps putting a limiter in front of the regeneration filter might solve this, although I should emphasize that this is only a problem with fairly extreme settings.

The controls are logically laid out, and while the parts are generally of high quality, I'm not so sure about the pots (that's a feeling I have about virtually all effects these days). However, maybe my prejudices are showing, or they make pots better than they used to, because most of the effects I own with apparently inexpensive pots have served me well, and continuously, for years.

But these are minor complaints. The bottom line for me on any special effect is the "inspiration factor"; in other words, when I plug into the thing, do I get a big grin on my face because the sound is so great, and promptly get lost in my playing for the next half hour? With the STD-1, this never fails to happen. My Telecaster and GR-300 [Roland guitar synthesizer] just plain sing through the thing, my voice sounds rich and full and synthesized bass spread in stereo... well, you're just going to have to hear it for yourself.

If you could only afford one delay unit, it probably wouldn't be the STD-1. Instead, you'd probably choose something general purpose with a wide sweep range and the ability to delay from 1 ms. to at least 1/3 second (preferably 1/2 second). However, if you think that the standard time delay sounds of flanging, chorusing and echo are the only game in town, the STD-1 will change—if not blow—your mind. The psycho-acoustic effects are indeed novel, and in my opinion, the STD-1 is one of the more creatively designed special effects units I've had the pleasure to use. Some people might not appreciate its subtleties; but other people, like me, will become intrigued enough to want to take the STD-1 to the limit. It's only then that you realize how distant those limits are.

I know that in these pages I've criticized some products for lack of imagination, but make no mistake, in this world of me-too products, the STD-1 is an original. Best of all, it does its job superbly—and it doesn't try to do anything that it can't do excellently.

MODERN RECORDING & MUSIC

LAB REPORT

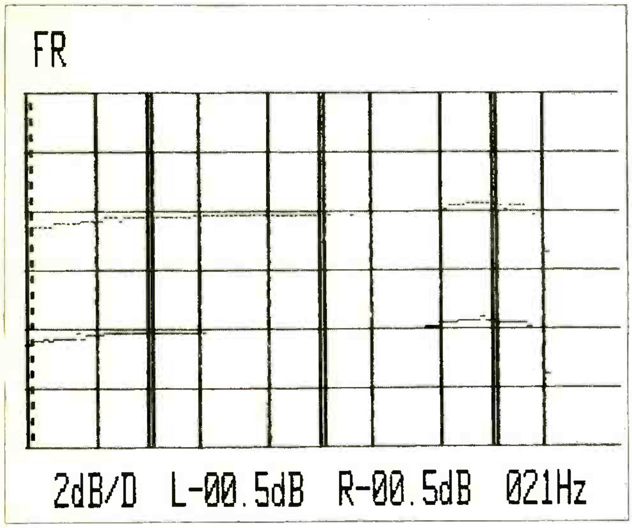
NORMAN EISENBERG AND LEN FELDMAN

Sony PCM-F1 Digital Audio Processor

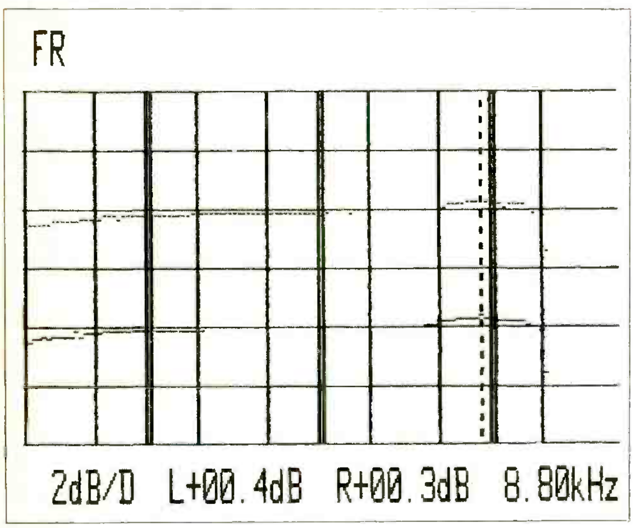
General Description: Sony's PCM-F1 (the letters stand for pulse code modulation) is a digital audio processor intended for use with video recorders of either the Beta or VHS format. The PCM-F1 converts analog audio signals into digital data for recording onto a video cassette, and it converts this same digital data into analog signals suitable for playback through a normal audio system.

Compared to the first PCM units that appeared about two or three years ago, the new Sony device is relatively compact, lightweight and significantly lower-priced. It actually can be used as a portable. The device may be powered by its own slip-in battery; or from AC lines via a power adaptor; or from vehicular 12-volt DC systems via another adaptor. The AC power modular also serves as a battery recharger if desired. The PCM unit itself is fitted with attachments for an optional carrying strap; the AC adaptor has a fold-down handle.

At the extreme left of the front panel are the device's power off/on switch; a stereo headphone level control (five steps marked for 0, −6, −12, −18, −24 dB); and the headphone jack. To the right is a multicolor display panel containing twin bar-graph fluorescent meters for left and right channels. Calibration here runs from "infinity" through −50 to "zero" dB, with the legend "OVER" indicated in the red meter section at the right. Percentage markings under the dB scale run from 0.3 to 100, the latter figure corresponding to "zero dB." Also on the panel are illuminated indicators for muting, emphasis, copy prohibiting and tracking. The muting light will come on in the event the associated VCR is mistracking or skewing the tape to indicate dropout portions during



(A)



(B)

Fig. 1: Sony PCM-F1: Deviation from perfectly flat response over the entire audio spectrum from 20 Hz to 20 kHz was only −0.5 dB (at 21 Hz) (A); and +0.3 or 0.4 dB at 8 kHz (B). Tests were made at 0 dB record level.

playback. The copy-prohibiting light shows that a given tape has been "copy-guard" coded so that it cannot be duplicated. The tracking indicator lights up when you adjust the tracking control on the associated VCR. The emphasis circuit comes on automatically except when "non-emphasized" tapes are used.

To the right of the display panel are several operating controls. These include left- and right-channel recording level knobs; a record-mute button; meter-select buttons for tracking adjustment and for signal level, as well as for

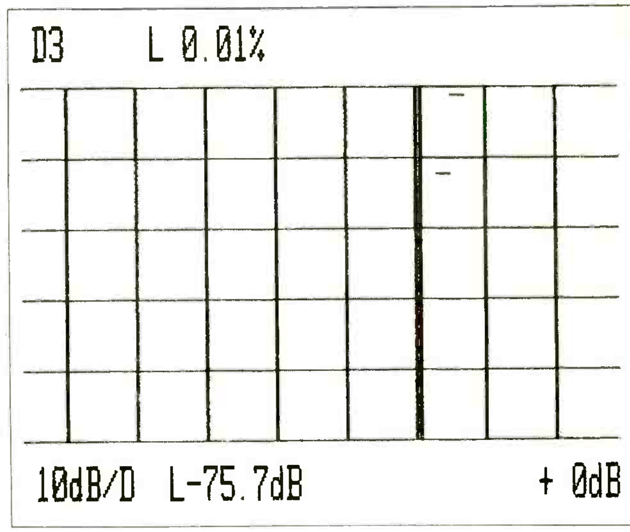


Fig. 2: Sony PCM-F1: Third-order distortion at 0 dB record level was approximately 0.01%.

peak hold and for checking the battery (if it is used). There also are toggle switches for muting on/off; copy on/off; and line or mic input selection. Two ¼-inch phone jacks for left- and right-channel microphone inputs are at the extreme right.

On both sides of the PCM-F1 near the front, are the attachments for using an optional carrying strap. Farther back on the right-hand side is the battery compartment. Pin jacks at the rear provide connections for audio line in and out (these may be interfaced with any analog signal device such as a conventional tape deck, mixer, amplifier, etc.); video in and out (these connect to the VCR); a copy-out option; and a resolution switch that selects either 14-bit (the EIAJ standard) digital encoding or a 16-bit system. Also at the rear is the operating power connector for 12 volts DC, which may be derived from the separate AC power adaptor or from a car-system adaptor. The 14 or 16 bit selection is a manual option in recording; for playback the machine automatically adjusts itself to the appropriate bit system.

Test Results: As a glance at our "Vital Statistics" table will show, the Sony PCM-F1 met or exceeded its published specs. At a "zero dB" recording level, frequency response remained flat within 0.5 dB from 20 Hz to above 20 kHz. Wow-and-flutter were literally unmeasurable, even on our sophisticated test setup. Signal-to-noise ratios were 88.1 dB and 89.9 dB for the 14-bit and 16-bit standards, respectively, and this was, of course, using the format "as is" without the aid of any noise-reduction system. Distortion at the zero-dB record level was virtually nonmeasurable. Of course, in a digital PCM system, distortion is expected to be vanishingly low up to the zero-dB level, but it characteristically rises sharply when maximum allowable levels are exceeded.

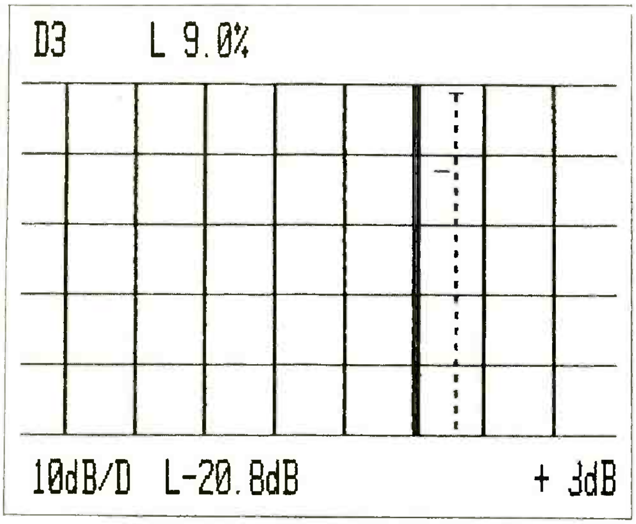
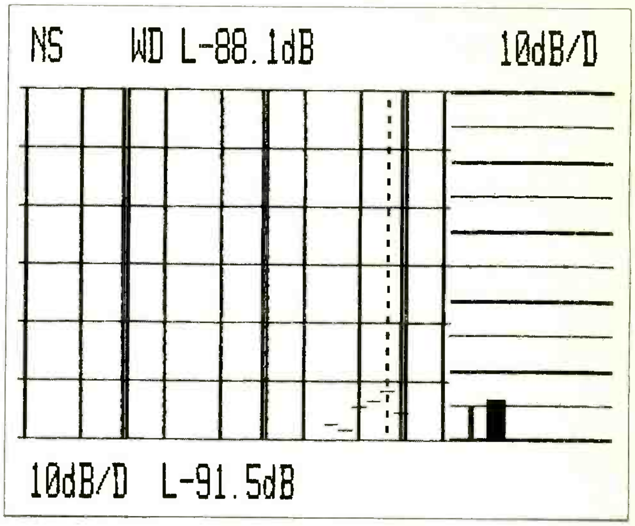
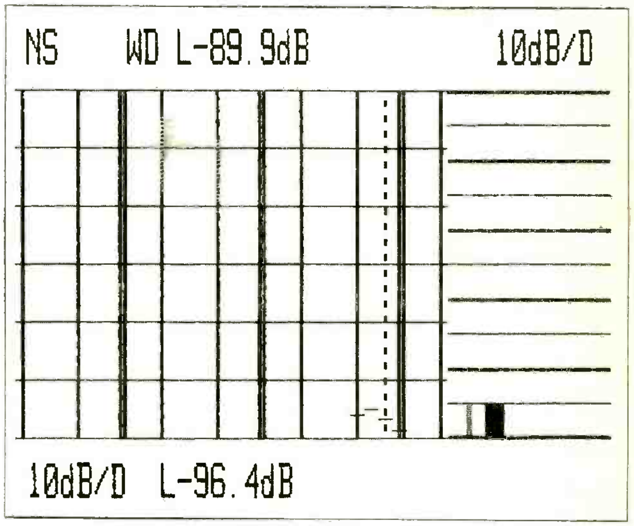


Fig. 3: Sony PCM-F1: Unlike analog tape recording, when max allowable levels are exceeded with a digital PCM system, third-order distortion rises rapidly. For the PCM-F1 it was 9% at +3 dB.



(A)



(B)

Fig. 4: Sony PCM-F1: Slight differences in S/N were observed when switching from the EIAJ (14 bit) standard PCM format [where we measured 88.1 dB, CCIR/ARM weighted (A)] to the standard 16-bit format also available on the unit, where we measured 89.9 dB (B).

Note that to show frequency response for the PCM-F1 we changed the vertical sensitivity on our response graphs (*Figs. 1A* and *B*) from the usual 10 dB per division to only 2 dB per division, in order to show the minute dB variation in the response. Had we used our usual 10-dB per division scale, the resultant response plot would have been literally a straight ruler-flat line. The expanded scale we used this time indicates a maximum deviation of 0.5 dB at the extreme low end, and between 0.3 and 0.4 dB at the high end.

Regarding distortion and recording headroom, with a PCM digital processor such as this, the operator must get used to the idea that "zero dB" as shown on the unit's meters, must not be exceeded. At the zero dB level, third-order harmonic distortion is so low that it "falls off" the bottom of the graph (*Fig. 2*), and the resultant distortion reading is a residual level rather than anything introduced by the processor. But, as mentioned earlier, recording over this level does shoot the distortion up dramatically—in this case it was nearly 10 percent for a level of +3 dB (*Fig. 3*).

However, the effective, usable recording headroom is still far above that of conventional analog recording since with "zero dB" as the maximum recording level, and working down from nominal "zero dB," the system provides available dynamic ranges close to 90 dB. The slight differences we measured between using the 14-bit and the 16-bit modes are shown in *Figs. 4A* and *B* (88.1 and 89.9 dB, respectively). Actually, the latter reading may be understated since it was fast approaching the limit of our test instruments (again!). Presumably, the already very low distortion improves somewhat when switching to the 16-bit mode, but again, the difference did not show up in our measurements, mainly because of the limitations of our test equipment.

Channel separation, or cross-talk, measured an impressive 59.7 dB at 20 kHz. At mid-frequencies (the double line down the center of *Fig. 5* represents 1 kHz), separation was even more astonishing at 83 dB!

As for wow-and-flutter, for all intents and purposes there simply was none. We show the wow-and-flutter "non-display" obtained with our Sound Technology test setup just to emphasize this point (*Fig. 6*).

PCM recording, when using a VCR as the tape medium, must "fit" its digital pulses in between the normal scanning lines of a standard NTSC television signal. The digital "words" (14-bit or 16-bit) are, in effect, interposed between the horizontal sync pulses of the standard

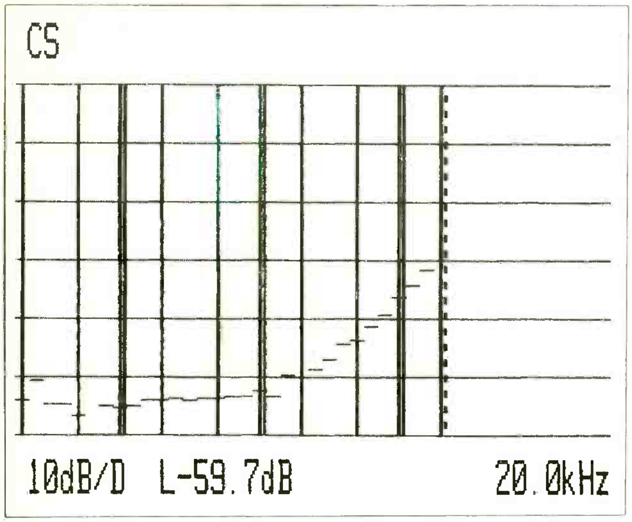


Fig. 5: Sony PCM-F1: Even at 20 kHz, channel separation measured nearly 60 dB—and measurement was probably limited by test instrumentation rather than by the PCM-F1.

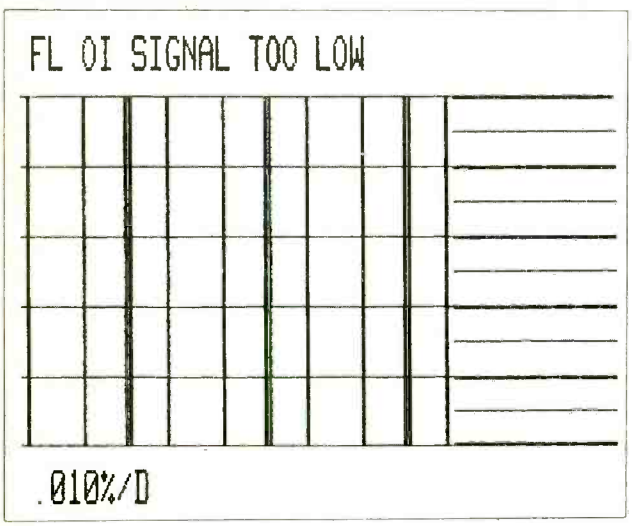
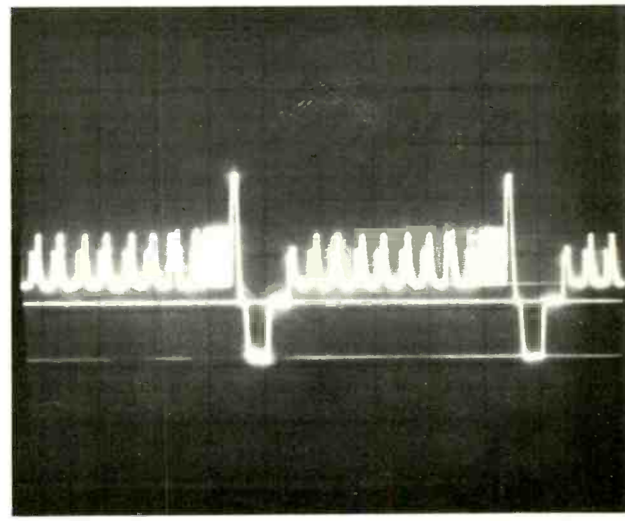
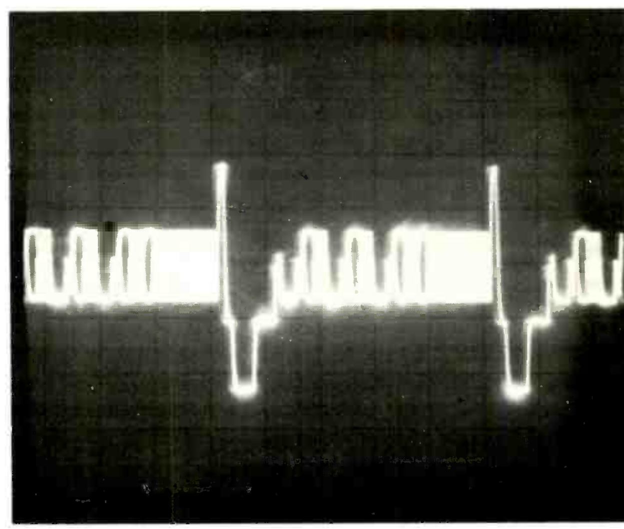


Fig. 6: Sony PCM-F1: The "message" at the top of this display "says it all" concerning wow and flutter for PCM digital recording; it's simply too low to measure.



(A)



(B)

Fig. 7: Sony PCM-F1: Signals produced by any PCM processor designed to work with video tape are really digital "pulse" signals superimposed within the horizontal sync pulses of a standard NTSC video composite signal. 7A shows the video output from the PCM-F1 with no audio input signal applied. 7B shows the same output with audio modulation applied to the input.

TV signal. To convey some idea of what all of this looks like in terms of its waveform, we photographed what appears on an oscilloscope connected to the video output (or video copy) jack of the PCM-F1. *Fig. 7A* shows a couple of video "lines" with no audio signal applied to the system. *Fig. 7B* shows the same video lines with a loud audio tone (zero dB) applied to the input of the PCM-F1.

As an item of further interest, the visual representation of the encoded (digitized) PCM signal as seen on a TV receiver screen is shown in *Figs. 8A* and *B*.

General Info: Dimensions of PCM-F1 processor are 8½ inches wide; 3¼ inches high; 12 3/8 inches deep. Weight is 8.8 lbs. Dimensions of AC-700 power module are 4¼ inches wide; 3¼ inches high; 12 3/8 inches deep. Price: $1900 for both units.

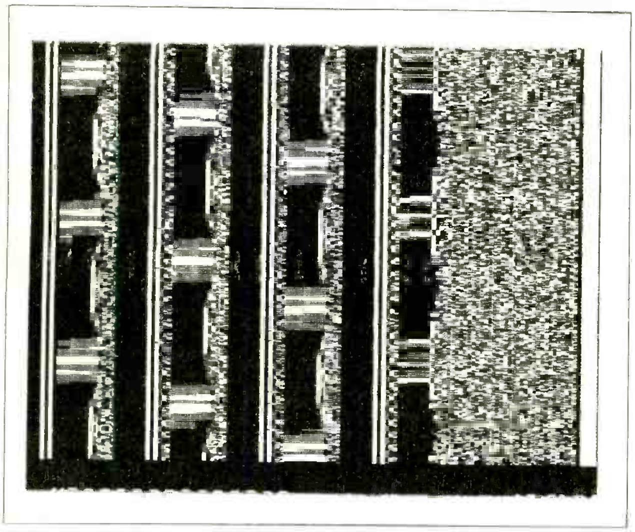
Individual Comment by L.F.: Since I have devoted my entire "Ambient Sound" column in this issue of *MR&M* to a discussion of PCM (digital) recording based on the use of video tape recording formats, I will limit these comments to a few points concerning the Sony PCM-F1, and what I learned about digital audio recording while using this remarkably compact, much. Even going to +3 dB pushes the distortion up to nearly 10 percent. You must think of dynamic range here in terms of *downward from 0 dB*, and in this manner the available dynamic range becomes tremendous (see our S/N figures). However, if like me, you are used to working with conventional decks and tapes that offer a certain amount of "headroom" above nominal "0 dB," then operating the PCM-F1 for the first time (or, more correctly, working with its metering system) will take a bit of getting used to.

Our tests documented the near-90 dB S/N of this system; its incredible channel separation; its nonexistent wow-and-flutter. In addition to the 'scope photos of the waveform, we were curious to see what the encoded or digitized PCM signal might look like on a TV viewing screen, since that signal is, after all, a video signal. We discovered that it appears as a "black and white" pattern (see *Figs. 8A* and *B*). I found this signal capable of creating the most fascinating (almost hypnotizing) patterns on a TV screen. Who knows—perhaps some bright kids in future generations will be able to watch one of these screens displaying a digital audio signal and will be able to hum the melody just from observing these complex patterns. If nothing else, studying these patterns is at least as productive as playing "Space Invaders™."

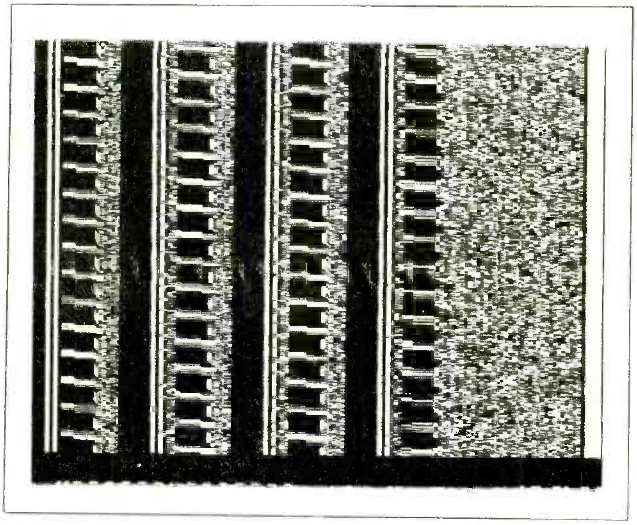
As a product, I think the PCM-F1 is an absolutely superb technological development. As recently as two years ago, it would have been impossible. As for its current usefulness in the world of audio recording (both pro and at home), I—once again—respectfully refer you to my "Ambient Sound" column.

Individual Comment by N.E.: "Future shock" is here and now with the Sony PCM-F1. This is not just a new product; it is a major turning point in sound recording and reproduction. The basic technology, which we had glimpses of a few years ago and had even played with in terms of costly first-off-the-drawing-board models and prototypes, is one thing—and a very remarkable thing at that. But now the fantasy of things to come has arrived in viable format. This system, according to our information, will be offered in increasing numbers to the pro, semi-pro and consumer markets. It is, in the vernacular, a new kind of "music machine"—and its essential link with video recording raises many intriguing ideas and questions in view of the tremendous upsurge in the sales of VCRs in the past year or so.

The plain fact is that with a device such as the Sony PCM-F1 and any VCR, you suddenly are equipped with a sound recording and reproducing system whose basic audio performance at least equals, and probably surpasses, that of the best studio-grade analog tape system (at least for two-channel work). Here is a sound storage and playback format that makes the 90-dB dynamic span with no help from ancillary devices. Recording at "0 dB" level, you get ruler-flat response, with non-measurably distortion and nonexistent wow-and-flutter. The system is limited, of course, by whatever associated analog audio material or devices used with it, which at



(A)



(B)

Fig. 8: Sony PCM-F1: Here's what PCM-encoded signals recorded on video tape "look like" when displayed on a TV set's screen. 8A, a low-frequency audio signal; 8B, a high-frequency audio signal.

lightweight processor. If you're interested in my more philosophical and subjective comments on this approach to digital audio tape recording, please flip back a couple of pages to my "Ambient Sound" column. Then turn to N.E.'s description and comments concerning this unit.

As most of you know, obtaining flat frequency response with a digital recording system is no problem. Our test data documents this, and at a zero-dB record level at that. Try getting that with *any* analog tape system.

With a PCM processor such as the Sony PCM-F1, you must remember not to exceed the indicated "0 dB" record level. At that level, distortion is so low, it literally cannot be displayed on the test instruments. But beware of "over-recording" or of exceeding "0 dB" by very this state of the art would seem to be mainly the loudspeakers—and these are getting audibly better all the time in terms of their own smoothness of response, and dynamic range with the power-handling capabilities associated with that range.

Stylistically and dimensionally, the Sony PCM-F1 is a perfect mate for Sony's model VCR model SL-2000 and its associated timer/tuner model TT-2000, both of which were made available for our tests of the PCM-F1. Of course, the PCM-F1 can be used with other VCRs, including the VHS as well as the Beta format. At any rate, using it with the Sony VCR afforded us a good sense of the practical aspects of taping this way. It is in some respects different from taping conventionally with an analog deck. For one thing, there's the matter of signal level watching—you do not exceed the unit's indicated "zero dB" marking without running the risk of rapidly rising distortion. On the other hand, there is no need to do so—you can, if you like, think of −15 dB on the Sony meter as your average level (this is suggested in the owner's manual) which still leaves a very healthy 15 dB of "headroom" for the very loudest clean signals any microphone is likely to be capable of delivering, and certainly this kind of level is quite ample for dubbing from existing program sources, including master tapes made by any method.

The system's muting arrangement takes getting used to. It comes on when the VCR itself is either mistracking or skewing the tape, and you hear nothing until you readjust the tracking control on the VCR.

For fast-wind of the tape on the VCR you must press "stop" first. If you engage fast-wind directly from play, you get a semi-fast speed as long as you hold down the button, and some of the signal will be audible, but not quite clearly enough for "cue searching."

Since the VCR has no separate play head, you cannot monitor directly off the tape while recording. You can, of course, listen to the signal going onto the tape—via speakers or headphones. The headphone facility on the PCM-F1 by the way, is exceptionally good, providing a really loud and clear signal.

The system, at least so far, is strictly two-channel. You can mix before it, of course, using an external mixer but all you can put onto a tape with this setup are two channels. Two superb channels, that is.

Beyond these, and some other, "mechanical" considerations is the fact of the sound itself. Sony applied a demo tape (a Beta video cassette), and I also dubbed some dbx-coded discs that had been cut from digital tape masters. It was the listening experience with these tapes that really got me off my seat, figuratively and literally. One passage, containing a lot of strong high-frequency percussives, actually caused me to look in the direction of my speakers to see if someone or something was actually there. The sound is simply tremendous. At any listening level from comfortably "soft" to unusually loud, it is so natural and convincing that it seems to obviate the need for using various devices that are intended to enhance the presentation of analog sound. As you listen it finally dawns on you that at last we have a sound system in which the program material and its immediate technology for playback, at least, no longer are limiting factors on musical realism. Not only that, but the nature of digital is such that making digital copies of a digital original need not introduce any signal degradation as it inevitably does with analog sound.

Listening to this system and mulling over these ideas, I got a feeling that this was an experience of discovery, not unlike how I felt when I heard my first FM broadcast or my first stereo years ago. I also thought of the various "criticisms" that have been made about digital, from its alleged "lack of ambience" to its alleged "ill effects" on some listeners' nerves. In my view, such allegations are utterly without validity. My experience with digital—unfortunately brief but hopefully to be repeated in coming months—was, in a word, exhilarating.

SONY PCM-F1 DIGITAL AUDIO PROCESSOR: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Frequency response (Beta video tape used)	± 0.5 dB, 10 Hz to 20 kHz	Confirmed
Signal-to-noise ratio re: 3% 3rd-order harmonic distortion, 14 bit; 16 bit	85 dB; 90 dB	88.1 dB; 89.9 dB
Record level for 3% 3rd-order HD	NA	+ 2 dB (*see text*)
Wow-and-flutter, WRMS	NA	Unmeasurable
Line output at 0 dB	316 mV	450 mV
Headphone output level at 0 dB	adjustable, 50 to 775 mV	Confirmed
Mic input sensitivity for 0 dB	0.435 mV	Confirmed
Line input sensitivity for 0 dB	316 mV	300 mV
Power consumption	17 W	15 W

Fostex A-8 Tape Deck

General Description: A rather unusual product, the Fostex A-8 offers the facility for recording up to eight tracks on quarter-inch open-reel tape. The format is divided into two groups of four tracks each, so that it is possible to record up to four tracks with one pass of the tape from supply reel to takeup reel. The tape then may be rewound, and a second group of four tracks recorded. Switching of the various gaps in the combined record/play head permits sync recording (a performer, for instance, can listen to what has been recorded while adding a new part), although the lack of a separate playback head precludes monitoring the full results on the tape itself until the recording is finished and the completed tape is put into the playback mode.

Having no signal level controls, no microphone inputs and no headphone output, the Fostex A-8 must be used with external equipment such as mic preamp or mixer. With a four-input mixer, performers can overdub to fill all eight tracks.

The A-8 runs at a single speed, 15 inches-per-second. It handles seven-inch diameter reels, which are held in place by screw-on clamps. A Dolby-C noise-reduction system is built-in; this circuit may be switched off and by-passed for optional use with an external noise-reduction device if desired. In addition to overdubbing and punch-in recording, the A-8 can be used for ping-pong recording, and for dump-editing. The deck's four inputs may be switched to either of the two groups of four tracks in recording. The eight outputs, for handling its maximum available multitrack arrangement, can drive a mixer with up to eight inputs for mixdowns.

The upper portion of the deck contains the transport and head assembly. A cue lever near the heads may be used to monitor signals on the tape in either of the fast-wind modes. Near it is a head shield that may be retracted for tape threading and editing.

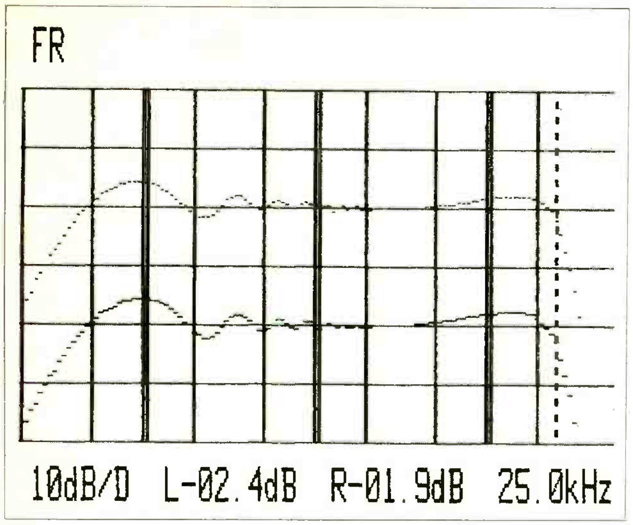
The lower portion of the deck contains the eight VU meters and various operating controls. At the extreme left is the AC power off/on switch. The VU meters (calibrated from −20 to +5) are arranged in two groups of four each, corresponding to the two groups of four tracks (1-4; 5-8). Above each meter is an LED which blinks to indicate a given track is record-ready, and which lights up steadily when that track is being recorded.

Associated with the meters and their tracks are eight pushbuttons. One row of four buttons is used in conjunction with a group-selector switch that determines which group of four tracks may be recorded at a given time. Thus, with the group-selector set for the "upper" tracks, the record-track buttons select tracks 1 through 4. With the group-selector pushed in for the "lower" tracks, the record-track buttons select tracks 5 through 8. Just below these buttons are four additional buttons that serve as monitor selectors, again choosing tracks 1 through 4 or tracks 5 through 8.

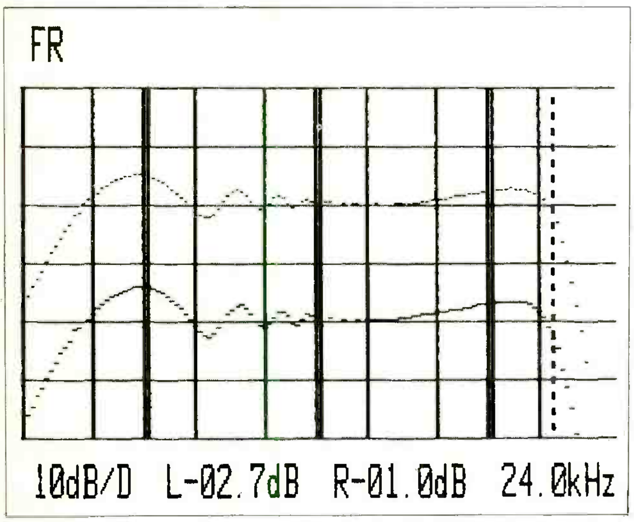
Above the buttons are a four-digit tape counter and reset button. To the right is a noise reduction switch which allows for activating an external system or the unit's Dolby C system. Another switch provides for the "dump" edit option. To the right is the pitch control which is operative in record and in play modes. A center detent position indicates 15-ips speed.

Finally there are six transport keys for "zero return," rewind, fast-forward, record, stop and play. The "zero return" button may be used as a "memory rewind" option—if the tape counter reset button is pressed to "0000" (regardless of how much tape has been used) the "zero return" button then may be used to return the tape in fast-wind mode to that "zero" point, where the tape will stop. The transport buttons are "feature-touch" and logic-controlled, permitting fast-button options between all modes, including run-in recording from "play" with no stop, and a brief stop from the fast-wind modes.

The rear of the A-8 contains four pin-jacks for the inputs, and eight pin-jacks for the outputs. Also at the rear are connectors for use with optional accessories—one, a



(A)



(B)

Fig. 1: Fostex A-8: Frequency response plots for two adjacent tracks of the recorder (record/play) at 0 dB record level (A); and at −10 dB record level (B).

remote-control for transport functions; the other, a jack for use with a foot-switch that may be used to put the deck into record mode during punch-in/punch-outs, and to return it to record-ready. The AC line cord completes the rear picture, although there is a series of alignment adjustments hidden behind a panel on the underside of the deck, some of these indicated as for possible use by the owner and others designated only for use by qualified technicians.

As stated, the A-8 has no signal level controls and no mic input jacks. All these facilities are expected to be used as provided on external mixing equipment. Fostex, with the A-8, has created a tape machine with the musician in mind, rather than the advanced recordist. The A-8 seems especially geared towards groups that need to record their music in preparation for the pro recording studio. It allows for overdubbing of parts that you may not be quite sure of during rehearsal or while writing the initial tune. While the cost may seem high for a machine with certain feature limitations, it does do its intended job well. And when comparing its cost versus the cost of *pre*-production time in a professional studio, an artist would come out well ahead.

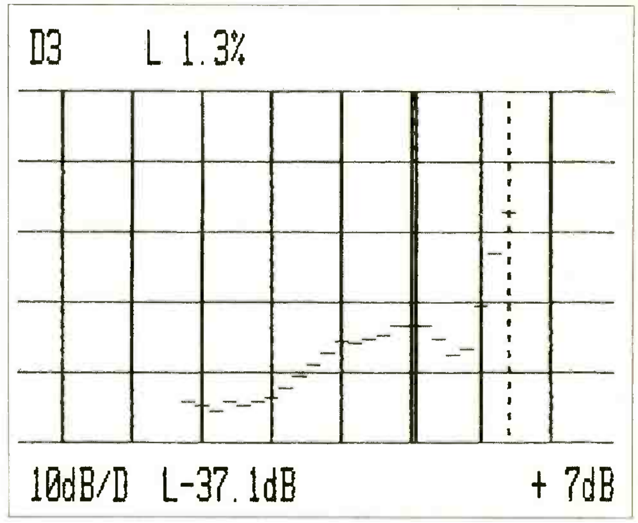


Fig. 2: Fostex A-8: Third-order distortion vs. recording level.

Test Results: In our lab tests, the Fostex A-8 met its published specifications, but qualifiedly so. To explain: *Figs. 1A* and *1B* show frequency plots of two adjacent channels of the deck from 20 Hz to 20 kHz. In *Fig. 1A* the recording level used was 0 dB which is referenced in this machine for 250 nWb/m. In *Fig. 1B*, recording level has been backed off by 10 dB. As might be expected, the high speed (15 ips) of the A-8 recorder provides an excellent high-frequency response that exceeds spec, going out to about 25 kHz for a −3 dB rolloff, using Maxell UD-XL open-reel tape (1800-feet, 7-inch reel, 1-mil thick). Indeed, at the frequencies indicated (25 kHz for *Fig. 1A*, and 24 kHz for *Fig. 1B*), the response has not quite rolled off by a full 3 dB.

In contrast, however, note the very poor contour effect at the low end of the response, caused primarily by the closely spaced, narrow track widths. We often see some of this effect when measuring cassette deck frequency response. In terms of audio performance, this contour effect would have a degrading effect on an attempt to maintain phase coherency in any future mixdown for stereo. It also adversely affects channel separation at the low end.

In *Fig. 2*, we show the results of our plot of 3rd-order distortion versus recording level. The double vertical line represents 0-dB record level, at which point we measured a distortion of 0.22 percent, obviously less than the spec. Notice, however, that at higher recording levels, the apparent distortion actually gets lower before rising again.

We suspect that what we really were reading at the 0-Db level was 3rd-order distortion plus noise which proved to be high enough to enter into the picture. In any event, at +7 dB, third-order harmonic distortion reached 1.3 percent (as indicated in *Fig. 3*), and we extrapolated the curve to estimate that the 3-percent distortion point would occur at about +9 dB—close enough to the claim of +10 dB so as not to quibble.

A direct correlation between channel separation and the head contouring effect mentioned earlier may be seen in *Fig. 3*. Mid-frequency channel separation, better than 50 dB at mid-band, is excellent. Moving down in frequency, the separation readings fluctuate wildly, and deteriorate to only −17.7 dB at 20 Hz.

On the matter of signal-to-noise, the instruction manual advises that the built-in Dolby-C facility be kept on at all times unless another type of noise-reduction system is externally connected. The specs also quote a figure of 72 dB for S/N "weighted" (the actual weighting curve used is not stated), as against a figure of only 60 dB S/N unweighted. Our tests show that with the Dolby-C on, and using CCIR/ARM weighting in making the S/N measurement, you get an impressive S/N of 77.5 dB. Remove the Dolby, and the reading drops to 61.3 dB. In our view, that discrepancy seems "suspicious" inasmuch as Dolby-C should add about 20 dB to the S/N figure. The reason for the discrepancy is that the machine yields a significant amount of low-frequency noise, which Dolby-C doesn't do much about. CCIR/ARM weighting tends to obscure this effect, however. To confirm our suspicion, we measured the noise with Dolby-C on again, but without any weighting curve. The results are shown in *Fig. 4*, with an overall reading of 65.7 dB—but notice, in the spectrum analysis of the noise (in third-octave increments) how it rises at low frequencies, peaking in the power-supply frequency regions of 50 to 60 Hz.

The one parameter we checked and found to be really fine without qualification was the machine's very low wow-and-flutter, attributable of course to the 15-ips speed. As shown in *Fig. 5*, we measured only 0.029 percent, and we should add that this figure was obtained as a peak-weighted number (IEC/ANSI) rather than as the usual WRMS figure which most manufacturers use because it results in a lower number.

General Info: Dimensions are 14 inches wide; 13.5 inches high; 6.75 inches deep. Weight is 29 lbs. Price: $2500.

Individual Comment by L.F.: I can't imagine a serious recordist who would be willing to cram eight master tracks of a recording session onto ¼-inch wide tape—especially after seeing the kind of low-frequency contouring effects that such narrow track widths produce. And, as I was soon to discover, the incorporation of Dolby-C—while certainly worthwhile in any high-quality deck—does not successfully cover up the fact that the unweighted signal-to-noise ratio achieved for each track is less than superb.

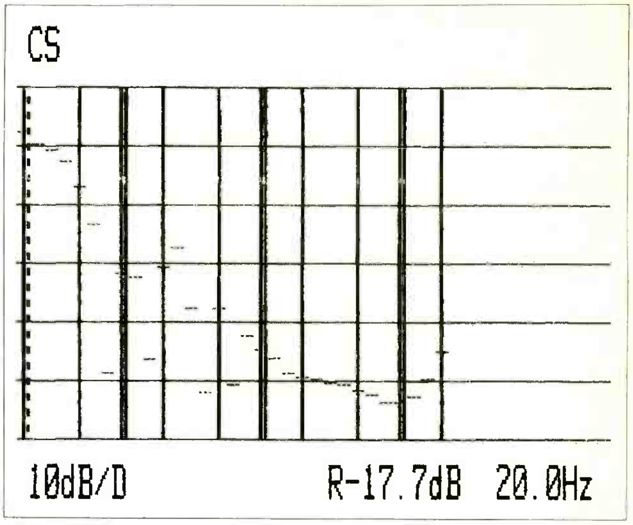


Fig. 3: Fostex A-8: Channel separation, adjacent tracks, versus frequency.

For that matter, the designation of "8 track" turns out to be a misnomer, too. Yes, there are eight separate "tracks" visible on the surface of the common record/play head (the machine uses a common R/P head, so you can't monitor recorded results as you record), but you can record on only four of them at one time. To get the second quartet of tracks you must punch the "group selector" switch, and record all over again. I suppose that's no problem for the solo musician layering tracks, but I can't imagine breaking up a musical group into two separate sessions ("You guys will record first, on tracks 1 through 4, and you fellows come back later when we're ready to record on tracks 5 through 8, okay!"). I could go on about some of the not-so-sophisticated arrangements incorporated into this deck, but our lab measurements—as detailed above—can serve to reinforce my negative posi-

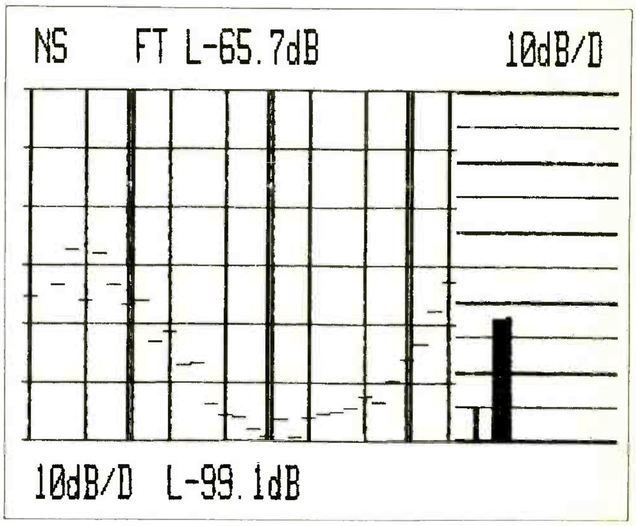


Fig. 4: Fostex A-8: *Unweighted* plot of noise shows high level of *low-frequency noise,* which is not helped by Dolby-C noise reduction system.

tion instead, such as the poor contour effect at the low end of frequency response and the fact that the machine yields an enormous amount of low-frequency noise which is not helped by the built-in Dolby-C. The 15 ips speed is good for low wow-and-flutter, but used with a 7-inch reel holding 1800 feet of tape, that speed limits you to a maximum uninterrupted recording time of only 24 minutes.

I expect now that I will get letters, from readers as well as from Fostex, telling me about how I missed the point of this product (which, incidentally, has no self-contained level controls and must therefore be used with some form of mixing console). Well, maybe so, and if I have missed something here I will be glad to rescind my negative comments in the future. But for now...

Individual Comment by N.E.: I think the design philosophy behind the Fostex A-8 is to offer a recorder that can do more than the typical home-type open-reel deck, but one which is hardly in the same class as professional-grade equipment. As such it could interest the owner of the "home studio" type of setup who wants to record a small group, or one or two performers as they build up successive tracks. Obviously, the total number of tracks possible in any given recording session is four, so that if a larger number wanted to record on the A-8 at the same time, they would have to go through a mixer which accepted their own number and mixed that down to four tracks. Alternately, of course, it becomes a matter of recording four directly and then syncing in the next four on the remaining half of the tape. At that, putting eight tracks onto quarter-inch wide tape is crowding things (the total width per track becomes less than it is on four-track cassettes), and the problems we encountered in the lab with low-frequency effects seems to confirm that view.

In any event, if the A-8 is intended to appeal to the less professionally oriented recordist, the very need to supply one's own level controls, mic inputs and headphone facility (via external mixing equipment) seems to run counter

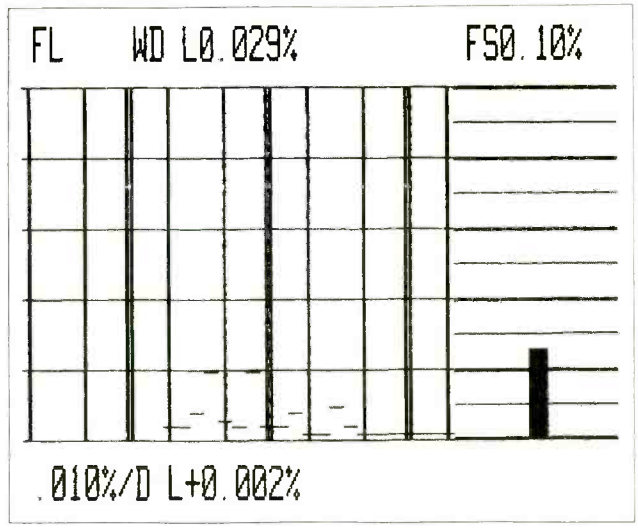


Fig. 5: Fostex A-8: Wow-and-flutter analysis.

to that approach, especially since that additional equipment does cost—and the asking price of the A-8 isn't exactly "economy class" to begin with.

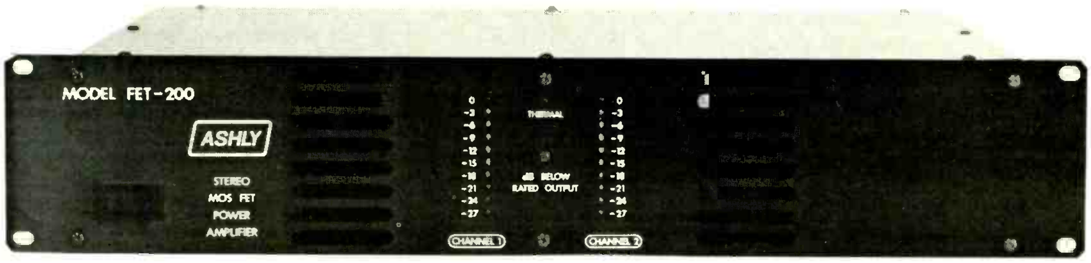
Setting up and using the A-8 takes some getting used to; it is very easy at first to press the wrong buttons. One more thing: The owner's manual contains instructions for checking alignment, which involves getting at some pot-adjustments. The pots are located behind a panel on the underside of the machine. The panel is next to impossible to remove unless one pops the rivet-pins holding it in place, which then makes it impossible to replace the panel. Why didn't they use ordinary machine screws? Moreover, in this position, the deck is literally standing on its "head" or lying on its "face" so that observing the eight front-panel meters for correct indications becomes the neatest trick of the year, if you can do it.

All told, it appears that Fostex had a workable idea with this deck, but perhaps between the drawing board and the finished product a couple of things went astray.

FOSTEX A-8 TAPE RECORDER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Frequency response, 15 ips	± 3 dB, 45 Hz to 18 kHz	± 3 dB, 40 Hz to 25 kHz
THD at 0 VU	less than 1.0%	0.22%
Record level for 3% 3rd-order distortion	NA	+ 9 dB
Best S/N ratio (std tape)	72 dB	77.5 dB w/Dolby C 61.3 dB wo/Dolby C (both CCIR/ARM wtd)
Wow-and-flutter	0.06%	0.029% (ANSI-peak wtd)
Fast-wind time, 1800 ft reel	130 seconds	2 min., 42 sec.
Line input sensitivity	300 mV	312 mV
Line output level	300 mV	312 mV
Erase ratio	greater than 70 dB	greater than 75 dB
Speed accuracy	± 0.5 %	Variable via ± 10% pitch control
Power consumption	60 watts	58 watts

Ashly FET-200 Power Amplifier



General Description: A stereo power amplifier, the Ashly FET-200 is rated for 100 watts output per channel into 8-ohm loads; 160 watts per channel into 4-ohm loads; and 320 watts output into 8 ohms bridged mono operation. The amp is intended primarily for professional audio use as evidenced by its extra-rugged construction and design, rack-mount appointments, input and output connector facilities and internal cooling fan. The action of this fan (which is very quiet by the way) and the circuitry of the amp itself, which makes use of MOS-FET (metal-oxide-semiconductor, field-effect-transistor) technology, is said to avoid any problems of "thermal runaway" which enables not only the design of a superior-sounding amplifier, but one which may be installed in a tight rack-mount situation with no need to provide air-space below or above it.

The FET-200's front panel contains the power off/on switch; and two indicator "ladders" showing power output separately for each channel. The markings for the ten LEDs in each ladder show peak dB levels below rated output in steps of −3 dB from "0 dB" down to −27 dB. Also on the front panel are the intake vents for the amplifier's fan and an LED showing thermal overload.

Input level controls for the two channels are at the rear, just above their respective input connectors. Each channel offers a choice of three types of connector: male and female XLR, plus ¼-inch phone jack. All input connectors on each channel are wired in parallel, but since only one connector would be used at any given time the remaining connectors for that channel may be used to feed additional amplifiers with the same signal source feeding the FET-200. The input to each channel is a balanced bridging type. However, the input also may be used in a single-ended, unbalanced mode if one opts to use unbalanced input connectors.

Two mode switches are located near the signal input connectors. One switch provides for bridging or normal; the other, for mono or stereo. With the latter switch in mono position, the channel-one input feeds both amplifying channels of the FET-200, and the level control for channel one adjusts the gain for both channels. With the amplifier in mono mode, and the bridging switch activated too, the FET-200 goes into its bridging mode, in which case the channel-one input still feeds both sides of the amplifier but the phase of the signal through channel 2 is inverted, so that the outputs from both sides of the amp are 180-degrees out-of-phase. This option, which provides the highest available output power for driving a mono speaker system, requires that the speaker be connected across the two "hot" output terminals of each stereo speaker output.

The speaker outputs are standard binding posts with banana jacks spaced on standard ¾-inch centers. Spacing between the hot posts of each stereo pair also is ¾-inch, so that a double banana plug may be used for the bridged mono operation. Of course, if preferred, speaker hookups also may be made with stripped leads. Above the connectors is a fuse for each side, and there's a third fuse for the AC line at the right-hand corner of the amp's rear panel. The AC cord requires a three-wire (grounding) outlet. The opening for the ventilating fan is centered on the back panel between the input and output connectors.

Test Results: Published specs for the Ashly FET-200 were readily confirmed or exceeded in our lab tests. Power output values were slightly better than claimed; distortion levels were generally much lower than stated. Damping factor was very high. The signal-to-noise characteristics (not listed in our preliminary owner's manual, but presumably to be filled in for subsequent editions) were—by any standard—excellent.

The power output LED scales were found to be extremely accurate, right down to the bottom indicator for −27 dB. With 100 watts referenced as "0 dB," this means that the first LED begins to "flash" with as little as 0.2 watt delivered to a speaker.

This kind of precision, combined with the unusual versatility for using various input connectors, plus the amplifier's general ruggedness are all commendable. The amplifier we tested is also quite rugged and, we are told, subsequent units will be made even more so with such niceties as thicker PC boards. Sound of the Ashly FET-200 was excellent, and the fan was possibly the quietest-running we have yet encountered.

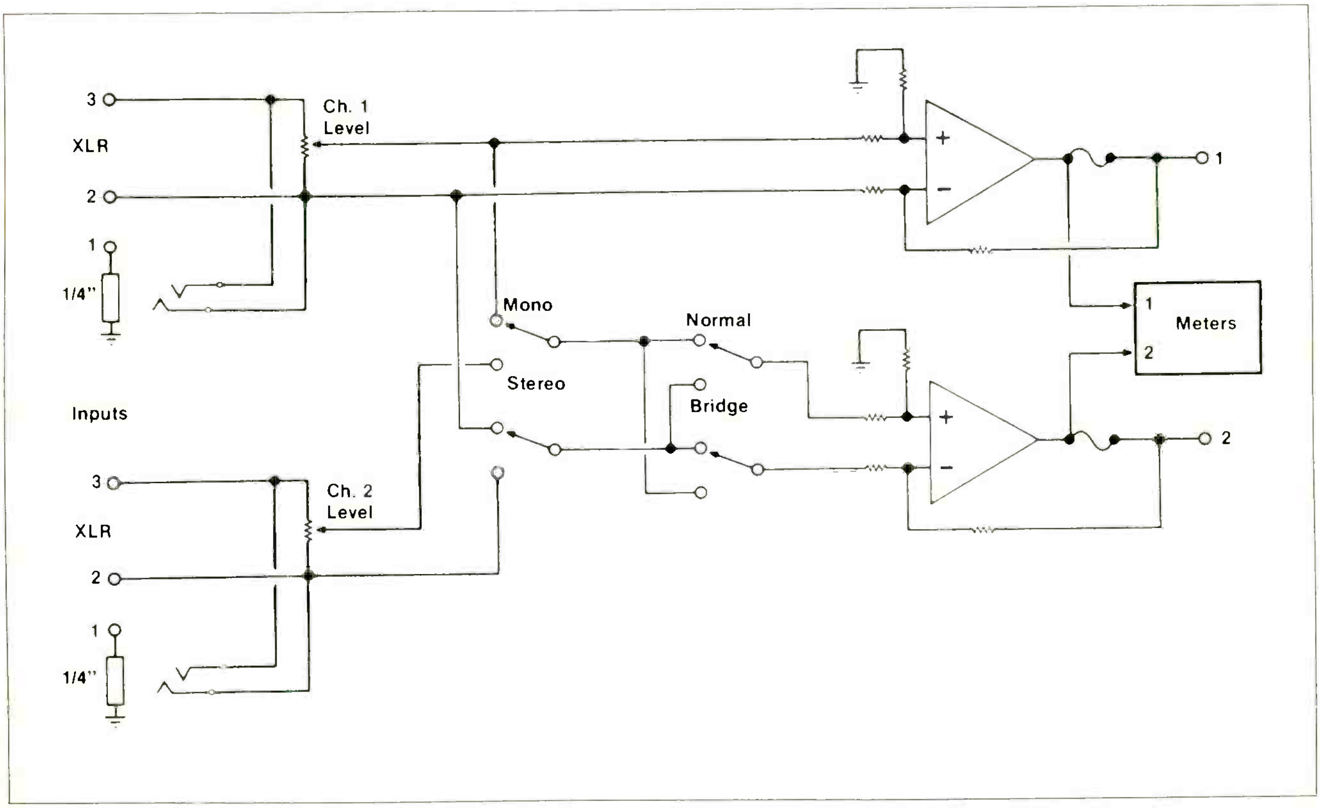
General Info: Dimensions are 19 inches wide; 3.5 inches high; 15.5 inches deep. Weight is 33 pounds. The price of the unit is $699.

Individual Comment by L.F.: I confess that when the Ashly FET-200 arrived my first reaction was: "Oh, another power amp to test. How boring." After all, I see dozens of new amps each year, and with only a few rare exceptions most of them are decently executed designs that can be neither faulted nor exalted. Surprisingly, the Ashly FET-200 turned out to be one of those few very rare exceptions.

To begin with, the amp utilizes MOS-FETs in its output stages. While MOS-FETs have been used in both hi-fi and pro amps before, most designers have stressed "better sound" as the reason. Actually, a more important reason is the fact that they have far less tendency than other transistor types to suffer from thermal runaway in power output stages. Thus they make a lot of sense as output devices in a pro amp that is going to be subjected to hard use. Given this reliability, the circuit designer then feels free to bias the output devices at a much higher quiescent (no-signal) output current which in turn results in more linear operation of the amplifier at all power levels. I strongly suspect that this characteristic is what prompts many critical listeners to say that "MOS-FETs sound better."

As for input connector facilities, I don't know how often you've been frustrated by finding the wrong "sex" XLR connector (for a balanced input) on an amplifier with regard to your own connecting cable that also had the same "sex" connector. Since there are no "world standards" for these input and output arrangements, Ashly has done a most logical thing and provided both female and male XLR connectors, not to mention ¼-inch phone jacks also. It is thus possible to make balanced or unbalanced connections to this amplifier no matter what kind of cable terminations you may be stuck with. This may not seem to important right now, but it can be very telling the next time you are caught with the "wrong" plug at the end of a cable that you're ready to plug into an amp.

It is my understanding that subsequent production



Ashly FET-200: Block diagram of the FET-200.

units will be accompanied with a more detailed owner's manual, and that the production-line amps themselves will boast more rugged circuit boards and that a few other minor discrepancies will be cleared up. That being the case, and considering that I liked the amplifier even in its not-quite-final prototype form, I have no reservations about giving it high marks for design, reliability and sonic performance. The under $700 price tag of the Ashly FET-200 seems very appealing for an amp of this quality.

Individual Comment by N.E.: A frequent question in audio circles is: Can an audio device be designed and built to function as a professional-grade "audio tool" and still interest the lay but critical audio-minded person because of this sonic performance and "product personality?" As with some other products we have reported on in the past, the answer with regard to the Ashly FET-200 is definitely "Yes." It seems apparent, based on our examination and listening tests, that the same design approach that contributes to the FET-200's reliability also contributes to its fine sound—that MOS-FET technology which can be credited with allowing the amplifier to respond "flat" and with minimum distortion at widely different power output levels and into varying load impedances.

Commendable also is the thoughtfulness that provides for three different kinds of input connector; the no-fuss and immediate availability of either balanced or unbalanced connections; the useful and accurate peak-reading output metering; the options for "normal" mono as well as "bridged" mono; the ease with which the unused inputs can be used to "jumper" other amplifiers (that is, feed them with the same signal that is being fed to the FET-200).

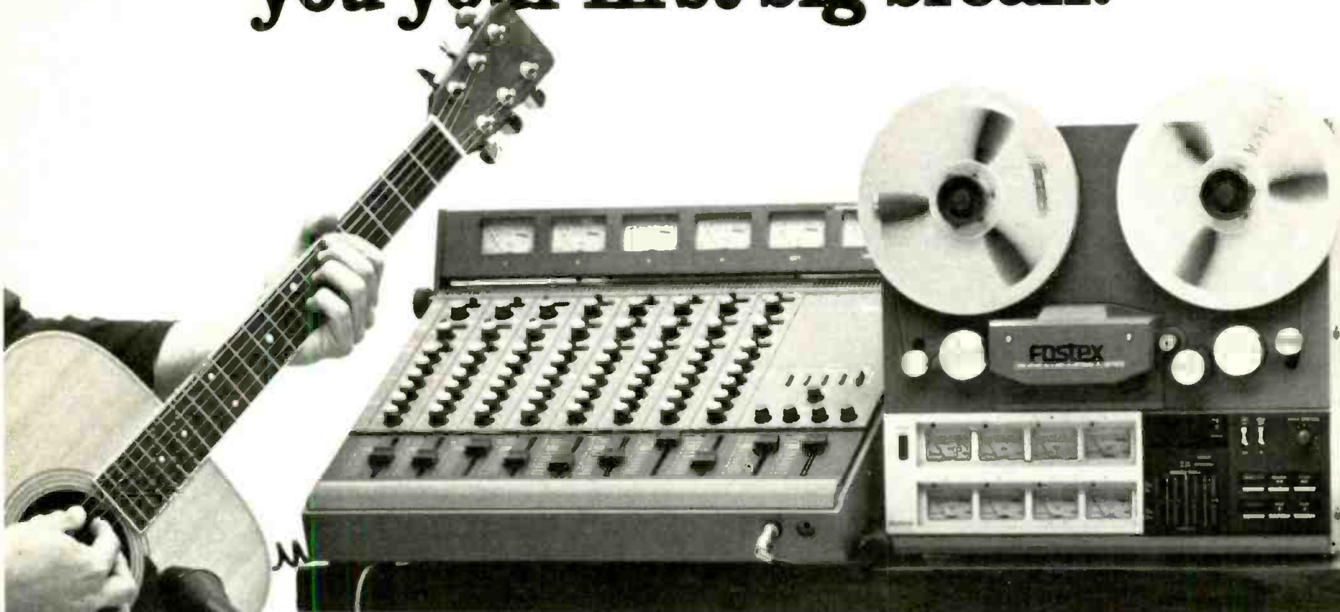
In terms of physical construction, the FET-200 is housed in a one-piece, welded 16-gauge steel chassis with an easily removable steel top cover. The power transformer is located near the front, which reduces stress on the chassis and also helps you bear the load a little when removing it from a rack. And that fan is really quiet. Ashly's avowed goal in producing this the FET-200 was to offer "an amplifier that combines the sonic excellence of a high-end stereo amp with the ruggedness and stamina necessary in pro audio." As far as we can determine, they have succeeded admirably in realizing that goal.

Actually, about the only thing that might even remotely be questioned about this amplifier is the location of its input level controls at the rear, and admittedly this is more of a subjective question than one of basic amplifier design and performance. Some users may prefer level controls up front; others might feel, as obviously Ashly does, that these are not really "operating" controls and should be set once for a given installation and then left alone. I would also guess that locating them close to the inputs does help reduce hum and noise another jot or so.

ASHLY FET-200 AMPLIFIER: Vital Statistics

Continuous power for rated THD		
8 ohms, 1 kHz	100 watts	109 watts
4 ohms, 1 kHz	160 watts	162 watts
FTC rated power (20 Hz to 20 kHz)	100 watts	100 watts
THD at rated power, 1 kHz, 8 ohms	less than 0.05%	0.004%
1 kHz, 4 ohms	NA	0.01%
20 Hz, 8 ohms	less than 0.05%	0.01%
20 kHz, 8 ohms	less than 0.05%	0.06%
IM distortion, rated output, SMPTD	less than 0.01%	0.007%
CCIF	NA	0.002%
IHF	NA	0.05%
Frequency response at 1 watt	± 0.5 dB, 10 Hz to 50 kHz	± 1 dB, 5 Hz to 80 kHz
S/N ratio re: 1 watt, "A" wtd, IHF	NA	91 dB
S/N ratio re: rated output, "A" wtd	NA	110 dB
Dynamic headroom, IHF	NA	1.02 dB
Damping factor at 50 Hz	greater than 100	100
Input sensitivity, IHF	NA	0.17 volt
Input sensitivity re: rated output	1.4 V	1.7 V
Slew rate (volts/microsecond)	50	confirmed
Power consumption, idling; maximum	NA	100; 500 watts

We broke the rules to give you your first big break.



1/4 INCH, 8-TRACK SYSTEM. $3,635.*

Finally, you can afford to sit down and make your own music. Instead of struggling as a part-time studio owner just to pay for equipment you can't afford.

It's called Fostex Personal Multitrack. An incredible technological breakthrough — 8 tracks on ¼" tape — from the same engineering team that first put 8 tracks on ½" tape.

And while the Fostex A-8 lets you build your tune for a fraction of your usual cost, we're not giving you economy at the expense of quality.

For one thing, we guarantee adjacent crosstalk at −45dB. That's exactly what the ½" 8-track gets, but you'll more than likely get −50dB or better. And with Dolby** C NR built-in you'll get a noise-free, sonically true recording of everything from a kick drum to a triangle.

You'll also be happy to know the A-8, like all Fostex recorders, can give you a third hand. A remote foot pedal for punching in and out. So you can make your own music and operate your own machine. Alone.

What's more, the A-8 is truly portable. Unlike a Hammond B-3, the entire Fostex 8-track system will fit in the front seat of your car.

Which brings us to the companion Fostex 350 mixer. 8-in, 4-out with an auxiliary stereo buss, 2 stereo phone inputs, 2 head phone jacks, multiple patch points and dual parametric equalizers on every input channel. Whew.

Plus, it's laid out so simply you don't have to be an engineer to work it.

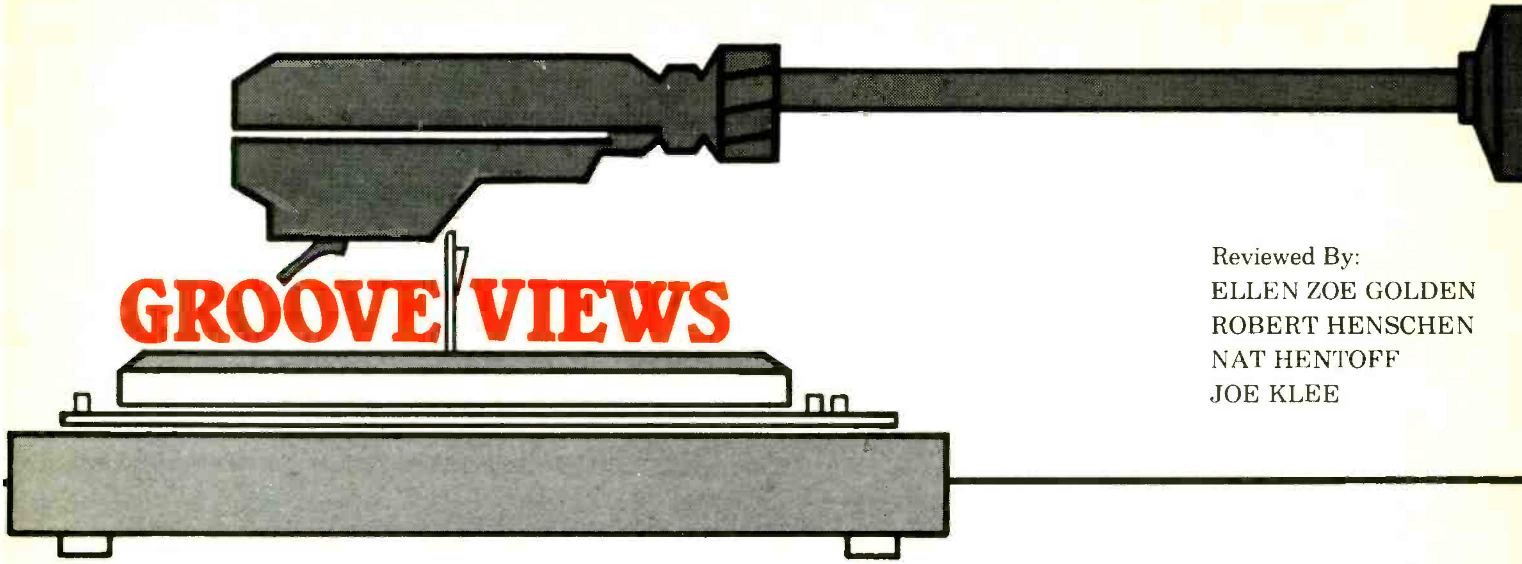
So don't get side-tracked by anything else. Head straight to your Fostex dealer. Then you'll see just how big a break this 8-track system really is.

FOSTEX®

Personal Multitrack

*Suggested list prices: A-8, $2500. 350 Mixer and Meter Bridge, $1125. Foot pedal, $15. **Dolby is a registered trademark of Dolby Laboratories, Inc.

CIRCLE 173 ON READER SERVICE CARD



Reviewed By:
ELLEN ZOE GOLDEN
ROBERT HENSCHEN
NAT HENTOFF
JOE KLEE

POPULAR

BOB WEIR: ***Bobby And The Midnites***. [Gary Lyons, producer; Gary Lyons, John Cutler, Gregory Mann, Peter Thea, engineers; recorded at The Record Plant in Sausalito, Le Club Front in San Rafael, Ca., and Media Sound in New York, N.Y.] Arista AL 9568.

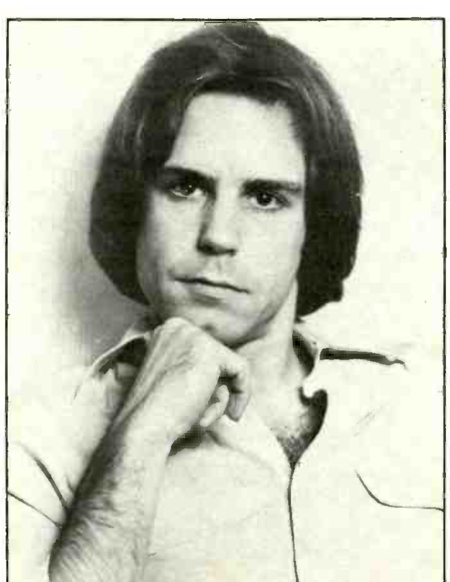
Performance: **Update on the Dead**
Recording: **Semi-polished**

The voice of The Dead meets the muscle and bone of Weather Report, the Mahavishnu Orchestra, Steppenwolf, and a battery of other apparently disparate outfits. It's a different kind of effort that Bob Weir unveils here with a superstar gathering of sidemen, and yet their outside influences seem to have very little bearing on what transpires.

Bobby And The Midnites, now a touring as well as recording group, boasts Billy Cobham on drums, Alphonso Johnson on bass, Bobby Cochran (from the Flying Burritos and Steppenwolf) on lead guitar, Brent Mydland (currently Dead) on keyboards, and Matthew Kelly on harmonica. You'd expect a hint of jazz from this ensemble, but slashes of steamy R&B and some hues of blues are as close as they get. For the most part, this is an album of rock songs that is leaning quite obviously in the direction of AOR airplay.

Blatant hooks are prevalent on "Too Many Losers," "Far Away," and "Me Without You," all of which seem geared for large scale concert staging and mass appeal. Weir seems to be into Springsteen ("Too Many Losers") and Tube sounds ("Me Without You") in terms of projecting energy, and he must have analyzed the charts to learn what is selling these days. To a large extent, half of these tunes find that formula without overcompromising the Grateful Dead legacy. "(I Want To) Fly Away" and "Haze" contain fine examples of Dead blues-rock.

"Book Of Rules" is an interesting departure into light stepping ska, even doing some acappella harmony vocals, and the Caribbean influences show up effectively again on "Fly Away." As for the less adorned blues and rock mainstream of "Josephine"—probably the closest cut to what you'd call a traditional Bob Weir sound—it's contrastingly dull. Likewise, "Carry Me" and



BOB WEIR: Abandoning the gameplan.

"Festival" lack the inspiration that occasionally burns in Weir's less characteristic material, and they sound somewhat dated.

Abandoning the old gameplan does wonders for Weir elsewhere. While neither the recording nor the performance could be deemed progressive at this point in time, Weir shows signs of contending with the enigmatic Eighties. As for his once spectacular sidemen, especially Cobham and Johnson, one has to wonder what's happening to *their* music lately... R.H.

KARLA DEVITO: ***Is This A Cool World Or What?*** [Dick Wingate, executive producer; Bill House, producer all cuts but "Heaven Can Wait," "Midnight Confession," and "Just One Smile," which are produced by John Jansen; mixed by John Jansen and Jay Messina at The Record Plant Studios, New York, N.Y.; recorded by Biff Dawes with Dennis Mays at Wally Heider/Filmways, Studio #3, Los Angeles, Calif.; Jay Messina at The Record Plant Studios, New York, N.Y.; Neil Dorfsman at The Power Station, New York, N.Y.; mastered by George Marino at Sterling Sound, New York, N.Y.] Epic 37014.

Performance: **So hot, it's cool**
Recording: **Clear as ice**

Karla DeVito asks, *"Is This A Cool World Or What?"*, but based on the strength of her debut LP, I think she already knows the answer. Her ambiguous question comes from playfulness rather than doubt since she doesn't appear to be puzzled by the trials and tribulations of today's life. No,

she sounds rather sure of herself and shows it by taking the bulls by the horn and singing out with an exuberance that's refreshing and addictive.

The music on the album explodes as a result of Karla's wonderful voice—something like a mix of Little Nell (remember the squeaky singer in *The Rocky Horror Picture Show*?) and a white Diana Ross—and excellent musicians (a whole slew of them), plus perfectly suited material. Producers Bill House, John Jansen and Dick Wingate (various capacities and cuts) have managed to sift out the important treble sound, i.e. lots of cymbals, keyboards and Karla's very stretching vocal range, while keeping the drum and bass in an appropriate place. The rhythm section maintains a pure pop stance with the drums pounding out a shakin' beat and the bass adding character wherever needed.

DeVito herself guides powerhouse vocals over an interesting array of songs, the best being those by Danny Lawson ("Boy Talk," "Big Idea"), and old chestnuts. Of the latter, DeVito adds a welcome silliness to John Fogerty's "Almost Saturday Night," while managing to outclass the classic Grass Roots number "Midnight Confession" with updated verve and vigor.

When DeVito writes her own material, such as "Is This A Cool..." and the *Pirates of Penzance*—like "Bloody Bess," it is only slightly less masterful than the covers, but they demonstrate the woman's tremendous potential behind the pen. More than that, her own material displays an honesty about the way people and the world work, and that's what gives Karla DeVito's confidence away. Yes, Karla, this *is* a cool world, and you definitely have a very special place in it. E.Z.G.

JAZZ

THE NEW YORK SAXOPHONE QUARTET: *The New York Saxophone Quartet.* [The New York Saxophone Quartet, producers; Bernard Brightman, executive producer; recorded at J.A.C. Studios, New York, N.Y., March 1980.]

Performance: **Episodes from third stream**
Recording: **Well-balanced, but the pressing's a bit noisy**

Sometimes a group is in a bit too much of a hurry to get a record out.

What happens is that they come out with a record like this one—a record that would have benefited from a little more planning and more judicious choices.

To be sure there are some gems here of which the foremost is Phil Woods' "Three Improvisations." This three movement suite by saxophonist Woods is a fine illustration of what they used to call Third Stream. I don't remember who coined the phrase but it was used during the hey day of composers/improvisers like Gunther Schuller and John Lewis to indicate music that was both concert and jazz in discipline. To be sure the scores were written down and therefore frozen in place, devoid of the sort of improvisation that made jazz jazz in the classic sense of the word. And yet the best composers (Schuller, Lewis, Gil Evans, etc.) were able to catch the jazz spirit in their pens and transcribe it to the manuscript paper and the best interpreters were able to improvise in feeling to give a distinctive jazz quality to the finished work. I would certainly class Woods' "Three Improvisations," Eddie Sauter's "Q.T." and especially Gene Di Novi's "La Blues" as prime examples of the third stream. Somewhat less successful are interpretations by jazz musicians of pre-existing classic works such as Milan Kaderavek's "Introduction And Allegro" or Marcel Mule's saxophone orchestration of Isaac Albeniz' "Chant D'Amour." Also on the minus side are works such as William Kerr's "Three Jays and a Bee" and David Mathews' "Chantefleur." More enigmatic, but certainly musically interesting, is Calvin Hampton's "Bach's Fireworks Music." Hampton begins with a saxophone quartet orchestration of the Cannon at the Octave from Bach's "Art of The Fugue" which leads into Hampton's own composition. It is to Calvin Hampton's credit that there is no suspension of credibility at which point one realizes that here Bach ends and Hampton takes over.

Given the excellence of these players, and I can't stress their professionalism enough, the problem with this recording is repertoire. String players have the quartets of Haydn, Mozart and Beethoven to choose from. Saxophone quartets have no such treasury of masterpieces. In my opinion, the New York Saxophone Quartet would have done themselves a favor by waiting until they had enough compositions that would stand up to the test of

SPONTANEOUS COLLECTIVE COMBUSTION: ENJA JAZZ

By Nat Hentoff

The huge and yet ever-growing Polygram company is now distributing in the United States a distinctive series of jazz albums on the Enja label of West Germany. Particularly absorbing are two sets focusing on exceptionally high-level *collective* improvising. That is, there are no leaders. Each of the players gets—and most ardently merits—equal billing.

In the first celebration of truly hot jazz interplay, *Three For All*, the brilliant gestalt is made up of alto saxophonist and clarinetist Phil Woods, pianist Tommy Flanagan, and bassist Red Mitchell. They complement each other with special zest and respect because they all share a similar clarity of conception, sound, and beat—as well as enormous authority. Furthermore, each of these players is incisively lyrical without ever being in the least sentimental. Add to this common ground a constant freshness of ideas—a distaste for the expected—and you have one of the more classic, timeless jazz sets of this generation.

The six tracks are all originals (three by Mitchell, two by Woods, and one by Flanagan); and they range from gently penetrating ballads to loping swingers. All the music, every moment of it, has a depth of relaxation that can only come from complete mutual confidence in each other's playing—of the order of the once and former Modern Jazz Quartet, and very few other combos.

As for the engineering, I can't really conceive of any improvement, so lucidly and vibrantly is this three-for-all captured.

The other Enja date that stunningly reveals the extended potential of collective improvisation is *Hampton Hawes: Live at the Jazz Showcase in Chicago*. With the late fervent pianist were drummer Roy Haynes and bassist Cecil McBee. Although Hawes is listed as the leader, this album—like the Woods-Flanagan-Mitchell effort—has no "sidemen" in the usual sense of the term.

While this is one of Hawe's most continually inventive performances—his customary fire being heightened by more surprises than usual—the most astonishing presence here is that of Cecil McBee. It is not that the bassist dominates the album, but he is its very pulsing soul. With a sound that is simultaneously huge and sinewy, a beat like a tidal wave, and an unerringly logical conception, McBee creates an historic definition of postgraduate bass work.

It has become a cliche to say that Roy Haynes may be the most underrated master drummer among the modernists, but alas, he still is insufficiently recognized. On these performances, Haynes' characteristic crisp sound and swiftly apposite ideas complete this trio triumph. And here too, as in the other set, the emotive force of the playing comes from a thorough mutual relaxation.

A mono set, unlike the Woods-Flanagan-Mitchell album, the sound is clean and full. I am grateful to have music of this presence and imagination, and so I don't at all lament the absence of stereo.

PHIL WOODS/TOMMY FLANAGAN/ RED MITCHELL: *Three For All.* [Horst Weber, Matthias Winckelmann, producers; David Baker, engineer.] Enja 3081.

HAMPTON HAWES: *Live at the Jazz Showcase in Chicago.* [Joe Segal, producer and engineer.] Enja 3099.

time and repetition that is implicit in recording to make a full LP of material as consistent in quality as that by Woods, Sauter and DiNovi.

Given the limited spectrum of soprano sax, alto sax, tenor sax and baritone sax it should come as no surprise that the engineers got it all down. However, I found the surfaces noisy to the extent of detracting from the music, which is a shame when as much of the music is as good as this is. J.K.

WARREN VACHE: *Iridescence.* [Carl E. Jefferson, producer; Phil Edwards, engineer; recorded at Soundmixers Studios, New York, N.Y., Jan., 1981.] Concord CJ-153.

Performance: **The Jersey Jazzman meets the Duke of Detroit**
Recording: **Clear, clean, crisp, like an autumn day in New York**

When I first heard Warren Vache he was imitating Bix Beiderbecke for the New York Jazz Repertory Company. When I first heard Hank Jones he was a straight ahead bebop pianist. In the intervening years both players have managed to transcend these first impressions. Warren is the sweet singer, the melodic voice of jazz but with a drive and bite for which he can thank his teacher, the late Pee Wee Erwin. Hank is the orchestra filling in the harmonic gaps behind Warren much the same way as Duke Ellington did for the many singers and soloists he accompanied through the years.

This record is one tune freaks will hold dear. It's not often enough you hear songs like "Sweet And Slow," "No Regrets" or "Autumn In New York." The rest of the all-star group are pretty much in the bop element. George Duvivier's bass playing is too well known around recording circles to need much comment and those who remember drummer Alan Dawson from Dave Brubeck's Quartet will be amazed the way he holds it down here when the situation calls for that. Oh, there's still the occasional bomb blast, but for the most part Alan stays out of the way and swings nicely.

Hank Jones has also contributed a beautiful jazz waltz, "Iridescence," to the album. It's a work easily the equal of his brother Thad's, "A Child Is Born," so now we have one more excellent jazz work in three-quarter time.

But the piece de resistance has to be Warren Vache's version of Vernon Duke's "Autumn In New York." To find a rendition of this tune as moving as Warren's I'd have to go back to the old 1947 Frank Sinatra Columbia recording. That's what I mean when I say that Warren Vache is a sweet singer. The words and meaning of the song are there—implied and pervasive in everything Warren does. That he sings with a cornet or flugelhorn rather than his voice is incidental.

The record is helped in no small measure by the sound at Soundmixers and the excellence of Phil Edwards at the board. It seems, on ballads, as though there's no attack to the music at all. Instead of hitting a note, Warren Vache seems to uncover the note tenderly and caress the right one in the right place at the right time. On an up tempo (check out "The Song Is You") he can punch and jab with the best of them but even then he never jars the nerves of the music or the listener. J.K.

DICK WELLSTOOD: *Live at Hanratty's.* [Charlie Baron, producer; Bob Cubbage, engineer; recorded at Hanratty's, New York, N.Y. 1981.] Chaz Jazz CJ 108.

THE BLUE THREE: *At Hanratty's.* [Same production credits as above.] Chaz Jazz CJ 109.

RALPH SUTTON AND EDDIE MILLER: *We've Got Rhythm.* [Same production credits as above.] Chaz Jazz CJ 110.

RALPH SUTTON AND JACK LESBERG: *Live at Hanratty's.* [Same production credits as above.] Chaz Jazz CJ 111.

RALPH SUTTON AND PEANUTS HUCKO: *Big Noise from Wayzata.* [Charlie Baron, producer; Paul Stark, engineer; recorded at the Woodhill Country Club, Wayzata, Minn., July 15, 1981.] Chaz Jazz CJ 112.

RALPH SUTTON AND THE JAZZBAND: *Ralph Sutton and the Jazzband.* [Charlie Baron, producer; Minnesota Public Radio staff, engineers; recorded at the Lafayette Club, Lake Minnetonka, Minn., 1981.] Chaz Jazz CJ 113.

Performances: **Magnificent, in the Condon tradition**
Recordings: **Great at Hanratty's, but it gets worse the further west they move**

The musicians involved here—Dick Wellstood, Kenny Davern, Bobby Rosengarden, Ralph Sutton, Eddie Miller, Jack Lessberg, Peanuts Hucko, Ruby Braff, George Masso, Bud Freeman, Milt Hinton and Gus Johnson—have this much in common. They are all excellent jazz musicians. They also, for the most part, have been associated with either The World's Greatest Jazz Band of Lawson and Haggart or one of that band's two predecessors, Bob Crosby's Bobcats and Dick Gibson's Colorado Jazz Parties. Also with few exceptions they have been associated with various enterprises of the late Eddie Condon. They tend to center around New York City for the most part, the Colorado area in the case of Sutton, Johnson and Hucko and the West Coast in the case of Eddie Miller. Bud Freeman has been rumored residing here or there for the past several years since his exodus from the World's Greatest Jazz Band but he seldom sets down any place long enough to take roots and call it home. Most of these musicians, Wellstood being the most prominent exception, have recorded for Chaz Jazz in their previous series. Each series consists of seven records but this series has only six album numbers because the Dick Wellstood at Hanratty's is a double album. Others making their debut on the label are Rosengarten, Miller, Hucko, Masso and Freeman.

What this presents is a pretty much dependable, consistent level of musical orientation. It is not too surprising to me that on this current release one selection is played three times, "Ain't Misbehavin'" and three others appear in two different versions: "Honeysuckle Rose," "I Got Rhythm" and "Everybody Loves My Baby." Condon always had a weakness for tunes written by ("Ain't" and "Honeysuckle") or associated with ("Rhythm" and "Everybody") the late Thomas "Fats" Waller. Wallerism is an enthusiasm shared by pianists Wellstood and Sutton. I must point out, however, that in addition to chestnuts like "Honeysuckle Rose," Dick Wellstood goes to amazing lengths to dig up the unusual such as Sidney Bechet's "Ghost of the Blues," "Don't Let It Bother You" a Gordon and Revel tune recorded by Fats Waller and the highly unlikely album opener, a jazz improvisation on "Jingle Bells," another tune favored by Fats. The same kind of tune-freak's pleasure of discovery follows into the album by the Blue Three (Kenny Davern, Dick Wellstood and Bobby Rosengarden) which includes such unlikely material as Thelonius Monk's "Blue Monk" and Rudy Wiedolft's "Oh Peter, You're So Nice" but the gem of this album, in my eyes (and ears) is "Please Don't Talk About Me When I'm Gone," done here as a ballad.

Despite the presence of Kenny Davern, Eddie Miller, Jack Lesberg and Bobby Rosengarden on the Hanratty's albums Hanratty's is basically a piano room. Dick Wellstood is the featured artist most of the time with Ralph Sutton a close second and Art Hodes, Dave McKenna and others taking up the slack. The wonders that Bob Cubbage has done in preserving the sound and ambiance of the room is little short of amazing. One can only wish that Cubbage had been along on the outings in Wayzata and Lake Minnetonka, Minn. Both recordings are far from the standard set by the rest of the series. The pressings also seem to be a good bit less noisy than those of the first Chaz Jazz series. The liner notes fall to some of the veterans of the first Chaz Jazz series like Dan Morgenstern of the Institute of Jazz Studies and Marty Grosz, famed jazz guitarist and raconteur. There are also some surprises there, including notes by yours truly for the Jazzband album which united Braff, Masso, Davern, Freeman, Sutton, Hinton and Johnson for one of those great old time blowing jam sessions like the kind Eddie Condon used to broadcast on Saturday afternoons from Town Hall before he became a psuedo-prosperous night club owner.

Chaz Jazz records are available only by mail from Chaz Jazz Productions, Inc., Box 565, North Hampton, New Hampshire, 03862. The confusion is in the price structure. Ordering any one album the price is $10.95 each. If you order two the price is $9.95 each; three, $8.95 each; four, $7.95 each; five, $6.95 each; six, $6.45 each until you get to the rock bottom price of $5.95 each for seven or more records. It's an unusual way of merchandising yet Charlie Baron has found it to be a successful way of pricing his records and one which those who have purchased recordings from the previous Chaz Jazz release have proven to be popular. The cost includes packing and shipping. Please be sure to include check or account number and expiration date of your Mastercard or Visa should you wish to charge these items. Oh yes, and don't forget your name and mailing address so Charlie knows where to send the records! J.K.

CLASSIFIED ADS

TEAC 2A MODS, Bring your 2A into the digital/dbx age. Also, 1/3 octave pink noise test tapes with warble tones. Synthesizer calibrators and more. ARCAS Engineering, RFD #3, Burthill Rd., Winchester, NH 03470.

CATHEDRAL SOUND STUDIOS sells the best in Professional Recording and Reinforcement Sound Equipment. AB SYSTEMS, AKG, AMPEX (tape and recorders), ANVIL CASES, BEYER DYNAMIC, CLEAR-COM, COMMUNITY LIGHT & SOUND, dbx (150's in stock), ELECTRO-VOICE, FURMAN, HME WIRELESS SYSTEMS, KLARK-TEKNIK, OTARI, RAMSA, SOUNDCRAFT, SOUNDCRAFTSMEN, SYMETRIX, TAPCO, and a full line of bulk tape and cassettes, reels and boxes. We are also a full service duplicator facility. Call or write for a quote. 1575 5th Street, Rensselaer, New York 12144. (518) 465-5689.

ALLEN & HEATH MIXING CONSOLES for PA or studio. Full range dealer (2-42 tracks). Competitive prices. ICB AUDIO, POB 2752, ERLANGER, KY. 41018. (606) 491-2421.

FOR SALE: STUDIOMASTER 24x8 CONSOLE, 1 YEAR OLD, EXCELLENT BOARD, EXCELLENT CONDITION, $3500. Lee (414) 781-4493.

PRO SOUND SYSTEMS: Technics, Ramsa, Rebis, Revox, NEI, Tangent, Klark-Teknik, Gollehon, Ampex, others. ERS Inc., 1373 McLaughlin Run, Pittsburgh, Pa. 15241. (412) 221-2737.

TANGENT SERIES 16 Pro Recording Consoles, ERS Inc. (412) 221-2737.

HISSS driving you and your clients crazy? ELIMINATE IT! 4-dbx 150 stereo noise reduction units, purchased new September 1981. $275 each or $1000 takes all. Lee (414) 781-4493.

STEREO REVERB, SOUNDWORKSHOP 262 Lo-Z Parametric EQ, New, Craig (517) 485-7440.

Roland Microcomposer (MC-8). Must sacrifice. Contact Bruce (403) 732-4768.

Professional Sound Labs

110 N. Franklin St., Hempstead, N.Y. (516) 486-5813. Over 70 major lines of Pro Audio Products including: Urei, dbx, Otari, AB Systems, Ramsa, Sound Workshop, Audioarts, JBL Pro, Altec, AKG, Orban, Tascam 8516, Fostex and more. Large stocking inventory on Altec, JBL and EV diaphragms. Full repair and reconing center in house. Authorized warranty service for Altec, JBL, EV, PAS. Get your best deal and then call us.

FOR SALE: 2 Altec 604-ES in utility enclosures $600/PR.; 2 Altec 9846-B's w/horns & drivers but without 411 woofers $500/PR.; 1 Ampex 300 4-track with Sel Sync Switcher with electronics $1000; 1 Ampex 300 2-track with electronics $500; 1 3M M-64 4-track with Selsync-excellent condition $4500; 6 Altec 771-B Electronic crossover B1-Amps $100/each; 4 Altec A7 "Voice of the Theater" systems-excellent condition $200/each; 1 Eventide 1745M DDL w/3 outputs plus pitch shift-excellent condition $3500; 3 Altec N-1500A crossover networks $50/each; 3 Electroamplidyne SE-III equalizers $45/each; 1 Altec 1566A line amp $25; 2 Altec 356A "Voice of the Stars Amplifier" $50/each; 1 3M Selectake I Tape locator $500; 1 Haeco CSG II Compatable Stereo Generator $1500; 2 UREI 527 1/3 Octave Graphic Equalizer $475/each; 2 UREI 529 1/3 Octave Graphic Equalizer $525/each; 4 DeltaLabs DL-1 $800/each. CONTACT: FRANK TARSIA (215) 561-3660.

PART-TIME OPENING for experienced dedicated Christian engineer/sales: Sweetsong Recording, Box 2041, Parkersburg, W. Va. 26102.

AUDIO SERVICES-Best prices on EV, Tapco, Lexicon, Ampex Tape, Symetrix, Audioarts, White, Revox, QSC, Plate Reverbs, more. Write for quotes. P.O. Box 8220, Greensboro, NC 27419.

FOR SALE: Teac Tascam 80-8 Recorder with dbx and Model 5 mixer. Low hours, excellent condition $4000 or best offer. Call (814) 944-8550.

FOR SALE: PULSAR LABORATORIES RTA-150 Real Time Analyzer, CL2000 Compressor/Limiter, A1500P, A2500P, A3500P, A5000P Amplifiers, Monolith Speaker Systems, are available in the west and shipped nation wide from CALIFORNIA STAGING SYSTEMS, Box 2737 Dept. PTL-1, El Centro, Ca. 92244. For information or to order CALL OR WRITE TODAY! ASK FOR BILL.

16 track heads for M79 $750. 9 reels Ampex 2 inch 406/7, used one session, $80 each. (709) 753-9841.

AMPEX TAPE AT DISCOUNT PRICES, SINGLE OR QUANTITY AVAILABLE, IN ¼", ½", 1" or 2" WIDTHS. ALL PRO MIC LINES ALSO OFFERED AT UNBEATABLE PRICES. CALL OR WRITE ISLAND SALES, PO BOX 242, CLARKSBURG, MD. 20734 (301) 972-1653.

SPECK 800C 16x8x16 CONSOLE w/208 pt. p/bay & pedestal. Demo $6800. ERS Inc. (412) 221-2737.

PROJECTS! KITS! BUILD YOUR OWN QUALITY RECORDING, PA, LIGHTING EQUIPMENT and SAVE! AVI, Box 264-MR5, Buffalo, NY 14215.

CLASSIFIED

BUY
SELL
TRADE

- **PRODUCTS**
- **EQUIPMENT**
- **SERVICES**

Classified Rates
75¢ per word

Minimum 10 words. Copy must be received at Modern Recording & Music, 14 Vanderventer Ave., Port Washington, N.Y. 11050 by the 1st day of the 2nd month prior to cover date (for example, the April issue closes February 1st). Payment must accompany order. Phone numbers count as 1 word. Zip codes are free.

DISPLAY ADVERTISING

$115.00 per column inch.

Advertiser's Index

Ecoplate, Master Room, Valley People, Symetrix. Top dollar for trade-ins, we buy used gear. Audiohouse Box 24164, Denver, Colo. 80224. Phone (303) 751-2268.

BLANK AUDIO AND VIDEO CASSETTES: Direct from manufacturer: Below wholesale: Any length cassettes, four different qualities to choose from. BULK & REEL MASTERTAPE: From ¼" to 2". Cassette duplication also available. BROCHURE, ANDOL AUDIO PRODUCTS INC., 4212 14th AVE., DEPT. MR, BROOKLYN, N.Y. 11219. TOLL FREE 1-800-221-6578/N.Y. RES. 212-435-7322 Ext. 2.

Famous "InterFax Ground Fault Detector" solves grounding problems. $12.95 includes report of electrocuted musicians. 5 yr. warranty. InterFax, 5833 West Silver Spring, Milwaukee, Wi., 53218. (414) 462-8840.

CUSTOM LENGTH CASSETTE BLANKS made to order featuring AGFA and Magnetic Media tape. Any length from C-2 to C-122. For price list wirte OMEGATAPE, Box 39253M, Redford, Michigan 48239 or call (313) 532-8103.

MASTER THE RECORDING SCIENCE Sherman Keene's "Practical Techniques for the Recording Engineer" is a book about the real world of studio recording. Acclaimed by magazines, reviewers, college teachers, studio owners and engineers. 381 pages, 28 chapters (4 on computer mixing). To order send $29.75 plus $2.25 shipping plus 6% (Calif.) to: OpAmp Books, 1033 N. Sycamore Suite C, L.A., Ca., 90038. Order by phone (213) 464-4322.

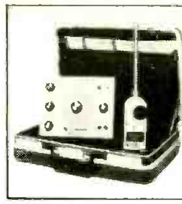
SHERMAN KEENE CORRESPONDENCE COURSE Author of acclaimed textbook "Practical Techniques for the Recording Engineer" invites you to study recording at home. Course includes reading and homework assignments in two textbooks with personal dialog via cassette. Eight lessons per level, three levels. $250 per level. For info write: Correspondence Course, 1626 N. Wilcox No. 677C, Hollywood, Ca. 90028.

SATIN PANTS, Etc. for entertainers etc. Write/call for catalog, enclose $1.00, refundable with order. **SATIN SALES.** Dept. 7, 7 South 5th Street, Minneapolis, Mn. 55402 (612) 333-5045.

FOR SALE: Ampex MM1100-1200 Head assembly, Excellent condition, $2000.00 or Best Offer. (415) 441-8934 or 527-6167.

FREE LIST OF USED EQUIPMENT FOR SALE. ALL TYPES. AVI, Box 264-R5, Buffalo, NY 14215.

Use
MODERN RECORDING & MUSIC
Classified Ads-
They Get
Results!



ACOUSTIC TEST INSTRUMENTS
USE TO ADJUST EQUALIZERS, OPTIMIZE SPEAKER PLACEMENT, ETC.

Complete Test Set	$449
Calibrated Sound Level Meter	$199
Fractional Dctave Pink Noise Generator	$339
Handbook on Acoustic Testing	$10

Free broch. HALL ENGINEERING. Dept R5
P.O Box 506. Martinsville, NJ 08836
(201) 647-0377

FOR SALE USED MINT: UREI 529, $485; Yamaha E-1010, $460; JBL 2231 15", $100 (4 ea); JBL K-110, $69. (2 ea); JBL 2205, $125 (8 ea); Beyer M260 microphone NEW $145. Barney O'Kelly, (614) 268-5643.

FOR SALE NEW: BGW AMPLIFIERS-model 600, $599 (12 ea); BGW 150, $450 (2 ea); BGW 250D, $585 (2 ea). JBL 2421, $195 (3 ea); JBL 2220-H, $120 (4 ea); JBL 2245H, $225 (5 ea). Phase Linear A-60 amplifier, $750 (6 ea); Phase X-20, $275. Ashly SC-22, $210; SC-80, $255. Shure SM-57, $99 each, SM-58, $119 each. Barney O'Kelly. (614) 268-5643.

WANTED: A few good studios and record companies who want the finest cassette copies from your master tape. Stereo cassette copies made in real time. Our quality and prices are the best. From 10 to 10,000. Call or write GG Productions, Box 24164, Denver, Co. 80224 (303) 751-2268.

WE HAVE IT ALL! Audio and Video Tape, Accessories, Recording Equipment, Fine Audio Components. KNOWLEDGEABLE Staff! COMPETITIVE Prices! FREE Buyer's Guide! SOUND IDEAS, Dept. MR, Box 671, Cary, NC 27511-0671 (919) 467-8462.

WOULD YOU LIKE TO LEARN THE GUITAR from the most respected teachers and performers playing today? Folk, Blues, Jazz, Bluegrass and many other styles are taught on cassette taped lessons. The teachers include John Renbourn, Stefan Grossman, Duck Baker, Dick Fegy and Eric Thompson. For your free catalog please write to Stefan Grossman's Guitar Workshop, Box 804, Cooper Station, NY 10003.

Wanted: McIntosh, Marantz, Tube amplifiers, JBL Hartsfield and Electro-Voice Patrician speakers. (713) 728-4343.

RECORD PRESSING Super Good Records Brilliant Jackets. FREE BOOKLET Tells How. WRITE: Outlet Recordings, Box 594, Rocky Mount, Va. 24151.

FREE RUBY: Rock & Mineral collectors join Mineral Sciences monthly mineral club and receive a genuine uncut Ruby from India FREE, along with the first of 12 beautiful mineral specimens, each with its own informative fact and data sheet. You'll receive a different specimen every month for one year. If dissatisfied just return the first mineral for a full and prompt cash refund. The Ruby is yours to keep, just for looking at this fabulous offer. A one year subscription is only $25. Mail check or Money Order to: Mineral Sciences Co. 400 East 73rd St. Suite 5E, New York, NY 10021.

FREE ADVERTISING for your used Stereo, Recording, PA, Musical equipment. Pay low commission ONLY IF YOU SELL! Free details: AVI, Box 264-M5, Buffalo, NY 14215.

The UREI Compressor/Limiters

To the audio professional, when a compressor or limiter is needed to tame the potentially disastrous consequences of uncontrolled level or to create special effects, one name stands out as the best: UREI.

Studio Standards for more than a decade, the compressors and limiters from UREI have earned their way into thousands of recording, mastering, and broadcast installations around the world.

Because we built our reputation for unparalleled professional performance and quality with our compressors and limiters, we have continuously advanced their engineering and technology to offer more reliability, features and performance. When you need the fastest, quietest and most flexible gain control instruments available, you can be totally assured that these products will prove to you why they've earned the title — Studio Standard:

The Model LA-4
A single channel, half-rack unit with patented electro-optical attenuator. Featuring smooth, natural sounding RMS action, it offers selectable compression ratios, a large VU meter, adjustable output and threshold levels and stereo coupling.

The Model 1176LN
A peak limiter which features adjustable input and output levels; individual attack and release time controls; selectable compression ratios; switchable metering; and stereo coupling. The 1176LN is the most widely used limiter in the world.

The Model 1178
A two channel version of the 1176LN in a compact (3-1/2") rack mounting design. Featuring perfect tracking in the selectable stereo mode, it additionally offers selectable VU or Peak reading meter ballistics.

From One Pro To Another — trust all your toughest signal processing needs to UREI.

UREI **From One Pro To Another**

United Recording Electronics Industries
8460 San Fernando Road, Sun Valley, California 91352 (213) 767-1000 Telex: 65-1389 UREI SNVY
Worldwide: Gotham Export Corporation, New York; Canada: E.S. Gould Marketing, Montreal

See your professional audio products dealer for full technical information.


PRECISION ENGINEERED FROM THE START
AUDIOARTS® ENGINEERING
A division of Wheatstone Corporation
Reinforcement and recording consoles by Audioarts Engineering
Wheatstone Project
8000 Series
44 Series
Signal processing equipment:
crossovers
limiters
feedback suppressors
equalizers
disc mixers
Wheatstone Corporation
5 Collins Road
Bethany, Ct 06525
203-393-0887