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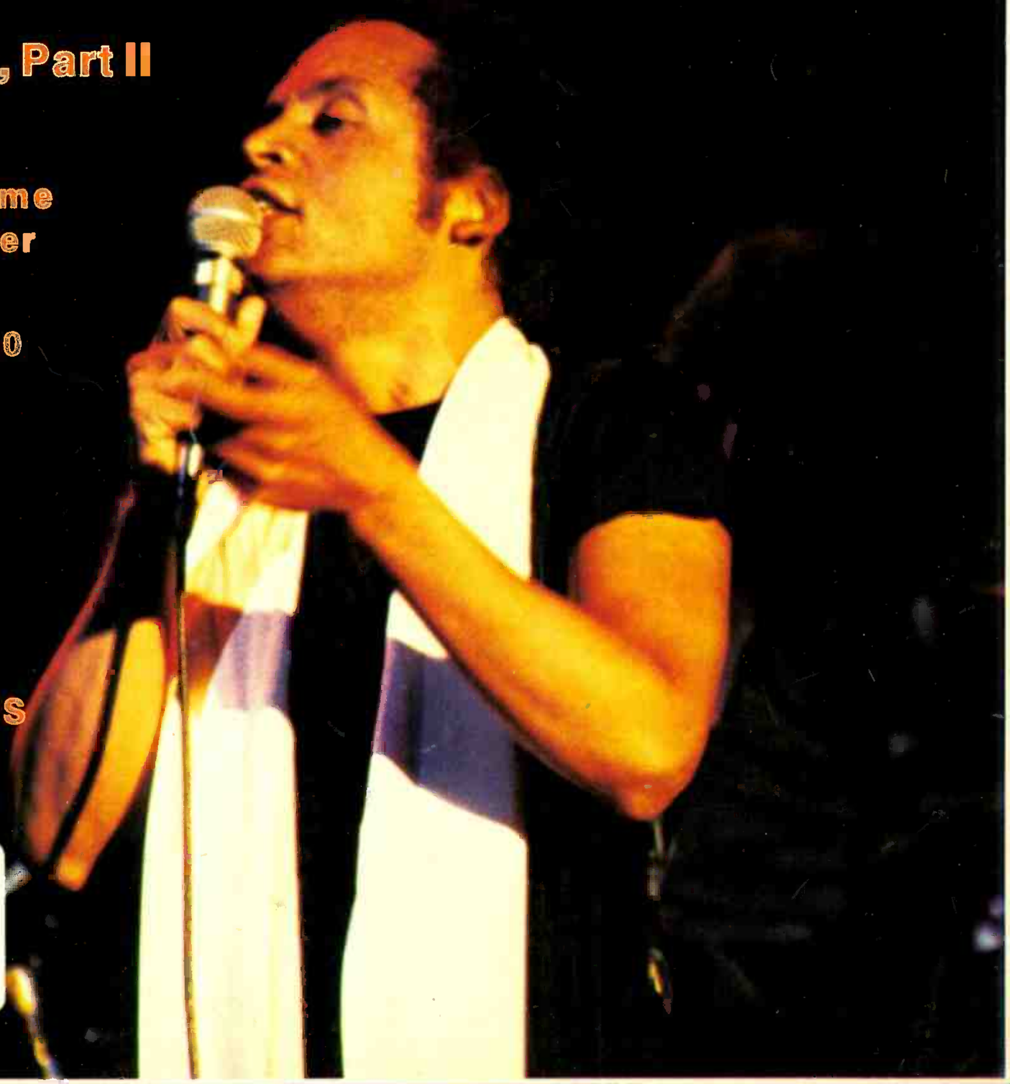
MODERN RECORDING & MUSIC

Vol. 7 No. 6
March 1982

Garland Jeffreys "LIVE!"



RECORDING TECHNIQUES, Part II

LAB REPORTS:

BSR(ADC) Real-Time Spectrum Analyzer

Sansui EQ SE-9B

Technics RS-M280 Cassette Deck

NOTES:

ASCAM
20

DUCTS

REVIEWS

The Unlimited Limiter.

In keeping with MXR's expanding commitment to the professional recording industry, our engineers have designed and built the Dual Limiter. A world class mono-stereo limiter offering total flexibility and ease of operation, the Dual Limiter produces a musically natural response in any compression-limiting application. All of this versatility is built into a compact, rack-mountable package.

The totally unique VCA's at the heart of the Dual Limiter provide an exceptionally wide dynamic range with low levels of distortion. Continuous bass distortion is much lower in level than typical compressor-limiters, allowing more freedom in setting release characteristics.

The Dual Limiter is also a forgiving limiter. Attack and release characteristics dictated by the front panel controls are modified by program dynamics and compression requirements. The slope increases smoothly past the threshold point, allowing a *gradual* transition into compression. Varying the Dual Limiter's threshold region produces a variety of intermediate slopes with the primary slope being that chosen by the slope switch. These features permit apparent dynamics to be maintained even though the dynamic range is being controllably limited.

The Dual Limiter's remarkable versatility is based on the fact that it can be viewed as two independent mono limiters that can be patched together via front panel switches for stereo limiting applications. Each channel has an In/Out switch, Slope switch, Input, Output, Attack and Release controls and an LED display, showing the amount of gain reduction. On the rear are both XLR and ¼" phone jack (ring-tip-sleeve) input and output connectors. Each channel's detector is accessible via rear panel phone jacks to permit external tailoring of the detectors' frequency response. This feature allows for de-essing (reduction of vocal sibilance) and a wide variety of frequency dependent limiting needs.

Because virtually every form of musical signal was used to evaluate the Dual Limiter's response during the initial stages of development, its sophisticated internal circuitry enables it to sound musically *natural* — even at extreme compression settings.

Balanced inputs, the ability to drive 600 ohm loads, +19 dBm input and output and standard rack dimensions (1¾" high) allow the Dual Limiter to be easily integrated into any professional system. With an extremely rugged case, metal knobs and reliable internal construction, the new MXR Dual Limiter reflects the highest professional standards and has been fully designed and built in the U.S.A.

The Unlimited Limiter — MXR's natural response to the question of performance and versatility in a space efficient and cost-effective package. See the MXR Dual Limiter at your nearest MXR dealer.

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The new Portastudio can record on all four tracks simultaneously. Which means you can make four-track recordings of live performances. And copy master tapes from one machine to another.

We also now have dbx* noise reduction. Which gives you more generations to work with without deterioration.

Advanced 2-band parametric type sweep equalizers on each input give more flexible tone control. And we now offer an optional remote foot switch for punching in and out.

So if you've been turned on by the Portastudio concept of a full-function multi-track cassette recording system, wait till you see how far the idea's gone now.

But then, with a name like "Portastudio," it could hardly be expected to stand still.

The New Tascam 244 Portastudio.

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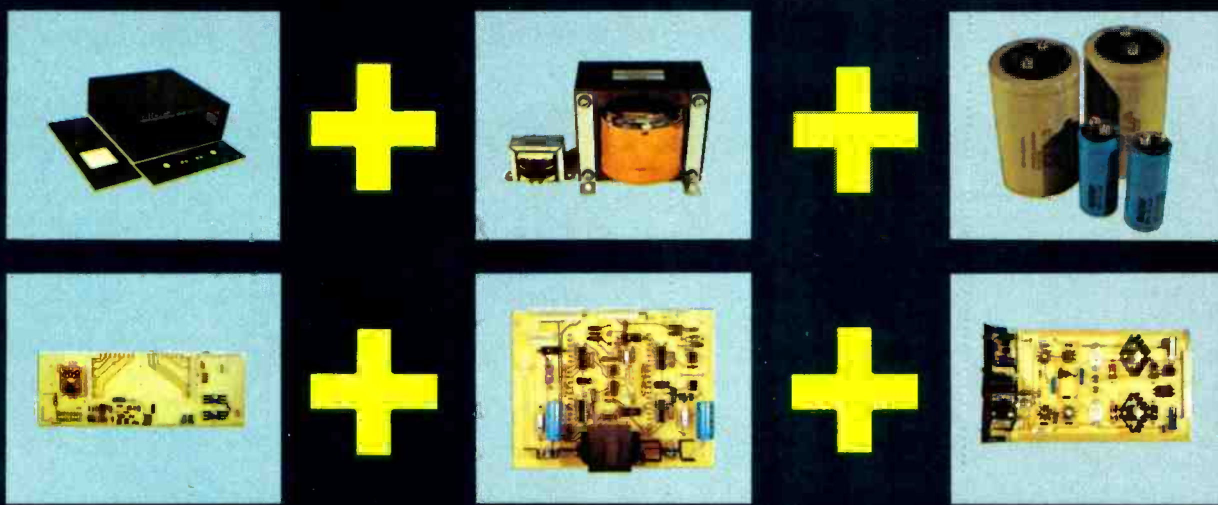

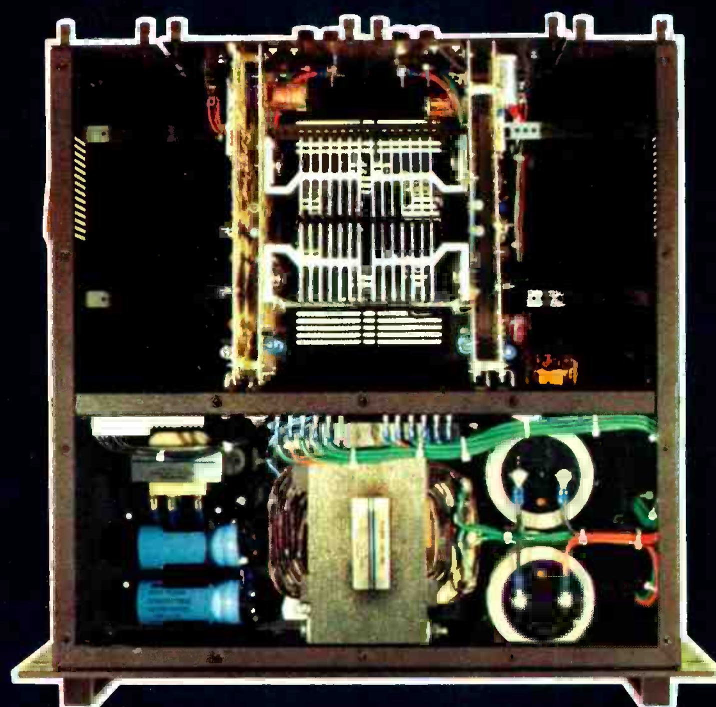

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MODERN RECORDING & MUSIC

MARCH 1982
VOL. 7 NO. 6

THE FEATURES

COMING NEXT ISSUE!
Todd Rundgren A to V
Profile: Michael Franks
An Overview of Synthesizers

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LETTERS TO THE EDITOR

Compression Debate

We can all imagine what Columbus felt like trying to convince the scientists of his time that the world was round. Can we possibly conceive of his frustrations if he had been happily conducting business in India—by sailing west—for many years and STILL returned to Europe to find their minds closed?

It seems that I have spent the first 20 years of my career making records and I will spend the NEXT 20 years trying to convince the public AND the press of the truth about the production process. Probably not since Columbus has there been such wide-spread misunderstanding (coupled with firm conviction) about the basic principles involved in a particular process.

I read Leonard Feldman's piece in the January issue about the CX process. I am certainly not going to join this debate because both sides of the argument are well fortified with protagonists already. However, I must respond to a side issue which emerges from his discussion. "Let's face it," he says, "recording engineers have been compressing sound...ever since...the early 1900s." Now, while that statement is probably true, the inference that ALL records have been compressed (so what's so bad about a little CX between friends?) is definitely NOT true.

For years I have heard from "experts," audiophiles, performers and consumers about the standard compression or limiting that goes into every long playing record. For years I have been telling them that it is not true, but here I am in 1981 having to repeat my argument again and again. Of course, I am not referring to "pop" records—but, I believe, neither is Mr. Feldman.

No "classical" record producer that I know inserts ANY electronic compressor or limiter in the mixing of a master tape and while my major contact with a disc cutting facility has been with Columbia Records, I feel fairly confident (knowing the goals and principles of the Producers involved) that NO disc mastering studio inserts any compressing or limiting in the process of transferring a "classical" master tape to lacquer.

As far as "hand compression" is concerned, this is either nonexistent or used so sparingly that it is not worthy of inclusion in this discussion. Mr. Feldman refers to "the guilty parties" criticizing CX because it compresses sound in an orderly way while they compress it in a random fashion. This is very tricky rhetoric and besides being based on an assumption that is false, it exhibits little logic. The plain fact is that the listener is not aware of what a theoretical or ideal dynamic range is. He only knows what is served up to him by the Producer and Engineer of a particular recording. If it *sounds* satisfactory then it *is* satisfactory. No matter how "orderly" the compression visited upon a non-decoded playing of a CX record is, if it *sounds* compressed then it is not satisfactory. If a record producer sanctions the shaving of 1 dB off of a single "peak" in a 20 minute movement, then it is not compressed in any practical sense—nor does it sound compressed, nor does it ruin the listeners' enjoyment of the performance, nor is the addition of any out-rigger expander called for.

By the way, if a listener *does* employ an expander on playing a record, then he surely will be artificially increasing the dynamic range of the recording, and who says that THAT'S kosher? There was a humorous axiom in the business of adjusting the frequency response of a tape recorder: "Better 5 dB *up* at 10 kHz than 1 dB *down*!" Are we now to apply this joke to dynamic range? "Better 10 dB artificial expansion than 1 dB artificial compression."

Just one more point. Lest I leave myself open to a barrage of comments on the question of subjective dynamic range. When I say "If it *sounds* satisfactory..." I am referring, of course, to the opinion of experienced, professional listeners. The fact that many amateur listeners may not hear or object to a large amount of compression does not excuse its use. As professionals we must strive to meet the expectations of the highest denominator and not slip by by catering to the inexperience of the lowest.

—Andrew Kazdin
Jamaica Estates, NY

Len Feldman replies to Mr. Kazdin's letter:

Mr. Kazdin's protestations are, primarily based upon at least two (and possibly a greater number of) misconceptions. Mr. Kazdin, first, assumes that I said that *all* commercial recordings are "compressed." I have carefully reviewed my AMBIENT SOUND column appearing in the January, 1982 issue and nowhere could I find the word "all" used in that connection. I recognize full well that there are a dedicated group of recording engineers and producers who try to capture as much of the dynamic range of live music in a vinyl recording as is possible, given the inherent constraints of the medium itself.

Mr. Kazdin then goes on to say that he is not, of course, referring to "pop" records (but rather to classical). As a producer of classical records, Andrew Kazdin must surely be aware of the fact that classical record sales constitute between 3% and 5% of the entire record business in the United States (it's somewhat higher in Japan and in some European countries, but still represents a small percentage of record sales even in those countries). Logic suggests that if Mr. Kazdin agrees that pop records are compressed during mastering, then between 95% and 97% of records made and sold in this country *are* arbitrarily compressed. On that basis alone, I would have been justified in generalizing or making an all-inclusive inference with regard to record compression which, I repeat once more, I did not actually do.

I suppose I could rest my case at this point, but since Kazdin went further, let me do the same. I never said or implied that critical listeners would not be able to detect compression in a CX-encoded record that is played back without CX decoding. But at least in the case of a CX-encoded record, critical listeners have the option of using a CX decoder, by means of which they will not only restore amplitude linearity but will experience a total dynamic range that is very close to the musical truth. No single-ended expander, applied to conventional compressed discs (excuse me, compressed "pop" records) can do as much for the very reasons that I first stated in my column—the compression is *arbitrary* and varies from engineer to engineer and from studio to studio. It seems to me that CX, if nothing else, offers exactly the alternatives that we, as professionals, demand: the ability to "meet the expectations of the highest denominator" (through the use of a decoder) *and* the ability to cater "to the inexperience of the lowest" at a cost savings, by allowing non-discriminating listeners to play CX discs without a decoder and still derive some pleasure from the music.

—Leonard Feldman
Technical Editor
Modern Recording & Music

Pressing Questions

Could you tell me what companies manufacture record pressing equipment?

—Terry Jenks
1804-C Lyman
Hutchinson, KA

We recommend that you get the Billboard International Buyer's Guide. *In it is a listing of plating, processing and pressing plants for every state in the United States. That listing is in the Services Section of the guide. You can obtain a copy of this guide by writing to Billboard Publications, Inc., 9000 Sunset Blvd., Los Angeles, California 90069. Their phone number is 213-273-7040.*

International Equalization— —Red Tape?

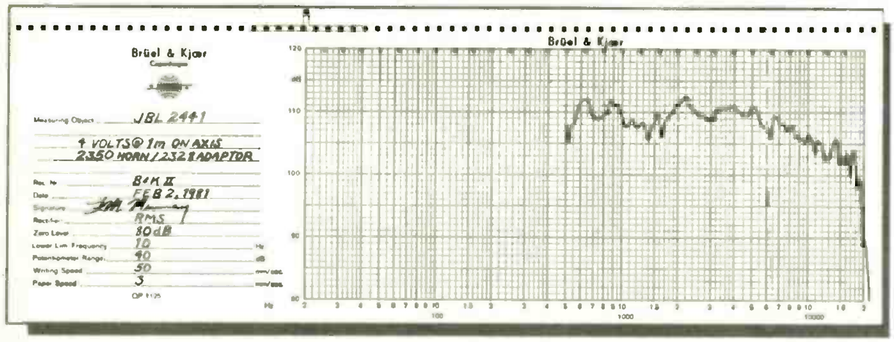
I read the letter in the December "Talk Back" column written by Mr. Ohba of Nakamichi Research, and I thought that I could shed some light on the playback equalization controversy. I spoke to Mr. Ohba about this matter, and we both agreed that information from a calibration tape manufacturer might clarify the issue.

Mr. Ohba's thorough description of the equalization process is accurate in every detail. An interesting point that might not be clear, however, is that the only way to measure magnetic flux on a tape is to measure the voltage induced across a head. When the German DIN standards established the 120 us calibration tape standard, BASF and Philips used the best ferrite heads available at the time (mid-60's) as reference heads. It is always an uncomfortable fact that the time for initial standards is also the time when little information is available and equipment is relatively crude.

When the cassette came of age, vastly improved heads, especially the Nakamichi crystalloy head with its incredibly small gap, showed how accurate the original reference head was. The calibration standard had too much high frequency compensation added. The new heads could better resolve the short wavelength flux and produced a rising high frequency response. In 1974 DIN decided to reduce the level of short wavelength flux on the calibration tapes but remain close to the original but technical "wrong" standard in order to maintain compatibility. DIN also made several other minor changes over the years, but Japan was never fully informed about what was happening in Europe. A great deal of misunderstanding arose from the lack of technical communication.

What everyone needed was communication and cooperation on an international basis. The IEC (International Electrotechnical Commission) was established to provide a forum and to set "*the* accepted standards throughout the world," as Mr. Ohba points out. BASF and TEAC worked together on the question of calibration accuracy and compatibility, and both companies manufacture the IEC calibration tapes used to align the heads and playback amplifiers of cassette recorders for flat frequency

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



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

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

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

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

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

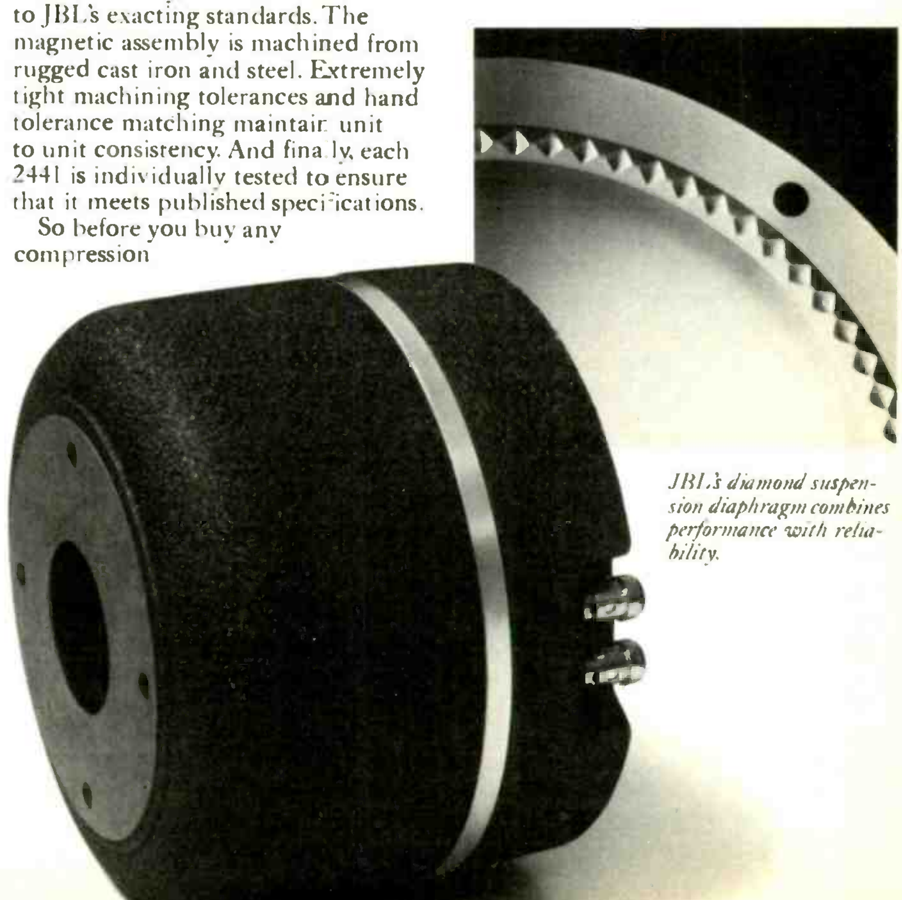
So before you buy any compression driver, ask your JBL professional products dealer about the 2441. It'll deliver a lot more than just an impressive frequency response.

1. Patent Applied For

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

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

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



JBL's diamond suspension diaphragm combines performance with reliability.

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response at both 120 us and 70 us equalization.

Playback EQ can be a complicated matter because mechanical azimuth misalignment can easily disguise the electrical accuracy of the tape and the amplifier. Nakamichi's ability to resolve incredibly short wavelengths for extended high frequency response is due to the design and finish of heads with extremely small playback gaps and not to "tricks" with equalization. The IEC calibration standard manufactured by BASF will show flat frequency response on all Nakamichi recorders produced for the last few years. This compatibility assures complete compatibility with all other recorders adhering to international standards.

—Terence D. O'Kelly, Manager
Technical Marketing Services
BASF Systems Corporation
Bedford, MA

Back Issues

After delving into electronics for some years as a hobby, I have decided to combine my favorite two hobbies, electronics and listening to music, into a part-time career. Having read your publication occasionally for four or five years because of its article on the use of electronics in the music industry. Since I decided on being a recording engineer, I felt it very important to read every single issue of your magazine because of the excellent articles and stories on the techniques and use of equipment in the music industry. Unfortunately I started reading every issue too late to read the issues which featured the following articles: "Multi-Track Magic" by Craig Anderton and "Studio Notebook #1" by James F. Rupert. I would like to know if these articles are available as a reprint or back issues. Thank you for your publication.

—George Carty
Brooklyn, NY

You can get back issues sent to you by mailing Modern Recording & Music *$2.20 for each back issue, plus .65 for postage and handling.*

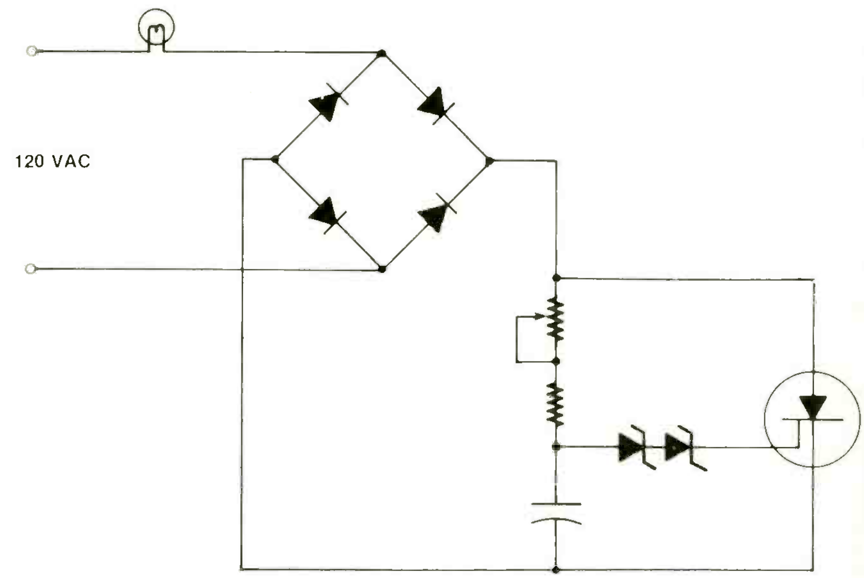
The "Multi-Track Magic" articles by Craig Anderton appeared in May, July and August 1981. "Studio Notebook #1" by James Rupert appeared in June 1981.

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.*

A TYPICAL SCR DIMMER CIRCUIT LOOKS LIKE THIS:



Brightening a Dim Situation

We have recently opened an 8-track studio here in Londonderry, Vermont, and are having a noise problem which our local electricians have been unable to solve. We hope you will be able to help us as this noise is creating some severe recording problems.

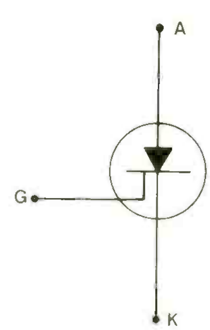
There are light dimmer switches located in the same table console as the mixing board. When we dim one or more of the switches, it throws an audible hum into our system, and the hum gets recorded.

Is there such a thing as a quiet dimmer? Is insulating the electrical wires the answer? We are not electrical wizards here, but we are willing to try anything to alleviate this problem.

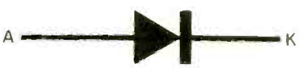
—Alan Joe Halstead
Production Manager
Creative Commercials
Londonderry, Vermont

Light dimmers are the noise demons of small recording studios. Electricians install these dimmers because they are not familiar with the electronic properties of the typical off the shelf SCR dimmer. SCR stands for silicon controlled rectifier. *SCR dimmers must not be used in any recording studio.* Even if you went to all the trouble of isolating your AC and ground systems according to my August 1981 *MR&M* talkback reply (see "Wired for Sound," page 16), you would still have noise problems on the AC system if you use general purpose dimmers.

In order to show you how SCR dimmers produce noise, we must first examine how an SCR works and then examine how SCR dimmers are built. The symbol for an SCR is:

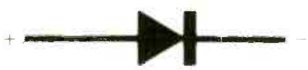


The A stands for anode, the K stands for cathode, and the G stands for gate. The SCR is a modified diode. The symbol for a simple diode is:



A diode consists of two pieces of silicon fused together to form a PN junction. One piece of silicon contains free negative charges and the other free positive charges as shown.

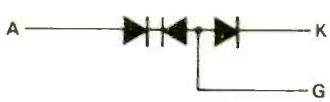
Current flows easily through a diode if it is forward biased. A diode is forward biased when the negative terminal of a battery is connected to the cathode and the positive terminal is connected to the anode. Current will not flow through the diode if it is reverse biased (the battery terminals are reversed).

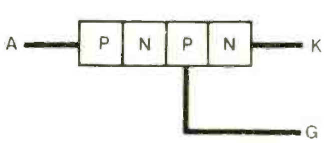


Forward Bias

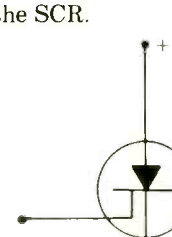
Reverse Bias

An SCR is composed of four pieces of silicon fused together. It appears as three diodes in series. An extra terminal is added to the second P layer to form the gate as illustrated.



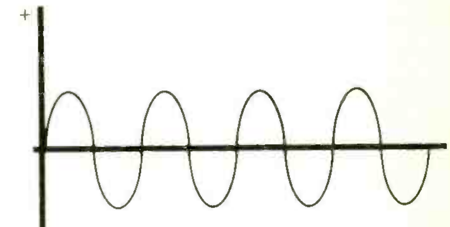


When an SCR is forward biased, it will not conduct current until current is injected through the gate. The turn on is rapid and continues until either current is removed or the polarity is reversed. Once the current is already flowing, the gate has no effect on the current flow through the SCR.

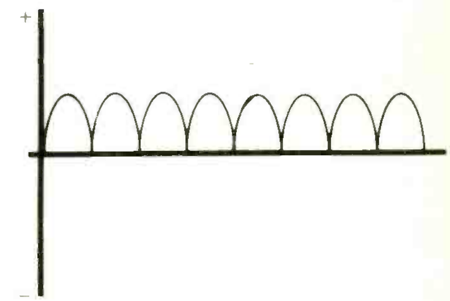


Forward Bias

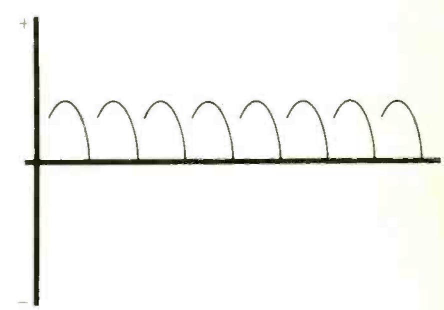
The four diodes in the SCR dimmer switch form a full wave bridge rectifier which converts the 120V AC sine wave into pulsating DC.



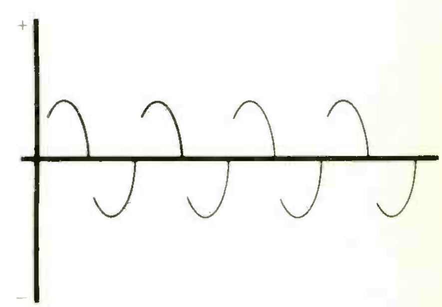
120 Volt Sine Wave



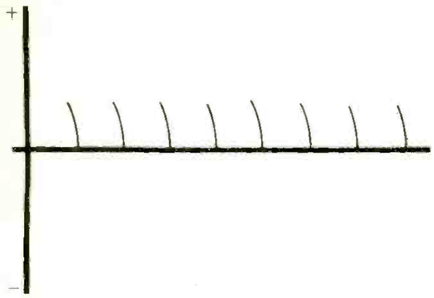
The components connected to the gate determine the turn on point for the SCR. If the dimmer is set for bright light, the current through the SCR will look like this:



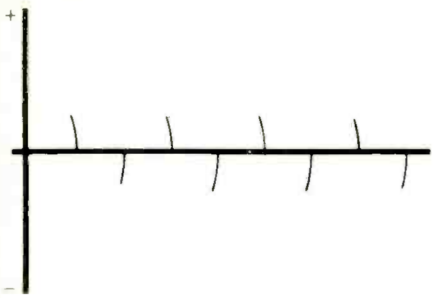
The current through the light will look like this:



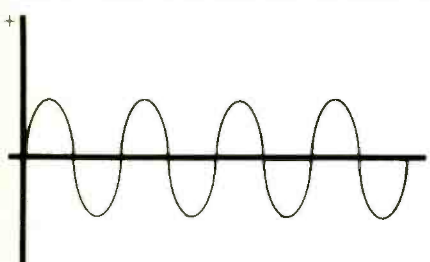
If the dimmer is set for low light, the current through the SCR will look like this:



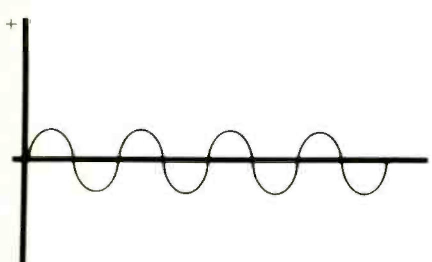
The current through the light will look like this:



The SCR turns off each time it drops to zero volts. It is the subsequent turn on that causes a pulse to appear on the AC line. Since this turn on occurs twice during each cycle of the 120V AC sine wave, the buzz that is created is 120 cycles. These turn on transient spikes are what cause all the headaches for studio owners. A 120 Hz buzz on the audio is not very pleasant to listen to.



Bright



Dim

Studios should use an autotransformer type of dimmer control. They are expensive and bulky. However, it is obviously necessary if you wish to have dimmers in the studio. The autotransformer consists of a continuously variable tape which picks the AC off at different points of the winding. There is no switching involved. The sine wave simply varies in amplitude.

—George Juodenas
Audio Architects
Nashville, Tenn.

Springtime for Germany

I greatly enjoy Craig Anderton's projects whether I find them in *Electronic Projects for Musicians, Home Recording, Polyphony, Guitar Player* or my favorite, *MR&M.* I've built many of them, and I've been quite successful with my efforts.

However, when I began to work on the "Hot Springs" Reverb project (*MR&M,* October 1980 issue) I encountered some difficulty. Specifically, it isn't possible to get the Accutronics springs mentioned in the article here in Germany. Can you kindly give some advice to your overseas readers how to modify (either by calculation or experimentation) the "tuning parts" (C1 and R9) to match with other springs?

—Jurgen Sprotte
Tonstudio
Rembrucken, Germany

We called Craig Anderton to see if he would be able to provide such a "modification" for you in the space we have available here. Unfortunately, Craig felt that such a revision would not be in your best interests. Apparently, the relationship between the springs and the tuning capacitor and resistor is a very critical one. The springs are tricky to tune, as well. Craig selected the Accutronics springs for their unique, "unspring-like" sound, which is the very reason the "Hot Springs" is such a phenomenally popular project. Even if he were to provide you with some alternatives, Craig would not be able to guarantee the quality of the sound you would then get from the reverb. Based on the these considerations, Craig felt it would be to your best advantage to order either the Accutronics springs or the entire kit from PAIA Electronics in Oklahoma City. A subsequent call to John Simonton of PAIA brought some better news for you. The kit and/or just the springs is available from PAIA Europe (based in Sweden, this rep also services Germany). They are marketing these items for the same prices quoted in the magazine (mailing costs, of course, might differ): the kit is $59.95 (specify #6740K), while the springs sell for $22.95 each. (A circuit board is also available for $7.95, ask for #6740-PC.) PAIA Europe can be reached c/o Wetab, D.R. Kristinas vag 31, 19300 Sigtuna, Sweden.

While we certainly understand the soul of the DIY'er, it probably is best for you to take this route. Hope this info does the trick!

Dokorder Diagnosis

[We received the following in response to James M. Lindsay's question regarding his Dokorder 1140. See "Dr. Dokorder," December 1981 Talkback column, page 24.]

I own two Dokorder 1140's (both superb machines), but I too had trouble with the multi-sync function—on *both* machines. From what you describe, I'm willing to bet that your problem is just where mine was on both machines: the "multi-sync circuit board" which is located directly under, and in back of, the head block.

The problems were two-fold. First, check intermittent capacitor C069 (100 mF/16 V) which is common to all four channels. Replacement eliminated the "static" sound and, in my case, oscillation build-up. Also check for poor solder connections (especially through-the-board foil connections). Re-heat them, adding solder as needed.

It took me a long time to find the trouble, but once it was corrected, I had myself two excellent, dependable machines—wouldn't trade them for any others. Hope these suggestions solve your problem.

—No name given
Fargo, N.D.

An Existential Muse?

I have two Muse binary sequencers which are in need of repair. The manufacturer, Tridex, is out of business, so no help can be gotten from that corner. Can anyone turn me onto someone who can repair these things? I am at a dead-end!

—Art Noel
Havertown, Pa.

Hmmm. Looks like we've joined you at the dead-end. Since we've come up dry, we'll once again ask our readers. Can anyone out there help Mr. Noel on this one?

THE PRODUCT SCENE

By Norman Eisenberg

FOSTEX MIXER; EQUALIZER

Fostex offers a new line mixer, the model 2050, designed for use in recording or mixing music, assembling sound tracks and in sound reinforcement. The device is capable of generating extra cue and monitor mixes instantly, and it also serves as an inexpensive keyboard mixer. A performer, for instance, playing in front of an audience, can use the 2050 and headphones "pre-listen" for tuning, sequencing and patching, before the audience hears the performance. During a rehearsal, the stereo output can be taken from another mixer or from a cassette player and mixed into the performance. "Live" instruments also can be mixed into a prerecorded program. Stacking 2050s also permits creating separate P.A. and recording mixes; separate house, monitor and recording feeds; or additional cue and monitor mixes. Basic layout includes eight line-level inputs (tape/effects), each with its own foldback which lets you re-route the input signal to another device as well as sending it through the 2050's own mixing circuitry. Each input chain has its own gain and pan controls.

The Fostex 3030 is a stereo graphic equalizer with ten bands on each channel providing 12 dB of boost or cut with click-stops. ISO frequencies run from 31.5 Hz to 16 kHz. Specs include S/N of better than 90 dB; THD, less than 0.03 percent; headroom of 25 dB. Each input jack has a foldback for sending the unequalized signal someplace else.

CIRCLE 1 ON READER SERVICE CARD

MODULAR MIXING SYSTEM

Canflex is a new modular mixing system offered by G.B.A. Inc. of New Rochelle, N.Y. According to the company, Canflex offers all the good features of a modular desk without the limitations of non-modular equipment, and at the price of non-modular equipment. "There is no expensive limiting main frame," says the manufacturer, so the variations are endless." Canflex comes in two formats, "standard" and "pro." Channels from either format can be mixed together with master modules, either 2 or 4 output formats. The Canflex board can be "broken" anywhere, and "stretched" at a later date to add more channels, master EQ, electronic crossovers and so on.

CIRCLE 2 ON READER SERVICE CARD

NEW LITERATURE

The complete Audio-Technica 800-series line of microphones and related accessories is presented in a 4-color, 24-page catalog. It's available at A-T mic dealers, or by writing directly to Audio-Technica US, Inc., 1221 Commerce Drive, Stow, Ohio 44224.

CIRCLE 3 ON READER SERVICE CARD

POWER AMP HAS BUILT-IN LIMITER

The new model A15 is a power amp in Phase Linear's pro series which is equipped with two separate precision variable-slope limiters for 15 dB of overload protection beyond rated power level which, says the manufacturer, provides the equivalent of more than 2,000 watts per channel of clipping headroom. This, explains PL, allows the A15 to be used to power a compression driver in a bi-amped system without danger of clipping or voice-coil damage. Exclusive control circuitry is said to be configured to make the limiter completely error-free.

Occupying a single rack-mount space, the A15 is rated at 65 watts per channel into 8 ohms; 100 watts per channel into 4 ohms. THD is listed as less than 0.05 percent from 20 Hz to 20 kHz. Features include balanced/unbalanced inputs; automatic mono input; calibrated level controls; indicators for fault, signal present and thermal overload. Construction and design are said to be well suited for fixed installation and touring applications.

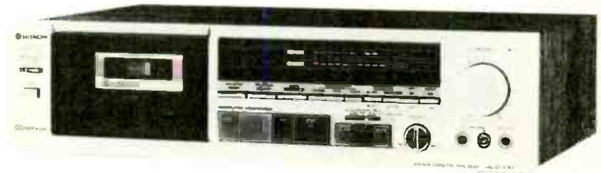
CIRCLE 4 ON READER SERVICE CARD

CREST AMPLIFIERS

Three new power amps in the Crest series have been announced by DMI Inc. of Hawthorne, N.J. The model 2500S is spec'd for 125 watts per channel for 8-ohm stereo; 200 watts per channel in 4-ohm stereo use; 300 watts per channel in 2-ohm stereo; 350 watts in 8-ohm mono use. Balanced and unbalanced inputs are provided. The model 3500S offers 225 watts per channel in 8-ohm stereo; 400 watts per channel in 4-ohm stereo; 800 watts in 8 ohm mono. Styling and general features are similar to those of the 2500S. Hum and noise of either amp is −100 dB below rated output, 20 Hz to 20 kHz. The model 4000 amp offers 350 watts per channel in 8-ohm stereo; 600 watts per channel in 4-ohm stereo; 800 watts per channel in 2-ohm stereo; 1100 watts in 8-ohm mono; 1600 watts in 4-ohm mono. Hum and noise are rated for −110 dB from 20 Hz to 20 kHz below rated output.

CIRCLE 5 ON READER SERVICE CARD

HITACHI OFFERS FIVE DECKS WITH DOLBY B AND C



To the model D-E57 cassette deck, its first to include both Dolby B and Dolby C, Hitachi has added four new decks that also offer both versions of the noise-reduction system. Top-of-the-line is the model D-2200M, a three-head, direct-drive unit with dual capstan drive, and Hitachi's ATRS (automatic tape response system) which uses a microprocessor and internal test circuitry to optimize bias, EQ and recording sensitivity for each tape used. Specs include S/N ratio of 69/75 dB (with Dolby B/C); frequency response with metal tape of 30 Hz to 20 kHz, ±3 dB; wow-and-flutter (WRMS) of 0.019 percent. An elapsed-time electronic counter is featured. Price is $750.

CIRCLE 6 ON READER SERVICE CARD

A step down in the new line is the model D-E99 ($570); and next in the line is the model D-E66 ($400). All three decks use separate heads for record and play (housed in a single unit). Specs for the D-E99 include the same S/N as for the D-2200M; frequency response with metal tape of 30 Hz to 19 kHz; wow-and-flutter (WRMS) of 0.032 percent. Specs for the D-E66 include: the same S/N again; the same response as for the D-E99; wow-and-flutter (WRMS) of 0.038 percent.

The fourth deck, model D-E33 ($200) is a two-head unit. The original Hitachi deck that included both Dolby B and Dolby C is the model D-E57 ($370), a two-head machine with DRPS (digital random program selector), which can select up to nine selections programmed on the tape in either forward or reverse. Specs for the D-E33 include: S/N of 73 dB with Dolby C; frequency response with metal tape of 30 Hz to 17 kHz; wow-and-flutter of 0.04 percent.

CIRCLE 7 ON READER SERVICE CARD

DOLBY C IN THREE-HEAD DECK

New from Onkyo is the TA-2070 cassette deck in which both Dolby-B and Dolby-C are included along with a host of features and options characteristic of the high-end type of cassette recorder. Among these are a direct-drive, dual-capstan, three-motor tape transport; three heads; feather-touch controls; and the newest version of Onkyo's "Accubias" system which automatically fine-tunes the recorder's bias current to the optimum level for the tape being used. The unit also can be set to "remember" the bias setting for a given tape, even when power is turned off.

Metering is by fluorescent "bar graph" scales with peak-hold; the tape counter shows real time; recording calibration controls are up front. An "auto space" option allows you to put five seconds of blank area onto a tape during recording, and then puts the deck into record-pause mode for subsequent recording. Three positions of tape selection are provided. For metal tape, response is listed as ±3 dB, 20 Hz to 19 kHz; signal-to-noise with Dolby-C, 80 dB; wow-and-flutter, 0.021 percent (WRMS). List price is $700.

CIRCLE 8 ON READER SERVICE CARD

MIXER/RECORDER COMBINATION

In a move which should entice musicians, recordists and audio/visual specialists, Studiomaster has announced the Studio 4. The unit is a 4-track cassette mixer/recorder combination. A 19-inch rack-mountable affair, the Studio 4 supplies the user with six balanced mic input channels (switchable to line inputs); three-band semi-parametric EQ; two aux sends; and direct in/out effects jacks. The recorder section features a head design which allows for simultaneous 4-track recording utilizing full-time Dolby "B" during playback and record. The transport mechanism is of heavy duty design with a Hall effect capstan motor and a DC reel motor. Also, the unit has remote recorder control capabilities and foot-switchable punch in/punch out facilities.

CIRCLE 9 ON READER SERVICE CARD

STUDIO MONITORS

Designed for large scale studio installation where extremely high sound pressure levels are required are two new four-way speaker systems in JBL's Professional Series. The model 4345 is engineered for either full range or biamplified operation and is designed in pairs with mirror-image driver configuration for good stereo imaging. Drivers include an 18-inch woofer (the model 2245H with a 4-inch diameter voice-coil and 20 pound magnetic assembly); 10-inch midrange; high-frequency compression driver bolted to a horn/lens assembly; ultra-high frequency unit fitted with a diffraction horn. The 4345's enclosure has separate chamber and sub-chamber for the low-frequency and midrange speakers.

The model 4355 is designed specifically for biamplification. Its two 15-inch low-frequency speakers, mounted in a ported enclosure, are driven independently of the other speakers which include a 12-inch midrange in a sealed subchamber, a high-frequency compression driver fitted with a horn/lens; and an ultra-high frequency compression driver load fitted with a diffraction horn.

The model 4345 has a maximum rated output of 120 dB/SPL; the model 4355, 126 dB/SPL.

CIRCLE 10 ON READER SERVICE CARD

COMPRESSION DRIVER

Renkus-Heinz, Inc. has added the model SSD 3301 to its relatively new but extensive line of professional audio components. The newly designed two-inch throat driver is a companion to the better-known SSD 3300 driver, and features an extended frequency range to 20 kHz. Created for two-way systems the SSD 3301 is said to deliver natural sound at very high pressure levels.

CIRCLE 11 ON READER SERVICE CARD

NEW FROM DBX

New from dbx is the model 180 Type 1 noise-reduction system. Intended for use with professional two-track tape machines, the model 180 provides two channels of encode and two channels of decode electronics. The 180 is designed for installation between the console or mic mixer and the line-level inputs to a tape deck. Says dbx, the model 180 can produce a stereo master tape that "fully preserves the dynamic range of live music and is completely free of audible hiss as well as distortion due to tape saturation."

The company also has announced the F-900U frame, which is the unpowered version of the dbx F-900 frame. Requiring an external power supply, the F-900U frame will accommodate up to eight active signal processing modules in the dbx 900 series. All units are of standard rack-mount width.

Of related interest is a new circuit chip or integrated circuit (IC) developed by dbx. The chip, which contains the active circuit components for stereo dbx noise-reduction, operates from 3 volts of power, and is expected by dbx to encourage the manufacture of a broad variety of consumer products that will be capable of reducing tape noise by 50 dB, and "without level matching requirements and with less dependence on flat frequency response than other noise reduction systems." First product expected, using the new chip, is a new decoder for dbx-encoded discs priced for mass marketing.

CIRCLE 12 ON READER SERVICE CARD

DOD SOUND PROCESSORS

A stereo double flanger, the model R-870, is offered by DOD Electronics Corp. of Salt Lake City. An analog delay processor, the R-870 is capable of flanging, vibrato, chorus, doubling, slap back and reverb effects.

The DOD model R-880 Dual Relay is intended for generating echo and reverb effects and is offered for use in mono or stereo P.A. applications. Using both companding and emphasis techniques, the unit provides delay times of 12 ms. through 500 ms. via front-panel adjustments. When used with separate amps, the R-880 also creates ping-pong effects.

CIRCLE 13 ON READER SERVICE CARD

CONTINUING THOUGHTS ON DIGITAL

As we enter a new year, some far-reaching thoughts occur to me regarding the future of sound in terms specifically of digital sound, and especially of a true digital disc (not the analog disc that has been processed from a digital tape), and how it may influence both playback and recording equipment. The digital disc, of course, will be played on a radically new kind of "record player"—one that operates in an utterly different manner than any of today's turntables and pickups.

The digital disc player will incorporate its own digital-to-analog converter which will supply a signal of approximately "line level" value to be fed directly into the "aux" or "high level" input on a standard amplifier—in much the same manner as the signal now available from such sources as a radio tuner, tape playback or preamplified mixer. Without the need for what we now think of as "preamplification/equalization," and with inherent source noise just about eliminated, a large section of what we now regard as the "preamplifier" will become irrelevant to digital sound. The result may well be a new kind of playback amplifier, possibly combined with some options for sound processing such as image or ambient enhancement. I also think there's a good chance that "tone controls" will disappear, and replacing them (as an option) will be the separate equalizer which of course is applicable to both playback and recording.

As for recording—as such—regardless of how the question is resolved over the legality of video-taping copyrighted material, the viability of the VCR format for "live" taping is not at issue. And, as Len Feldman and I have pointed out in past issues of *MR&M*, the VCR is admirably suited for making digital audio tapes, with the addition of course of a suitable converter. If, as is expected, the cost of such a converter comes down in the next few years—or if the converter portion of the digital disc player can somehow be adapted for this use, all those who have bought a VCR may one day discover that they actually own a basic part of the equipment needed for digital audio recording, and this market which manages to cut across consumer and pro users alike could then suddenly become a more important market than what already exists today for analog audio taping.

CIRCLE 14 ON READER SERVICE CARD

MUSICAL NEWSICALS

SYNTHESIZER EQUIPMENT

The MC-4 MicroComposer is an advanced new digital sequencer from Roland Corporation. The MC-4 is actually a powerful computer specially designed for the writing and performance of musical compositions, and is actually a major expansion of Roland's well-known MC-8 MicroComposer despite the fact that it retails for a significantly lower price than the earlier unit. The MC-4 is available with up to 48K bytes of RAM (random access memory) for the musician's use in storing programs, allowing up to 11,500 notes of composition to be available for performance at any given time. The unit also includes 16K bytes of ROM (read-only memory) which contains the unit's internal programs and operating systems which were designed to offer the user almost limitless musical flexibility and functional options. Up to four pairs of voices can be programmed from a synthesizer keyboard and/or the MC-4's own calculator-style keyboard, with the programmed parameters including pitch, rhythm, dynamics and tempo. Editing facilities are provided and the unit may be synced to tape for building multi-track compositions of any complexity. The composition programs may be stored on a conventional tape machine for later retrieval with the MC-4's self-contained tape interface circuitry, or the user may opt for Roland's MTR-100 Digital Data Cassette Recorder which includes automated search for specific data tracks and offers exceptionally fast and reliable storage and retrieval of data.

CIRCLE 18 ON READER SERVICE CARD

New from Oberheim Electronics is the DSX Digital Polyphonic Sequencer. The DSX features a standard 3,000 note memory which may be expanded to 6,000 notes, and has sixteen-voice capability although it is nominally an eight-voice unit with eight independently controlled control voltage outputs and eight gate outputs, each with selectable positive- or negative-going outputs. For flexibility, the unit may be programmed in real time or in a "single step" mode, and sequences may be edited, overdubbed or merged in any order, allowing a degree of compositional control previously unavailable in the price bracket. Other features include the ability to synchronize with a tape for multi-tracking, non-volatile memory with battery backup, cassette storage interface for long-term off-line program storage and a sixteen-character alpha-numeric display.

CIRCLE 19 ON READER SERVICE CARD

Oberheim Electronics recently introduced a new product which is rather a departure from its usual lineup of synthesizers, sequencers and so forth. The new product is designated the Oberheim DMX, and is a fully programmable drum machine using real drum sounds stored digitally in solid-state memory chips. The DMX uses digital control circuitry for complete programmability of song structure, sequence length, time signature and tempo in a similar fashion to a sophisticated digital sequencer (which, in fact, it is). The unit may be programmed in real time or in a "single step" mode, and may be synchronized to a tape machine or to other Oberheim synthesizers or sequencers using its external sync function. The DMX has twenty-four percussion voices which are available separately or as a mixed stereo output; the voices include soft, medium and loud bass drum; soft, medium and loud snare drum; open, closed and accented hi-hat; six tom-toms; three cymbals; handclaps; rim-shot; normal and accented tambourine; and a shaker. Additionally, the DMX has provision for external modulation of pitch and volume of its voices through gate and control voltage inputs. A sixteen-character alpha-numeric display, battery backup for non-volatile memory during power off periods and built-in interface for long term data storage on audio cassettes round out the package's convenience features.

CIRCLE 20 ON READER SERVICE CARD

Sequential Circuits has introduced two accessories to significantly expand the capabilities of its popular Prophet-5 programmable synthesizer. The Polyphone Sequencer for the Prophet-5 is a self-contained unit with its own power supply and digital cassette deck which will expand the Prophet-5 to the same capabilities as the larger Prophet-10's sequencer, namely 2500-note storage in up to six separate sequences, instant transposition, variable playback speed, and flexible record and edit capabilities. Sequences may be recorded in real time or single-stepped, with the single-step mode optionally controlled by a footswitch to leave both hands free for the keyboard. Notes may be overdubbed in the single step mode to allow for corrections, and single notes, chords or entire sequences may be edited or deleted at will. Program changes may be initiated from the sequencer as well as notes and chords and these, too, may be edited or deleted. Sequences and Prophet-5 programs may be stored on digital cassettes for later re-use; when loading from tape, only the sequence data or both the sequence data plus Prophet-5 programs may be loaded depending on the position of the Record Enable switch on the back of the Prophet-5. The Polyphonic Sequencer interfaces directly with all current Prophet-5's of Revision 3 or new (serial number 1301 or higher), while older units will require the installation of an interface connector on the back of the unit by any Sequential Circuits service center. The Remote Prophet is, as the name implies, a remote keyboard for the Prophet-5 which gives the synthesist the same freedom of movement as a guitarist. The unit features a four-octave keyboard plus pitch and modulation wheels on a short "neck" and weighs in at less than ten pounds, not counting the including 20' connection cable. Program selection is accomplished via 5 band select and 8 program select switches located adjacent to the outer edge of the keyboard for easy access. Interfacing of the Remote Prophet is identical to that for the Sequencer.

CIRCLE 21 ON READER SERVICE CARD

GUITARS

The Hamer Prototype guitar is the product of over two years of development which included consultation with over twenty top guitarists including Andy Summers of The Police. The Prototype was designed to have the widest variation of tone available while retaining a simple control layout, and to this end Hamer designed a unique new pickup with both a bar-type magnet and individual polepiece magnets. This new pickup design gives the Hamer Prototype the ability to change from a true single-coil sound to the classic Hamer humbucking sound with a single switch;

additionally, the pickup produces a totally unique thick sound with a special "stringiness" when all three pickup coils are engaged. Only a single pickup is used on the Prototype since a wide range of sounds is available from the pickup and a unique tone-midrange control allows a wide range of modification to the basic sounds of the instrument. On the physical side, the Hamer Prototype is handcrafted from Honduras mahogany in a design which features contouring of both the front and back of the body and exceptionally deep double cutaways which allow totally unrestricted fingering access to all 22 frets. Hardware includes Hamer's half-pound milled brass sustain block bridge and top quality tuning machines. The Prototype is available in natural, sunburst or four solid-color finishes.

CIRCLE 22 ON READER SERVICE CARD

A new six-string electric has been added to Electra's Phoenix model line from St. Louis Music. The Phoenix X-150 has two Magnaflux humbucking pickups which produce a very wide range of sounds thanks to a three-position pickup selector switch, a single-coil splitter switch and a phase reverse switch, and this versatility is further enhanced by an electronics package which includes a built-in preamp/EQ circuit in addition to the usual master volume and tone controls. The preamp/EQ circuit is powered by two 9-volt batteries, and can be used to boost the treble and bass or to produce distortion-inducing gain. The body of the X-150 is made of ash wood, finished with a walnut stain, with a center strip of laminated maple and walnut, while the neck is hard maple with a rosewood fingerboard. Brass bridge, head nut and control knobs, and custom, chrome-plated tuning machines round out the package.

CIRCLE 23 ON READER SERVICE CARD

A new line of acoustic-electric Alvarez guitars was recently announced by St. Louis Music. The five-model Fusion Series from Alvarez are said to feature a well-balanced fusion of true acoustic sound and feedback-free electronics. Fusion guitars are flat-top single-cutaway designs available in either the standard dreadnought body thickness or a special thin-line body for more of an electric guitar feel. For improved projection of their acoustic sound, the Fusion guitars use a special bowed back and internal lacquering. The various models have spruce tops, maple or mahogany sides and two-piece backs as well as rosewood fingerboards and compensated rosewood bridges. On the electronic side, the guitars use an Alvarez balanced phase pickup which produces a natural sound free of feedback and requires no preamps or batteries. A special EQ circuit with two control knobs is located on the upper part of the instrument, and consists of a treble/bass control and a volume/presence control, allowing the musician to blend high and low frequencies for acoustic sound.

CIRCLE 24 ON READER SERVICE CARD

Ambient Sound

By Len Feldman

The Right to Make Tape Recordings

This month I am going to review for readers of *Modern Recording & Music* some events which took place in the Ninth Circuit U.S. Court of Appeals and which concern video tape recording.

More than five years ago—shortly after the Sony Corporation introduced its world-famous Betamax video recording machine to the United States—Walt Disney Studios, in cooperation with Universal Pictures, instituted a lawsuit in U.S. Federal Court against Sony, one of its distributors, its advertising agency and, as a test case, against an actual "civilian" consumer who had been "caught" using a Betamax video recorder to copy a movie (both the video and audio portions) that had been broadcast over a U.S. TV network station. Those who instituted the lawsuit maintained that for anyone to copy a TV program, even in the privacy of one's own home, constituted a violation of the 1976 Copyright Act (a completely re-written version of an earlier Copyright Law).

After much deliberation, the Court ruled in favor of Sony. In other words, they ruled that making a copy of a TV program on a home video recorder was legal so long as the copy was not made to be sold or to be shown where admission fees were charged. "Commercial use or benefit" was the key aspect of this decision.

Sony was, of course, delighted with the decision, even though the Disney people *et al* (as they say in legalese) went ahead with an appeal (common practice). The rest of the industry had forgotten about the entire matter and went right along selling video tape decks. In fact, video recorder sales have shown the greatest increase in sales from year to year of any home electronic product—including audio and hi-fi equipment. In 1978, total video recorder sales to dealers in the U.S. amount to just over 400,000 units. In the first nine months of 1981 alone, sales of video recorders were already nearly 900,000, which, if sales continued for the rest of the year at the same rate, would mean an increase of about 200% from 1978 to 1981.

Then, on October 19th, 1981, much to everyone's surprise and shock, the United States Court of Appeals *reversed* the lower court decision and ruled that home video recording was, in fact, a violation of the Copyright Act of 1976 and was *illegal*. You can just imagine the outcry and indignation that was voiced by the consumer electronic industry members. Almost immediately, several parallel courses of action were initiated. The very next day, a proposal for legislation was introduced into the United States Congress which would specifically exempt the private, non-commercial recording of copyrighted works on video recorders from copyright infringement. If enacted, such a bill would simply specifically exclude home video recording for non-commercial, private use from the copyright laws and that would be the end of the matter. There is, of course, no assurance that this bill will be enacted. Another possible course of action which is being pursued is an official request to the appeals court that it re-hear the case, and, failing that, a request that the entire matter be further appealed to the highest court in the land, the United States Supreme Court. As of this moment, no one knows which of these actions will take place and it is pretty well agreed that the matter is likely to drag on for several years before it is finally resolved.

What About Audio?

All of which brings us to the matter of audio recording from FM and stereo FM programs, or the copying of records onto tape. Both of these practices are, of course, widespread in the United States as well as in other countries of the world. The "victory" of Disney and Universal in the matter of video recording has given encouragement to radio broadcasters and record companies to make another attempt at extracting royalty payments from either consumers or, if that is impractical, from raw audio tape manufacturers to compensate them for the loss of sales which is a direct result of people copying material off the air or from friends' phonograph recordings.

Several journalists, in discussing the taping of audio programs and the question of possible illegalities connected with such activity, have pointed to a congressional act which was passed way back in 1971. Like the bill that is currently being proposed for home video recording, which would exclude it from the general provisions of the Copyright Laws, this bill or law specifically excluded home audio recording from the then-existing copyright laws (back in 1971) so long as the *audio* recording was not used for profit or sold. Many of these journalists have suggested that because of this specific exclusion, passed in 1971, that the home *audio* tape recording enthusiast has nothing to worry about. Unfortunately, that is not quite the case.

The Copyright Laws of the United States were completely re-written in 1976, and the newer copyright law does not include or carry over the specific 1971 congressional act which excluded home recording from records or radio from the act and made it legal. Thus, record manufacturers and radio broadcasters may well feel that if they were to bring a legal action against one or several manufacturers of cassette or open-reel recorders, as well as against distributors and even users of such machines that they would have a good chance of success—in view of the more recent video recording court decision.

What would a negative decision on audio tape recording mean? No one expects that policemen will start knocking on private citizens' doors and carting them off to prisons if they are caught in the act of recording a radio program or copying a record. A more likely result would be that a fixed amount of royalty money would be added to the price of each cassette tape when it is bought as raw, unrecorded tape. The money collected in this way would go into a general fund which would then be distributed in a manner that is as yet undetermined to all those artists, songwriters, members of acting and performing unions, etc., who feel that they are being cheated of the rewards of their artistic efforts. Even the most ardent supporters of such an idea admit that the problem of handling the distribution of monies collected in this manner would be something of a nightmare. It has also been suggested that an extra amount of money should be charged for the purchase of an audio tape recorder in the first place, and that this sum of money should also go into the general royalty fund. Both of these suggestions have been offered as possible solutions in the case of video recording as well, should all the other efforts to overturn the court of appeals decision fail.

Copyright owners (those who stand to benefit in one way or another if this latest ruling by the courts is upheld) were clearly encouraged by the recent ruling. The recording industry association (RIAA), which has long been concerned about the recent decline in record sales, attributes at least part of that decline to the fact that so many millions of people own cassette recorders now, and it takes only one audio enthusiast in a group of friends who own such recorders to supply a single phonograph record that all of them can copy onto tape as they create their own libraries of recorded music. It is rumored that the recording industry is, therefore, considering copyright cases of its own, based strictly upon the premise that it is now illegal even to record *audio* alone. But all copyright owners know that a much larger battle has really just begun, in the midst of a reproduction revolution. Photography, film, videotape, audio tape recorders, photocopiers all could be interpreted as being the means for violating the copyright laws of the United States. The average American has, therefore, been given the ability to "violate" the copyright laws quickly, cheaply and privately. Further compounding the problem are the many new methods of storing and distributing information, such as computer software, microprocessors, cable television, satellite television and audio transmission, microfilm and holography. What, for example, would the courts decide if in the near future we get into home digital recording? A defendant could argue that what he is recording is, in reality, nothing but a vast series of digital pulses ("0's" and "1's") and not really audio or video at all.

"In short, technology in both the audio and video domains is advancing at a more rapid pace than the non-technical lawyers and legislators are able to handle."

In short, technology in both the audio and video domains is advancing at a more rapid pace than the non-technical lawyers and legislators are able to handle. Those on the side of the copyright owners argue that with all these advances in information processing and storage there will, ultimately, be nothing left for anyone to copy from. Nobody will bother to create new music or intellectual and visual works if copyright protections are abandoned. Meanwhile, the electronic industry and those millions of video recorder and audio recorder owners take the position that the new rulings, if upheld, are nothing more than a serious invasion of privacy—in direct contradiction to the freedoms that are supposed to be guaranteed by the U.S. Constitution. As you can see, the matter goes far beyond Disney—or Sony.

Recording Techniques Part 2

by Bruce Bartlett

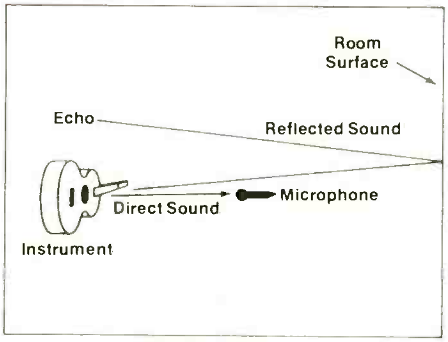
This second article in our series of home recording techniques focuses on the way sound behaves in rooms and on the musical instruments that produce the sound. The acoustic properties of a room greatly influence the sonic character of musical instruments in that room.

Poor recording-room acoustics can muddy your recordings, add noises, and color the tone qualities of instruments. In addition, overly "dead" (muffled) or overly "live" (reverberant) rooms are uncomfortable to play music in. It's important to understand and control room acoustics to provide a good recording environment. Herein we'll divide "room acoustics" into several phenomena: echoes, reverberation, room modes, leakage and noise.

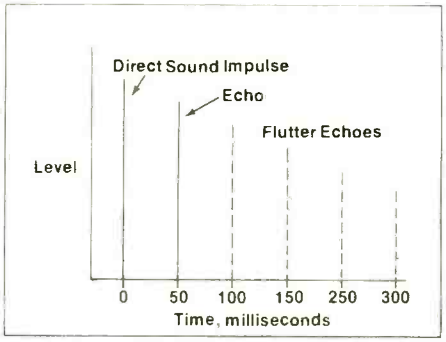
Echoes

To produce sound, most musical instruments vibrate against air molecules, creating sound waves that travel outward in all directions. Some of the sound travels directly to the listener (or to a microphone) and is called *direct sound.* The rest of the sound strikes the walls, ceiling, floor and furnishings of the recording room. At those surfaces, some of the sound energy is absorbed, some is transmitted through the surface and the rest is reflected back into the room.

Since sound waves take time to travel (about 1 msec. per foot), the sound reflections are delayed after the direct sound. The delayed arrival of a reflected sound causes a repetition of the original sound called an *echo* (see *Figure 1*). In some rooms, we sometimes hear discrete single echoes; in other rooms, we often hear a short continuous succession of rapid echoes called *flutter echoes.*



Echo Formation



Intensity vs. Time of Direct Sound and Echoes

Fig. 1. Echo.

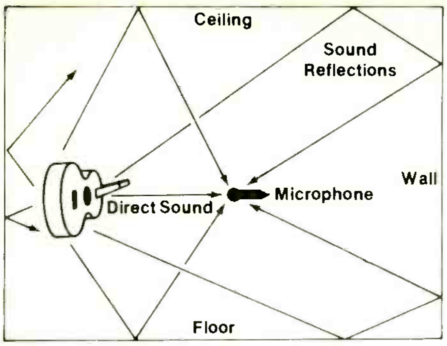
Parallel walls or diagonally opposite corners create flutter echoes by reflecting sound back and forth between each other many times. You can detect flutter echoes by clapping your hands next to one wall and listening for a low "fluttering" sound. Since echoes can reduce the clarity of a recording, they should be eliminated by adding patches of absorbent material (cork, acoustic tile, carpet, fiber glass insulation) to one or both of the offending walls. Putting the material in patches, rather than all together, promotes an even distribution (*diffusion*) of sound in the room.

Use enough acoustic damping to eliminate flutter echoes, but do not deaden the room completely. The room still should have some early reflections (within 25 msec.) so that the room is comfortable to play in. Early reflections also enhance the apparent loudness, apparent transient response and timbre of acoustic instruments.

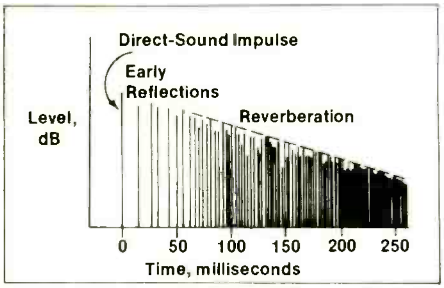
Reverberation

Of course, sound reflects not just once but many times from all the surfaces in the room. These sonic reflections continue the sound of the instrument in the room for a short time even after the instrument has stopped playing. This phenomenon is called *reverberation*—the persistence of sound in a room after the original sound has ceased. For example, reverberation is the sound you hear just after you shout in an empty gymnasium. The sound of your shout persists in the room after you shout and gradually dies away (*decays*). The time it takes for sound at a certain frequency to decay to 60 dB below the original steady-state sound level is called the *reverberation time* (abbreviated T_{60} or RT_{60}).

In physical terms, reverberation is a series of multiple echoes, decreasing in intensity with time, so closely spaced in time as to merge into a single continuous sound, eventually being completely absorbed by the inner surfaces of a room. The timing of the echoes is random, and the echoes increase in number as they decay. *Figure 2* shows



Reverberation Formation



Intensity vs. Time of Direct Sound, Early Reflections, and Reverberation

Fig. 2. Reverberation.

reverberation as a decay-in-time of room reflections.

Note that "reverberation" refers to a *continuous* fade-out of sound, while an "audible echo" is a *discrete* repetition of sound.

Reverberation comes to the listener from every direction since it is a pattern of multiple sound reflections off the walls, ceiling, and floor. Due to our ability to localize the direction of sounds in space, we can distinguish between the direct sound of an instrument coming to us from a specific direction, and the reverberation coming to us from everywhere else. Thus we have the ability to "tune out" the reverberation and concentrate on the sound source. In fact, we are normally not even aware of reverberation.

But a microphone does not have this ability to distinguish between the direct sound and the reverberation. In a playback of a recording, the recorded reverberation is no longer heard from all sides. Instead, it comes from the same point as the original sound (in front of the listener, between the playback speakers). As a result, the reverberation may seem much more noticeable on the tape playback than it was when heard "live."

Too much reverberation in a recording gives a muddy, boomy, distant sound quality, reducing clarity and presence. Consequently, popular-music recording requires a fairly nonreverbant studio, with a reverb time of about 0.4 seconds or less. Classical music, however, should be recorded in "live," reverberation concert halls (RT_{60} about 1 to 3 seconds) because reverberation is a desirable part of the sound in classical music.

Controlling Reverberation

One way to reduce the amount of recorded reverberation is to mic instruments closely. The closer a microphone is to its sound source, the less reverb or ambiance is picked up. Why? As the microphone is placed closer to the sound source, the recorded level of the source gets louder, but the level of the reverberation stays constant. A microphone placed close to a source picks up a much higher ratio of direct sound to reverberation than does a distant microphone.

To further minimize recorded reverb, use microphones with a directional pickup pattern (such as cardioid, supercardioid, or bidirectional). Cardioids, for example, are up to 4.8 dB less sensitive to reverberant sounds than are omnidirectional microphones.

Another effective technique is to record in a room that has little reverberation (an acoustically "dead" room). Since reverberation is caused by sound reflections off room surfaces, any surface that is highly sound-absorbent helps to reduce reverberation. High frequencies are best absorbed by porous, fibrous materials such as fiber glass insulation, acoustic tile, foam plastic, carpeting, and curtains. Spacing these materials several inches from the wall extends their absorption into the mid-bass region. Low-frequency absorbers (called *bass traps*) can be formed of flexible surfaces, such as wood paneling or linoleum mounted over a sealed air space of several inches. Cavities such as closets or air spaces behind couches are also effective sound absorbers.

It's important to have equal sound absorption at all frequencies up to approximately 4000 Hz. For example, if a room is highly absorbent at high frequencies, but not at low frequencies, the highs will be quickly absorbed but the lows will continue bouncing around the room. In other words, the reverb time will be short at high frequencies and long at low frequencies. If you record in a room with that characteristic, both the "live" sound in the room and the recorded sound are likely to be bassy, boomy, and muddy, due to the persistence of low-frequency echoes. Translated into material terms, if your home studio has an abundance of fibrous absorbent materials, but has no bass traps, you can expect dull and muddy sound. Tacking carpet to all the walls is not the way to make a good sounding studio.

We don't want the room to be completely absorbent (dead) because such an environment is stifling—the musician feels he's playing in a vacuum; he gets no reinforcement or enhancement from early reflections. Some reverb and early reflections are beneficial, not only for the musician's comfort, but for the sense of "air" and liveliness they add to the recorded sound.

You can either look around for a good recording room, or you can take an existing room and treat it acoustically. Below is a list of some simple acoustic treatments to reduce reverberation. Add absorption a little at a time until the recorded room acoustics sound good to you.

1. Open closet doors, and space couches and books a few inches from the walls.

2. Carpet the floor.

3. Hang canvas from the ceiling in deep folds.

4. Hang thick curtains or blankets at least 2 feet from the walls, if possible.

5. Attach open-cell acoustic-foam wedges (commercially available from such companies as Alpha Studio, 2049 W. Broad Street, Richmond, Va. 23220, 804-358-3852, "Sonex" brand name; and Esotech, Inc., 7778 Mitchell Road, Minneapolis, Mn. 55344, 612-934-5790, "Wedge" brand name), on or near the walls. The thicker the foam, the better the low-frequency absorption. Four-inch-thick foam on the wall completely absorbs frequencies from about 400 Hz and up.

6. In a basement studio, nail acoustic tile to the ceiling joists, with fiber glass insulation in the air space between tile and ceiling.

7. For bass trapping, nail 3/8" thick plywood panels to 2" x 6" and 2" x 4" studs that are spaced 4 feet apart on the existing wall. Put fiber glass insulation in the air space behind the panel, as shown in *Figure 3.* Or, use linoleum on 2" x 4" and 1" x 1" studs. Cover about half the wall area in this manner.

8. For wide-range absorption, attach 2" or 4"-thick pressed fiber glass board (Owens-Corning Type 703, 3 lb/cu. ft.)

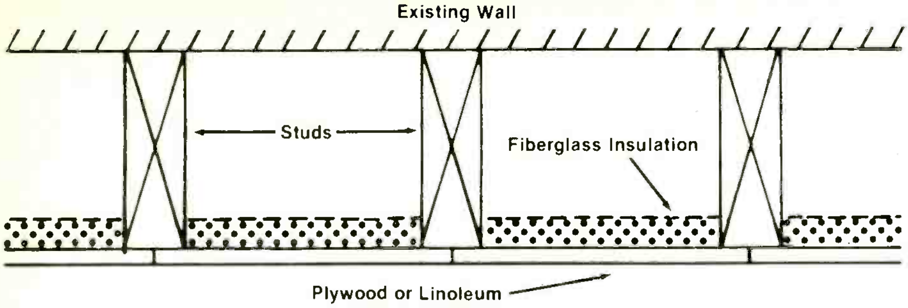


Fig. 3. Example of bass-trap construction.

onto 2" x 6" studs, spaced 4 feet apart on the existing wall, with fiber glass insulation in the air space. Cover with thin, open-mesh fabric. Check your local building codes before you start.

A more thorough discussion of acoustic treatments can be found in the following books: *Building a Recording Studio* by Jeff Cooper; *How to Build a Small Budget Recording Studio from Scratch* by F. Alton Everest, Tab Books, Blue Ridge Summit, Pennsylvania 17214, (1973); *Teac Multi-track Primer* by Dick Rosmini, Teac Corporation of America, 7733 Telegraph Road, Montebello, California 90640; *Home Recording For Musicians* by Craig Anderton, Guitar Player Books, Distributed by Music Sales Corp. 33 W. 60th Street, N.Y.C., N.Y. 10023.

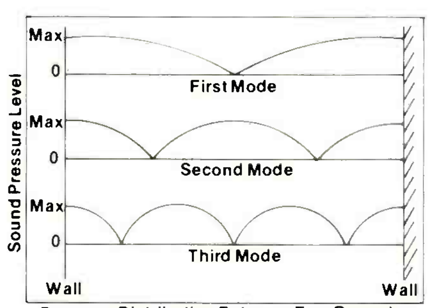
Room Modes

If you play an amplified bass guitar through a speaker in a room, and do a bass run up the scale, you will hear some notes that the room resonates with or reinforces. These resonating frequencies, most noticeable below 300 Hz, are called *room modes* or *normal modes*. Resonance peaks up to 10 or 20 dB can occur. They give a tubby or boomy coloration to musical instruments and should be minimized.

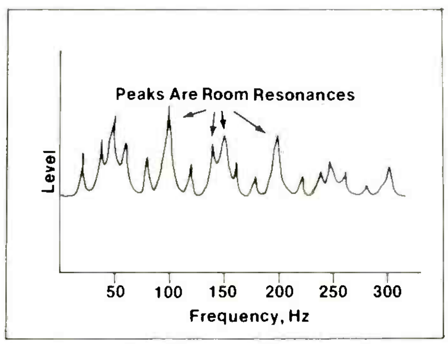
Room modes occur in physical patterns called *standing waves*. Standing waves are uneven sound-level distributions in a room caused by sound waves continuously reinforcing themselves as they reflect between opposing surfaces. Walls opposite each other (or the ceiling and floor) can support standing waves between them (see *Figure 4*). Weaker modes can occur between other surfaces.

The frequencies at which the room resonates depend on the dimensions of the room—its length, width and height. The formula for the most basic room-mode resonance frequencies is f=n 565/d, where "d" is a room dimension in feet, and "n"=1, 2, 3, etc. For example, a room 12 feet long will have room modes at 47 Hz, 94 Hz, 141 Hz and so on. Those frequencies or notes will be overemphasized in the music unless there is a sufficient bass trapping in the room to dissipate them.

Other frequencies will be reinforced by the other room dimensions. If the length, width and height of the room are identical, the same modal frequencies will be reinforced in all three dimensions, creating a terrific buildup of certain low frequencies. On the other hand, if the dimensions are not multiples of each other, the modes will be different for each dimension. Then each room mode will be reinforced in only one dimension and there will be a more even distribution of resonance frequencies.



Pressure Distribution Between Two Opposing Walls, For First Three Room Modes.



Example of Frequency Response of a Room with Standing Waves

Fig. 4. Standing wave phenomena.

There are certain ratios of room dimensions that distribute the modal frequencies in the best way:

1:1.6:2.33
1:1.4:1.39
1:1.28:1.54
1:1.59:1.25
1:1.7:1.47
1:1.45:2.1

Taking the top ratio as an example, if the ceiling height is 10 feet, the room width should be 16 feet and the length should be 23.3 feet for best distribution of modes.

Large rooms are generally preferred for recording over small ones because the room resonance frequencies are lower; hence, more likely to be below the musical range.

A common misconception is that nonparallel walls eliminate standing waves. Actually, low-frequency standing waves are not significantly affected by surface irregularities less than 1/4 wavelength in size. For example, waves of frequencies below 280 Hz do not "see" a skew of 1 foot in a nonparallel wall. A better solution is to use bass traps tuned to the resonance frequencies of the room.

Leakage

Sound from an instrument travels to the nearest microphone, and also "leaks" into the microphones intended to pick up other instruments. For example, a piano microphone may also pick up the drums. The microphone intended for acoustic guitar also may "hear" the electric guitar amp. This overlap of the sound of an instrument into the microphone of another instrument is called *leakage* (or *bleed* or *spill*).

It's very important to minimize leakage—to make each microphone pick up only its intended instrument. Suppose the piano microphone hears a lot of drum leakage (see *Figure 5*). The recorded mix of these instruments will contain not only the clean, tight sounds of the piano and drums picked up by their microphones, but it will also have the delayed, distant, and muddy drum sound that leaked into the piano microphone. The net drum sound will be distant and "trashy," rather than present and clean.

Leakage also lessens your ability to mix the various instruments independently. Increase the level of the piano microphone and the leaked drum sound will increase also. Decrease the level of the drum microphones, and the bad-sounding drum leakage will remain in the mix.

Fortunately, leakage can be minimized in several ways:

Spread out the musicians in the studio—in moderation. The sound of an instrument gets quieter as you move away from it. Specifically, the sound level drops 6 dB per doubling of distance from a "point" sound source in a "dead" room. So, to decrease the loudness of the leakage, put the instruments farther apart.

There are limits to this separation. If the musicians are too far apart, they won't be able to play together in synchronization. Some closeness is needed for ensemble playing. Also, the sound level of the instrument in a room decreases with distance only up to a point, then stays constant due to the room reverberation; so great spacing between instruments may not further improve separation. In addition, if instruments are placed too far apart, the time delays in the leakage are very long, making the leakage sound even more distant and muddy. You sometimes can achieve more *apparent* isolation by placing the instruments closer to each other to shorten the time delays (say, 6 feet to 12 feet apart).

It's instructive to record each instrument one at a time as the ensemble is playing. You can hear how much leakage each microphone is picking up, and you can hear the tone quality and apparent distance of the leakage.

Place microphones very close to instruments. As a microphone is placed closer to a sound source, the recorded level of the source gets louder, but the level of the leakage does not. So, a microphone placed close to an instrument picks up a high ratio of wanted-to-unwanted sound. With close miking, each microphone probably "hears" little except the instrument it's aimed at.

Use directional microphones. A microphone with a directional pickup pattern (such as cardioid, supercardioid, or bidirectional) discriminates against leakage approaching the microphone either from the rear or the sides, or both. For example, a cardioid microphone is most sensitive to sound sources it is aimed at, but rejects sounds coming from the sides by about 6 dB, and rejects sounds from the rear by up to 20 dB. Such microphones reduce leakage pickup as long as the leakage is not coming from in front of the microphone, or is not reflected into the front of the microphone. The bass boost (proximity effect) of closely placed directional microphones can be used to advantage: If you roll off the excess bass at the mixer, you also reduce low-frequency leakage.

Record direct. Amplified instruments such as electric guitars can be recorded *direct* by connecting their output directly to the mixing console through a transformer or a *direct* box. Since the microphone is eliminated, *no* leakage is picked up in the signal of an instrument fed directly to the mixer.

Use acoustic baffles (gobos) between and around instruments. Gobos are portable wall-like structures, usually built of several layers of wood (say, 4 inches thick) and covered with absorbent material on one or both sides. By preventing sound from going through them, they isolate the sound of one instrument from another. Unfortunately, gobos can color the tone quality of direct sound and leakage, and can degrade the performance of directional microphones. Use them only as a last resort.

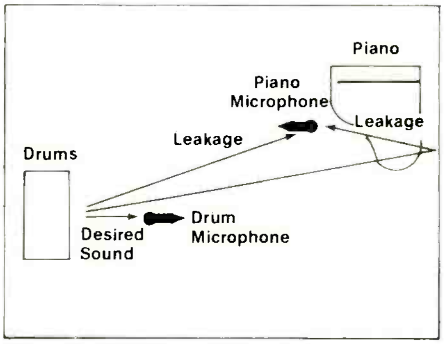


Fig. 5. Example of leakage.

Overdub instruments. Leakage becomes a problem whenever loud instruments and quiet instruments are recorded at the same time. The microphones used on the quiet instruments must be turned up a lot on the console to get sufficient level, and that makes them sensitive to any leakage from loud instruments. It may be best to record all the loud instruments (electric guitars and drums) at one time, then go back and *overdub* (record later) the acoustic guitar, strings, piano, and vocals. Or record all the quiet instruments first, then the loud ones. Overdubbing also lets you mike farther away to pick up a more natural timbre.

Some studios make a practice of overdubbing every instrument for perfect isolation and cleanest sound. That method, however, loses the control interaction between musicians that occurs when they all play together.

Record in a large room. This allows greater physical separation between players, and also weakens leakage reflections due to the longer travel paths to the walls.

Record in a room that is fairly dead acoustically. The absence of reverberation in such a room prevents reflections from various instruments from bouncing into microphones meant for other instruments. The lack of reverberation also reduces the loudness of the leakage.

Note that if you use all these leakage-reducing tricks except for treating the room acosutically, you may be able to make very good recordings without spending any money on room treatment! Unless your recording room is very poor acoustically, you probably can avoid the expense and trouble of building a studio by following the previous suggestions.

A little leakage is not always a bad thing. *Creative leakage* is the use of some controlled leakage to achieve a certain "loose," "live," or "dirty" feel in the recording. The microphones are placed a little farther from the instruments than normal to pick up some leakage.

Noise

Noise is unwanted sound, from such sources as appliances, air conditioning, traffic, airplanes, and noisy neighbors. Here are some suggestions to keep noise out of your home studio:

- Turn off appliances while recording.
- Pause for ambulances and airplanes to pass.
- Close windows; put on storm windows.
- Close doors; weather-strip doors all around—including underneath.
- Replace hollow doors with solid doors.
- Block openings in the room with thick plywood and caulking.
- Remove small objects that can rattle or buzz.
- Put several layers of plywood and carpet on the floor above the studio, and put insulation in the air space between the studio ceiling and the floor above.
- Place microphones close to instruments, and use directional microphones. These methods won't reduce noise in the studio, but they will reduce noise pickup by the microphones.

Choosing a Recording Room

Let's summarize the requirements for a good pop-music studio:

THE PROJECT ONE™
The system that wasn't created equally.

The Project One™ is a fully integrated, perfectly matched, sound reinforcement system capable of performance levels **significantly beyond** any other commercially available system in today's marketplace. This new system represents a new dimension in the art of sound reinforcement and the specs prove it's designed and made by people who know and understand the special problems encountered in high level, concert-type situations.

The FH-2™ low-frequency enclosure is a single enclosure that contains two low-frequency, folded horns oriented in such a way that the combined output exceeds the summed output of two independent horn enclosures. This multiple-driver enclosure provides extremely high power handling (300 watts continuous), wide bandwidth, and very low distortion. Frequency response is 60 Hz to 400 Hz +-3 dB with a 1 watt, 1 meter rating of 110 dB.

The MB-1™ is the heart of the Project One™ system. This specially designed mid-bass horn handles the "problem region" in the frequency spectrum from 150 Hz to 1.2 KHz +-3 dB. Geometry is nominally 60°H by 30°V and continuous power handling is 150 watts RMS. A 12-inch Black Widow® Super Structure™ is the driver for this highly efficient mid-bass horn. One watt, 1 meter rating is 109 dB.

The CH-4C™ high-frequency horn is a constant directivity design that will provide consistently optimum pattern control within its rated frequency range of 800 Hz to 16 KHz. The CH-4C™ is ideally suited for applications where sound must be projected over long distances such as outdoors, large halls, or long rooms. One watt, 1 meter rating is 116 dB.

The high sound pressure levels, smooth frequency response and wide bandwidth operation of the Project One™ will provide unbelievable clarity and projection throughout the sound reinforcement frequency spectrum.

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711 "A" Street / Meridian, MS 39301

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1. *No flutter echoes*—achieved by nonparallel or absorbent walls.

2. *Fairly low reverberation time* (about 0.4 seconds)—achieved by sufficient sound-absorbent surfaces on the walls, ceiling and floor.

3. *Equal reverb time at all frequencies* up to about 4000 Hz—achieved by equal sound absorption at all frequencies. Use flexible panels and cavities, plus fibrous materials, or use fibrous materials spaced from the wall or ceiling.

4. *Some early reflections* (to enhance the sound of acoustic instruments)—achieved by having some hard reflective surfaces in the room. Too many of such surfaces will increase leakage, however.

5. *Minimized, well-distributed room modes*—achieved by large rooms, optimum ratios of room dimensions, and bass traps tuned to room resonance frequencies.

6. *Low leakage*—aided by a large sound-absorbent room.

7. *Good diffusion* (an even distribution of sound)—achieved by nonparallel walls or by installation of sound absorbers in evenly distributed patches, rather than all together.

8. *Low noise*—achieved by the suggestions offered earlier.

"Get a good 'live' sound from the musicians *before* recording them."

So, we have a large, well-sealed room with optimum dimensions. It should have some soft surfaces (carpet, acoustic-tile ceiling, drapes, couches), and some hard vibrating surfaces (wood or gypsumboard walls on studs). Ideally, this room is in a quiet neighborhood.

Because of its size, a club or auditorium where a band plays can be a suitable recording room, although a recording made there is "on-location" rather than "at-home." A large living room opening into other quiet rooms also can make a good studio. The openings act as effective sound absorbers, as do couches, open closets and bookshelves. The walls on studs help to absorb lows and provide early reflections, while the carpeting, stuffed furniture, and drapes help to absorb highs.

Instrument Layout

Assuming you have found or built a good recording room, how should the musical instruments be arranged in the room for a recording session? You can spread them out along the four walls about 3' from the walls to minimize leakage; but too much separation gives diminishing returns, as stated earlier. Try to find a compromise between separation and closeness that the band feels comfortable with.

Arrange the loud instruments together and place the quieter instruments together, farther away from the loud ones. Locating the bass speaker cabinet away from a corner (say, near a long wall) will minimize the buildup of standing waves. The drum set can go in a corner to increase its acoustic bass output, or away from the corner to decrease it. Vocals typically are recorded near the middle of the studio. String sections should be recorded in a hard-surfaced, reverberant room.

Player Talent and Instrument Quality

It's important to get a good "live" sound from the musicians in the studio *before* recording them. To illustrate this, suppose you record a mediocre bar band with a particular recording setup. If you substitute a top-name group of professional players for the bar band, while keeping the same recording setup, chances are your tapes will turn out much better sonically. The pro band musicians have developed many musical attributes that influence the recorded sound quality, such as their playing technique, their overall rhythmic tightness as a group and their arrangements. Also, they probably have better-sounding instruments and they know how to adjust them to obtain tasteful sounds.

Listen to the musicians' "live" sound in the studio and do what you can to improve it. For example, rewrite musical arrangements if necessary to make the overall mix less cluttered. Adjust instrument-amplifier tone controls and special effects. Try different instruments, if possible. Have the musicians modify their playing techniques or instruments for a better sound (as long as that doesn't hamper their ability to play).

We've briefly covered methods for improving the sound of musical instruments and the room they play in. The next article in this series will explore the mysteries of microphones.

GARLAND JEFFREYS

by Jeff Tamarkin

Garland Jeffreys gained his primary musical education on the streets of Brooklyn, where singing groups huddled around street lamps and in subway stations making the harmonies of the 1950s. Jeffreys came up from that humble start through the folk clubs of Greenwich Village, and finally landed his first recording contract in 1973 with Atlantic Records. His debut there didn't do as well as it might have, and Jeffreys went next to A&M, where he recorded the critically acclaimed Ghost Writer *LP, featuring the now-classic "Wild In The Streets" (which had actually appeared on the Atlantic LP first). After two more for A&M—*One-Eyed Jack *and* American Boy & Girl*—Jeffreys recorded* Escape Artist *for Epic. It was possibly his best, and definitely his most noticed record yet.*

Jeffreys took his show on the road following the release of that Epic debut, and recorded the results for his first "live" LP, Rock & Roll Adult. *Following its release last year, Jeffreys met with* Modern Recording & Music *at a Greenwich Village cafe to discuss his career and the following evening's upcoming concert at the Ritz, a New York City concert hall/club. Jeffreys' tour manager and sound engineer, Steve "Siggy" D'Ambrosia explained Jeffreys' stage setup during the soundcheck the next day.*

Modern Recording & Music: In your song "R.O.C.K.," you sing about your involvement in music as being an escape from "a fate worse than death." What do you think your fate would have been had you not entered music, and would it really be something as drastic as that statement in the song?

Garland Jeffreys: Basically, perhaps more so than any other writer, I write about my life and the difficulties I have, and the changes that I've been able to make. I write about the difficulties I experienced in my childhood, and I confront those issues, rise above them and transcend them. But never transcending to the point of looking away from it. That particular line represents a period between '72 and '75 when I did a lot of writing and a lot of soul-searching and a lot of trying to figure out who in fact I am. It was a period when I had done an album for Atlantic (*Garland Jeffreys*), the one with "Wild In The Streets," and nothing happened. Then *Ghost Writer* came along and the title track says, *Then I was so innocent/but now I know the score.* So the songwriting aspect of my career has been a tremendous way of getting through a lot of stuff—making me aware of me. If I hadn't been able to do that I wouldn't have a job and done various things, but I don't think I would've gotten to know myself. To me, that's a fate that's worse than death.

"LIVE!"

Dying is, in a sense, easy. But living in a world where you're not conscious of who you are, you make the same mistakes over again, and you have no way of stepping out. You have no way of growing from the past. Writing and performing on stage has been a tremendous release. You might have heard the song "Cool Down Boy." It's even on my latest album (*Rock & Roll Adult*) in a long "live" version. Well, I don't think I can do that song anymore. I'm finished with it.

MR&M: Do you feel as though you've outgrown the song and the idea of people telling you to cool down?

GJ: Partly that. It's not a question of outgrowing the lyrics, it's one of solving the problem. For me to do it onstage would be very mannered. It wouldn't be spontaneous anymore. I've only recently discovered this. If I do perform the song I'll do a very compact version. It's too personal a song with physical things: of ignoring the child, of there being no communication, no real respect for the child. When I get the time, I'm gonna do what I can to help people out.

MR&M: Under those circumstances, what made you become involved with music in the first place? You were also involved with art. Were those escapes, or releases at first?

GJ: I was already a performer and singer at four or five. I got sidetracked with my folks' definition of what I should be.

MR&M: What did they want you to be?

GJ: They wanted me to become "legitimate," to have something with security: a nuclear engineer or a physicist or something. I had no interest in that, but I kind of went along with it. I made my own decisions and I was always interested in painting and stuff, and always into performing and singing. of music did it to me. I was speaking to this 13-year-old kid on the phone the other day—he's a fan of mine—and he's got a real emotional attachment. It's hard to describe in words; it's just a feeling that is there. There's a feeling you can get when you're eight years old and hearing someone sing "I love you" on the radio that you can't always get at home with your family. It's a very private thing among you and your friends. It was your secret against everybody else's.

MR&M: There's a certain element that was prevalent then in the 50s and 60s that's somewhat missing now, though: the idea of musicians as your peer group. One of the reasons a kid got into Elvis Presley or the Beatles was because his or her parents weren't into them; it was yours and they couldn't have it. Now you have teenagers who listen to the Doors and they have parents who listened to the Doors. It's

to pull out and do onstage every night. If I do it, I'll pull it out on special occasions. It's interesting to me that I've come to that conclusion because I never thought I would. But then I might feel differently some day and do it again; it's hard to say what will happen.

I feel I've solved the problem [discussed in the song] of my father and mother. I'm more at peace with them now.

MR&M: The subject of childhood and growing up is mentioned in a number of your songs. How much of that commentary is based on personal experience?

GJ: I had a tough life growing up. There are kids that have had it a lot rougher than I have, but I got knocked around and hurt. Then there's the question of child abuse that has nothing to do

MR&M: You were first involved with group singing, or "doo-wop," right?

GJ: Yeah, but, by the way, I hate that term "doo-wop." It's minimizing what the music is: it was a great cultural music that was a big salvation for a lot of people, me included.

MR&M: What do you recall from growing up in Brooklyn? The New York area has a very rich rock and roll history...

GJ: Yes, and I recall listening to Alan Freed on the radio and Jocko's "Rocket Ship" [a radio program]. And as everybody now knows about me, I was very much into Frankie Lymon and the Teenagers. I liked groups such as the Harptones, the Cleftones, the Drifters, Louis Lymon, Frankie Lymon; that kind

all changed now.

GJ: Absolutely. All the adults hated Elvis Presley, so the kids could love it. If you hate it, then I love it. In Brooklyn where I grew up, it was so beautiful to go down to the beach and hear a five-piece group singing by the boardwalk at the beach. It was just like you heard on your records: they could do it and you could do it.

MR&M: That's another thing: now there's a gap between musicians and fans. A kid can't possibly imitate what he hears on the radio because you have to be a master musician and record in 48 tracks to make the music heard on the radio.

GJ: That's a far cry from going down to the subway station at Sheepshead Bay in Brooklyn and trying to get the right echo.

Performer Series

music & electronics . . . naturally!

Effect Your Imagination!

DL-2A, ACOUSTICOMPUTER®

DL-4, TIME LINE®

DL-5, HARMONICOMPUTER™

The *Performer Series* — designed for you, the *creative* musician.

FEATURES

- 90 db Dynamic Range
- 15kHz Bandwidth at all Delay Settings
- Advanced State-of-the-Art Digital Encoding
- Each unit occupies only one rack space
- Flanging
- Doubling/Chorusing
- Reverberation/Echo
- Musical Harmonization
- VCO & Feedback Controls
- Easy-to-use Front Panel Operation
- External Control Inputs for Remote Operation
- System Compatibility

ON STAGE

The Performer Series has been specifically conceived to meet the demanding needs of the creative musician. Each of these units has been tailored to provide optimum performance and reliability with a total ease of operation not generally found in competitive units. No longer is the musician compelled to use bulky studio equipment that often requires a "graduate engineer" to understand the controls and theory of operation before any useful musical benefit can be derived . . . not to mention the expense of purchasing such devices. Nor, is the musician forced to use less-than-ideal single function boxes; which, when connected together are usually very noisy and limiting.

Now, the truly creative musician has a clear choice. The Performer Series offers the musician studio quality with on-stage ease of operation to enhance his performance.

IN THE STUDIO

Although the Performer Series has been tailored for on-stage use, no sonic compromises have been made. As such, these units are as much at-home in the professional recording studio as they are on-the-road. The professional recording engineer will find them to be the most creative signal processors in his effects rack. Their ease of operation will provide a flexibility which instills confidence and encourages experimentation.

Certainly then, the musician's home studio is also a perfect home for the Performer Series; demo tapes will now take on a new sonic professionalism. Because the units are not bulky, they can easily serve the dual role of providing special effects for both the on-stage performance and the at-home recording.

A WORD ABOUT AUDIO-TO-DIGITAL CONVERTERS

Aliasing is a particularly annoying form of distortion caused by a sampling rate which is too low for accurately reproducing high frequencies. Although frequency related, aliasing is a foldover (non-harmonic) distortion and is noticeable . . . even at modest levels. The audible effect is generally a disconcerting "wrongness" or "sounds digital" quality. Steep anti-alias filtering is employed on properly constructed PCM units, which tends to correct the aliasing while introducing its own set of ills (most noticeably, phase shift at high frequencies). In some low cost PCM units these filters are not included, rendering the unit useful only at short delay times or non-critical applications. DeltaLab's patented ADM (Adaptive Delta Modulation) encoding technique is unique in the world of digital audio. Our minimum phase-shift circuitry and high sampling rate (up to 10 times faster than PCM) prevents aliasing and ensures natural sound quality totally eliminating the "sounds digital" reproduction often heard in PCM systems.

WHO IS DELTALAB?

In digital audio, experience counts. DeltaLab is a consortium of musicians, engineers, scientists and consultants. Our equipment is found in the best "world-class" studios, as well as on stage with the world's finest performers. We maintain an active R&D program and are constantly advancing the state-of-the-art in digital audio. The Performer Series has been designed to provide the creative musician with the benefits of this experience.

DL-2A, ACOUSTICOMPUTER®

The DL-2A is an updated version of our highly successful DL-2. It is a true stereo (2-channel), studio quality, digital delay and special effects processor.

DL-6, HARMONICONTROLLER™

MEMORY MODULE

DLB-1 FOOT PEDAL*

The DL-2A is also a flexible, rugged, all-electronic "space machine" with impressive performance and proven roadworthiness. Originally designed to be at-home in the studio, it is equally at-home on stage and on the road. The DL-2A provides well known functions such as flanging, double and triple tracking, chorusing, vibrato, reverb, echo and resonant effects plus new effects unique to the ACOUSTICOMPUTER.

DL-4, TIME LINE®

The DL-4 is the most popular single channel digital delay and special effects device designed for the creative performing musician. Over ½ second (512 ms) of full bandwidth (20 to 15K Hz) delay! Compare other "musician oriented" effects devices with the TIME LINE and you'll see why major artists throughout the world consistently choose the DL-4.

DL-5, HARMONICOMPUTER™

The HARMONICOMPUTER is a high quality pitch-shifting device featuring keyboard type controls for precise musical intervals. The DL-5's ease of operation instills instant confidence and encourages impromptu experimentation. Now the performing musician can achieve the kind of performance previously expected from only the costliest of studio quality pitch-shifting devices. The DL-5 maintains the same full bandwidth, low distortion, dynamic range and construction techniques that you have come to expect from DeltaLab.

DL-6, HARMONICONTROLLER™

The HARMONICONTROLLER is an electronic musician. This device has been programmed with basic music theory to provide control signals for up to (3) three DL-5, HARMONICOMPUTERS. The DL-6 listens to the musical note being performed and compares it to the musical key signature addressed in the stored program. By this process, it creates a control signal to externally direct the attached HARMONICOMPUTER(s) to the proper pitch-shift to, in turn, create "real" musical harmony. *No other device like it exists* — it is indeed a unique concept which removes the burden of "tap-dancing" (recommended by some competitors) on foot switches to achieve real harmony.

Only DeltaLab has it. Used to its maximum capacity, a single voice can sound like a duet, trio, or quartet — musically!

MEMORY MODULE

The MEMORY MODULE allows up to two seconds of full bandwidth delay to be externally added to DeltaLab units. All performance specifications are maintained with absolutely no change in performance other than longer delay times. Foot switch bypass and internal front panel bypass allows the memory module to be effectively used by the musician. Plug-in capability is standard for the DL-2A ACOUSTICOMPUTER and DL-4 TIME LINE.

DLB-1 FOOT PEDAL*

Designed to be an integral part of the DL-4 TIME LINE, the DLB-1 allows remote, hands-off, control of the following functions: up-delay count, down-delay count, VCO sweep, infinite repeat and system bypass and activate. Built to our specifications by MORLEY, the DLB-1 is constructed of heavy gauge steel with optical electronics to ensure reliability and long life. The DLB-1 can also be effectively used with the DL-2A ACOUSTICOMPUTER.

SYSTEMS CONCEPT

Unlike most competitive units, the Performer Series has been designed for system operation. Each device can be interfaced to additional devices externally with no degradation in performance. For example; two DL-4's can be cross-coupled externally to form a stereo pair that is both independent and interdependent. Signals processed by DL-5's can be further processed by DL-4's or DL-2A's to create spatial effects.

All of the units can be controlled by foot switches for such functions as bypass and infinite repeat. In addition, the internal VCO's can be modulated with an external voltage source from a synthesizer or foot pedal (DLB-1). MEMORY MODULES can be added in tandem to both the DL-2A and the DL-4 to provide extra-long delay times (2-seconds per MEMORY MODULE) without degrading sonic quality.

The DeltaLab Performer Series systems concept allows the musician to confidently invest in quality products without fear of obsolesence. As new products are developed by DeltaLab they are designed to fit into this system concept.

*Designed for DELTALAB RESEARCH by MORLEY a division of Tel Ray Electronics Mfg. Co., Inc.

A Versatile Special Effects Processor Featuring *True Stereo* (2-Channel) Operation

FEATURES

- 16 Reverb Programs
- Two Independent Pre-Reverb Delays
- Sinusoidal VCO
- Infinite (non-deteriorating) Repeat
- Flanging/Doubling/Chorusing/Reverberation
- Stereo Imaging on input to extract ambience
- Stereo Imaging on output for Haas-effect panning
- Foot Switch Bypass and Repeat Controls
- 15KHz Bandwidth at all delay settings
- 90 db minimum, 95 db typical Dynamic Range
- External Memory expansion capability
- Single rack space (1¾ x 19 x 10 in.)

DESCRIPTION

The DL-2A, ACOUSTICOMPUTER® is simply the cleanest sounding, most flexible, widest-range, digital delay and special effects processor. It is actually two versatile, high performance products in one.

The DL-2A is a stereo digital delay line (DDL) providing two independent channels of time delay. Delay times are selectable in precise discrete steps and in a continuously tunable sweep between adjacent steps. Any delay from 0.25 millisecond to 160 milliseconds may be selected in the stereo mode; by selecting the serial mode, delays up to 252 milliseconds are available. Full audio bandwidth (15 kHz) and full dynamic range (over 90 dB) are maintained in the DL-2A at all delay settings. If still longer delays are desired, they can be obtained by connecting the optional MEMORY MODULE to the DL-2A providing up to two seconds (1-second per channel) of delay per module.

The DL-2A is also a versatile, special-effects processor providing well-known and widely-used effects such as reverberation, comb filtering, flanging, vibrato, vocal doubling and enchancement; in addition, a number of effects are provided which are unique to the DL-2A. Unlike many special-effects processors, no compromises in bandwidth or dynamic range are made in order to achieve the widest array of effects. Its flexibility, sonic performance, compactness, ruggedness and reliability make the DL-2A ideal for the performing musician.

APPLICATIONS

The DL-2A is capable of providing a very large array of effects from straight Delay to multiple-feedback reverberations.

The DL-2A can be used as a straight two-channel digital delay to provide pre-reverb delays, discrete echoes (also known as slap-back), doubling and tripling used to broaden and thicken vocals, and Haas effect imaging to pan signals left-right, etc., without lowering amplitude in either channel.

Sixteen reverb programs are stored in the DL-2A. These programs consist of four separate delay taps, two per channel, (independent of the pre-reverb delays). The four reverb delay taps can be recirculated via the feedback controls to create room-like reverberation. The programs have been designed for both realistic as well as far-out spatial reverb. In addition, the DL-2A offers both serial and parallel combinations. Thus, the DL-2A actually has 32 useable reverb programs.

Various flanging effects can be created by the internal VCO, from a straight flange to the UFO's-have-landed flange. These include both positive (in-phase) flanging to augment low frequencies and negative (out-of-phase) flanging to diminish low frequencies. By adding feedback to the flange, deep notches are created and the resulting flange effect is astounding. Using the "sample mode" creates flanging "space" effects that are unique to the DeltaLab DL-2A, ACOUSTICOMPUTER.

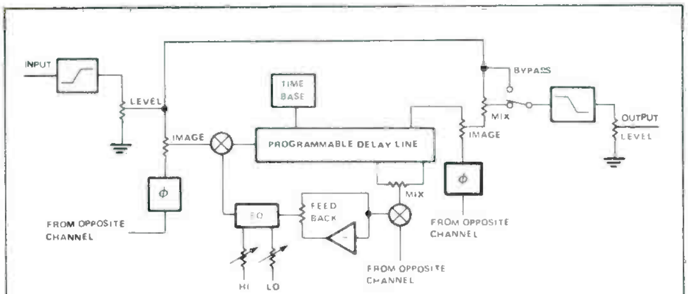
DL-2A, ACOUSTICOMPUTER

The ACOUSTICOMPUTER can be used during live performances to both open up a dead room and to enhance the performance with special effects. Because the DL-2A is an updated version of DeltaLab's highly successful DL-2, professional module, the musician can feel confident that this device is most suitable for use as an all encompassing special effects processor for his home studio. Coupled with a good tape recorder and mixer the DL-2A provides all of the necessary signal processing, including reverberation, required to make professional demo tapes. In this application no other device can provide as much flexibility.

The DL-2A can be interfaced with the MEMORY MODULE to extend its delay capability. Up to one second per channel is available in each MEMORY MOUDLE. This combination provides some off-the-wall effects, again, not found in other units.

When you consider its road-worthy construction, highly-musical sound quality, tremendous flexibility . . . and you hear for yourself what it can do in the creative process, you will discover that the ACOUSTICOMPUTER is indispensable.

DeltaLab ACOUSTICOMPUTER DL-2A



Block Diagram - ACOUSTICOMPUTER (One Channel Shown)

SPECIFICATIONS DL-2A, ACOUSTICOMPUTER®

Delay Range	0.5 to 160 ms (chA), 0.25 to 92ms (chn B) 0.75 to 252 ms (ch A&B, serial mode)
Frequency Response[1] **(@-14db) +1, -3dB**	20 to 15kHz @ all delay settings
Dynamic Range	
A-weighted	90dB min, 95db typ
C-weighted (Broad band)	85 db min
Headroom above OdB	6dB
THD (Distortion plus Noise)[2] **ref 1 KHz OdB**	0.2% max @ x1 0.4% max @ x4
Inputs	
Level	0 to 18 dBm
Impedance	47K ohm
Outputs	
Level	Up to 24 dbm (balanced) Up to 18 dbm (unbalanced)
Impedance	600 ohms
Delay Factor	Continuous from X1 to X4
VCO Depth	0 to 100% of Delay Factor (4:1)
VCO Rate	Sinusoidal from 0 to 10Hz
Sustain	Repeats signal in memory indefinitely without degradation. Sample mode is used to update with new information at rate determined by the VCO
Reverberation	16 Programs displayed via binary weighted LED's.
Size	1¾ x 19 x 10 in. (4.45 x 48.3 x 25.4 cm)
Shipping Weight	12 lbs (5.5 kg)

[1]Straight Delay Mode
[2]With C-Weighted Broadband Filter
Manufacturer reserves the right to make improvements without notice or obligation; therefore, all specifications are subject to change.

DL-4, TIME LINE

A Special Effects Digital Delay Line Featuring the *Longest* Full Bandwidth Delay Times Available

FEATURES

- Unique VCO with infinitely variable waveshape control
- 512 milliseconds of full bandwidth delay
- 2.5 seconds of full bandwidth delay with external memory module
- Flanging/Doubling/Chorusing/Echo
- Digital Octave Flanging (a new effect)
- Infinite (non-deteriorating) Repeat
- 15k Hz Bandwidth at all delay settings
- Resonant and doppler effects
- Footswitch control of effects
- 90 dB minmum; 95 dB typical Dynamic Range

DESCRIPTION

The DL-4 is the most popular single channel special effects processor. It has been accepted and used by some of the best performers in the world. In fact, its sonic quality is so superior that it has found its way into many world-class recording studios.

The DL-4 can be used as a straight delay line providing over ½ second of delay. Delay times are selectable from 0.25ms to 512ms with no signal degradation. Delay times are selectable in precise discrete steps in three ranges and in a continuous tunable sweep between adjacent steps. Full audio bandwidth (15KHz) and full dynamic range (over 90dB) are maintained in the DL-4 at all delay settings. If longer delays are desired, they can be obtained by connecting the optional memory module to the DL-4 providing up to two additional seconds of delay per module.

The DL-4 is also a single channel special effects processor providing well known and widely used effects such as flanging, doubling/chorusing and echo. By adding feedback and a VCO, vibrato, tremolo and chorusing effects are possible. The DL-4 also provides unique effects which are not even creatable in our own DL-2A. At no time is there a compromise in sonic quality. Both bandwidth and dynamic range are preserved at all delay settings. As with all DeltaLab products, its rugged construction and reliability make the DL-4 the popular choice for the performing musician.

APPLICATIONS

Like the DL-2A, the DL-4 is capable of providing a very large array of effects from straight delay to multiple feedback echoes. Such effects are generally well known to the creative musician.

The DL-4 can be used as a single channel pre-reverb digital delay line providing very long delay times which can also be used for discrete echoes. The DL-4 also features high quality doubling and chorusing to broaden and thicken vocals.

The DL-4, via its feedback and VCO controls, provides the capability to create various flange effects with short delay times. These include both positive (in-phase) flanging to augment low frequencies and negative (out-of-phase) flanging to diminish low frequencies. By adding feedback, some very deep notches are created providing very distinctive characteristics to the DL-4. Both the positive and negative flange can be mixed either in or out of phase with the source. In addition, the VCO that controls the flange, can be altered from a triangular to a sinusoidal to a square wave function which, in itself, provides great flexibility. Also the DLB-1 foot pedal can be used to control the VCO, giving the musician total control of the generated flange effect.

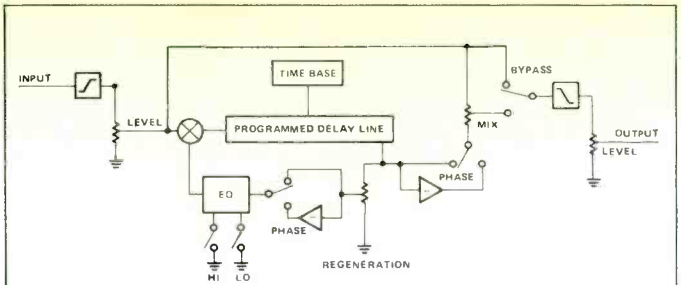
By using delay times in the vicinity of 16 thru 64ms the DL-4 provides a high quality doubling. Adding a little feedback yields a chorus effect. With the use of the VCO controls, a vibrato/tremolo can be used to add realism to the doubling and chorusing.

Extending the delay times to over 64ms will create noticeable echoes. These can be used in conjunction with the feedback and infinite repeat controls to create space-like effects.

The DL-4 by itself is an amazing tool for generating effects. But when two DL-4's are cross-coupled, the result is mind boggling. The individual effects are expandable to such a degree that even our own DL-2A cannot match or recreate them. The pair of DL-4's can also be used to create stereo effects.

The DL-4 delay times are expandable via a MEMORY MODULE. Up to two seconds of additional delay can be added per MEMORY MODULE. Again, this combination provides some awesome special echo effects when used with the feedback or infinite repeat controls.

The DLB-1 foot pedal, designed for DeltaLab by Morley, allows the musician to remotely control the DL-4. The DLB-1 switches provide the means to alter delay times, bypass the DL-4, and put the DL-4 into infinite repeat while the foot-pedal externally controls the VCO. This is a powerful combination for on-stage use.



Block Diagram - TIME LINE

SPECIFICATIONS DL-4, TIME LINE®

Delay Range	1 to 512 ms
Delay Factor	.25 to 1.0 of setting
Frequency Response[1] (@-14dB) +1, -3 dB	20 to 15k Hz @ all delay settings
Dynamic Range	
A-Weighted	90 dB min, 95 dB typ
C-Weighted (broadband)	85 dB min
Headroom above 0 dB	6 dB
THD (Distortion Plus Noise)[2] Ref 1kHz, 0db	0.2% max @X.25 0.4% max @x1
Inputs	
Unbalanced (Phone)	0 to 18 dBm Hi Level -20 to 0 dBm Low Level
Impedance	47K ohm
Outputs	
Unbalanced (Phone)	Up to 18 dBm
Impedance	50 ohms
Time Base Modulator	
Delay Factor	Continuous from X.25 to X1
VCO Depth	0 to 100% of Delay Adjustment Range (4:1)
VCO Rate	From 0 to greater than 10 Hz
VCO Shape	Triangular to sine to square wave-continuously variable
Repeat	Repeats signal in memory indefinitely without degradation
Regeneration	Recirculates delay setting to create multiple echo repeats
Size	1¾ x 19 x 10 in. (4.45 x 48.3 x 25..4 cm)
Shipping Weight	12 lbs. (5.5 kg)

[1]Straight Delay Mode
[2]With C-Weighted Filter
Manufacturer reserves the right to make improvements without notice or obligation. Therefore, all specifications are subject to change.

A Special Effects Pitch-Shifting Device Featuring Keyboard Type Controls For *Precise* Musical Intervals

FEATURES

- Full two octave operation — one octave higher and one octave lower.
- Simplified Keyboard Style operation.
- Glitch free operation — advanced digital technique eliminates splice-glitches.
- State-of-the-art digital encoding technique maintains maximum bandwidth.
- Selectable pitch shift control allows full range (two-octave) sweep or fine tune (with keyboard).
- VCO provides vibrato effects.
- Feedback circuit creates recirculating harmony and arpeggios.
- External control inputs for pitch shift, bypass and additional delay.
- 90 dB minimum; 95dB typical Dynamic Range.

DESCRIPTION

The HARMONICOMPUTER™ is a high quality pitch-shifting processor designed for the creative professional musician in live performances as well as the recording studio. By using DeltaLab's most advanced encoding system coupled with special multiplying and multiplexing techniques, the DL-5 totally eliminates the typical splice-glitch. The result is a clean sounding harmonizing unit that is *not* plagued with unnecessary side effects.

The DL-5 features a unique keyboard control for precision and repeatability. The front panel consists of an arrangement of rocker switches set up to simulate a keyboard. This configuration allows the user to control the desired harmony in common musical intervals. The recording engineer has the option of disabling the keyboard control and using the fine tune control as a full, two-octave, continuously variable sweep pitch control.

A feedback control, which recirculates the harmony, is used to create electronic chords and/or an arpeggio effect. This becomes even more impressive when used with an external delay such as the DL-4, TIME LINE®. A full vibrato control is provided to give the musician or recording engineer the flexibility to create new dimensions in natural sounding harmony. The HARMONICOMPUTER is also an effective tool for the studio boasting a virtually distortion free 20-15kHz bandwidth and 90 dB dynamic range — in a compact package with the same rugged construction common to all DeltaLab products.

APPLICATIONS

By itself the DL-5 can be used on stage or in the studio to pitch-shift a music or voice signal within a two octave (± 1 octave) range with the front panel keyboard making it easy to select a musical interval. Also, the null control allows the musician to fine tune the interval if required. By a simple flick-of-the-switch the fine-tune control becomes a full-range control so that any pitch shifting interval, musical or otherwise, can be selected.

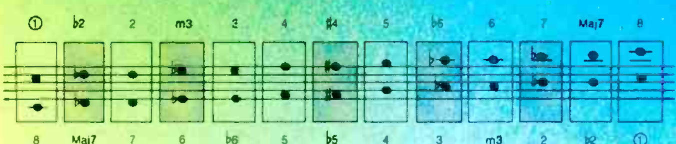
By selecting certain intervals, a "fixed harmony" results. Although this "fixed harmony" is not always musically correct for every note played into the DL-5, it is still a result of an accurate pitch shift. Since music theory dictates the need for the changing intervals to harmonize to each note in the scale, a ±½ tone control switch input is externally available. This input will vary the selected interval ±½ tone from its setting. For exam ple: A third can be lowered to a minor third or raised to a fourth via this input with a foot switch. This amounts to tap-dancing while performing and is a common solution recommended by competitors with their pitch shifting devices. If you like to tap-dance the solution is viable.*

The DL-5 can be interfaced to an external music synthesizer via its external full-range control input. This allows the musician to select the musical interval on the synthesizer keyboard. As such, the external input overrides the front panel controls. The external control input can also be used with the DL-6, HARMONICONTROLLER™ (described in the following pages,) to create natural musical harmony, automatically.

*DeltaLab has developed an automatic controller, the DL-6,HARIMONICONTROLLER™, which eliminates the need to tap-dance. The DL-5 accepts a control signal from the DL-6 and accurately shifts the pitch as the musical notes change. The result is amazing and almost foolproof. Up to three DL-5's can be connected to a single DL-6 to provide a very convincing quartet from a single musical voice. Check it out . . . music will never be the same.

DL-5, HARMONICOMPUTER

"GETTING AROUND THE KEYBOARD"



The keyboard consists of 13 rocker switches (8 white and 5 black) to represent a full octave. For descriptive purposes each white switch is represented by a note from the C-major scale. The black keys represent the ½ tones between the notes of the scale. Each of the rocker switches can toggle either up or down. By depressing each white switch in the up direction, the harmony produced by the DL-5 will *increase* in a major scale-like fashion. Similarly, by depressing each white switch in the down direction, the harmony will *decrease* in major scale-like fashion.

Depressing either the left most switch in the up position, or the right most switch in the down position, results in no pitch-shift. These positions are referred to as the root. Depressing each switch in succession results in harmony which increases or decreases in a chromatic (½ tone) scale-like fashion.

It should be noted that "ups" and "downs" of each switch are identified by the most commonly used musical intervals. These intervals are referenced to the root. In each case, depressing a switch from up to down and vice-versa results in an octave difference. For example; up a minor third (m3) is exactly one octave higher than down a sixth (6).

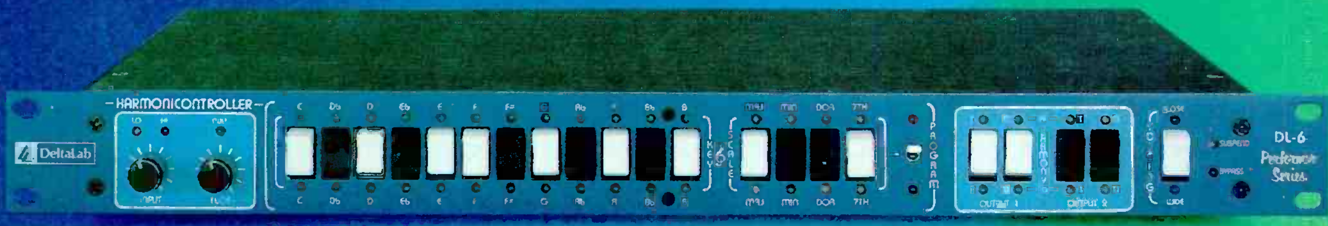
When using the feedback control, depressing certain switches will yield musical chords or effects. Recirculating a minor third (m3) will yield a diminished chord; a third (3) will yeild an augumented chord. Other combinations will give interesting results although not related to simple harmony. The fine tune control can be used to precisely tune or tailor a particular setting or to allow harmonies that are in the "cracks" (between notes).

SPECIFICATIONS* DL-5, HARMONICOMPUTER™

Frequency Response[1]	
+1, −3 dB	20 to 15kHz @ no pitch-shift
Dynamic Range	
A-Weighted	90 dB min
C-Weighted (broadband)	85 dB min
Headroom above 0 dB	6 dB
Inputs	
Unbalanced (Phone)	0 to 18 dBm (Hi Level) −20 to 0 dBm (Low Level)
Impedance	47K ohm
Outputs	
Unbalanced (Phone)	Up to 18 dBm
Impedance	50 ohms
Pitch-Shift	
Full Range	±1 octave min
Fine Tune	±½ tone min
External Pitch Control[2]	Each ½ tone increases/decreases by 0.083 volts
Octave Lower	+1 Volt
Null (No Pitch-Shift)	+2 Volts
Octave Higher	+3 Volts
Time Base Processor	
Vibrato Width	0 to maximum depth
Vibrato Speed	0.1 Hz to 10 Hz
Feedback	Recirculates pitch-shifted signal 0 to almost unity
Size	1¾ x 19 x 10 in (4.45 x 48.3 x 25.4 cm)
Shipping Weight	12 lbs (5.5 kg)

[1]Frequency Response varies accordingly with % of pitch-shift within 20 to 15kHZ.
[2]External control jack is located on rear panel. When used, all pitch-shift controls on the front panel, including vibrato, are automatically disabled.
*Manufacturer reserves the right to make improvements without notice or obligation; therefore, all specifications are subject to change without notice.

DL-6, HARMONICONTROLLER



A Pre-programmed Music Computer to *Personalize* Your Music.

FEATURES

- Preprogrammed Music Theory
- Three output control signals for four part harmony
- Performs in all musical keys
- Four musical scales
- Two distinct harmonic voicings
- Instant operation — no noticeable delay
- External Bypass
- Single rack space (1¾ x 19 x 10 in)
- Controls one, two or three DL-5's

DESCRIPTION

AT LAST! . . . Real harmony from an electronic device. The DL-6 monitors the musical signal, detects the actual note being performed, and in turn, creates a voltage control signal for each of up to three DL-5's. The DL-6 does not alter the input audio nor does it create its own audio; it is transparent to the audio signal path. It merely tells the DL-5 what to do; i.e., what amount of pitch to shift its input signal to. The DL-5's do all of the audio processing. The DL-6 is, in fact, a music computer that is pre-programmed with basic music theory. As such, it eliminates the need to tap-dance on foot switches to produce real musical harmony. All that is necessary is to select the musical key signature and scale desired and the DL-6 does the rest.

The front panel is set up in keyboard fashion much like the DL-5. Two separate musical key signatures and scale settings are selectable. This can be done either by the front panel program control switch or by an external input for remote control. These features give the DL-6 the flexibility to instantly change key in the middle of a performance.

The DL-6 also offers an external "panic button", a foot switch control input that can be used in the rare instance when the musical harmony generated is not quite desirable. By switching this "suspend" control externally, the harmony will adjust automatically.

APPLICATIONS

The DL-6 is programmed to handle all types of music. Four scales have been programmed and are addressable at the front panel: Major, minor, seventh and Dorian modes are selectable by a flick of the switch. In addition, two separate voicings, close or wide harmony, can be addressed. The difference in the voicings are most noticeable when the DL-6 is controlling more than one DL-5, HARMONICOMPUTER. The close voicing mode is very jazz-like in nature, while the wide voicing is intended for Country Western and Rock music. Each of the two DL-5's can be controlled by selecting the harmony for each output. Two harmonic inversions (notes in the chord) are selectable, either above or below the melody indicated by HI or LO. The third DL-5, when used, is automatically controlled depending on the setting of the voicing control at the DL-6, HARMONICONTROLLER front panel.

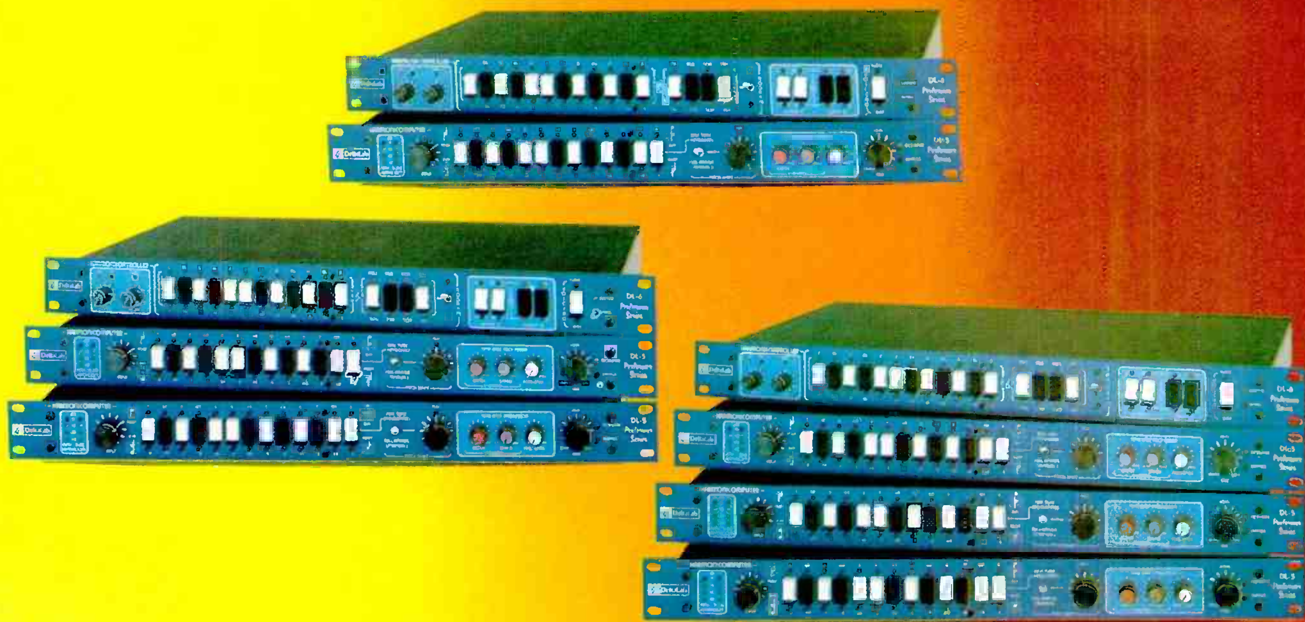
The total system can be bypassed with the external bypass control input on the DL-6. Furthermore, each DL-5 can be individually bypassed with its own respective bypass control input. The DL-5, HARMONICOMPUTER allows a source/harmony mix; by this means, the individual loudness of each harmony part can be adjusted. Of course, with an external mixer, this becomes unneccessary.

DUETS (with one DL-5)

When the DL-6 is used to control one DL-5, the harmony is amazingly accurate. The resulting two-part harmony, which can be positioned above or below the melody, is very effective with both musical instruments and the human voice. Simply select the musical key signature and scale of the music and address the desired voicing and output. This combination can be very tastefully used in all Rock and Country Western music. The effect is both amazing and natural.

TRIO (with two DL-5's)

Using the DL-6 to control two DL-5's simultaneously results in a trio. Two distinct harmonic parts are created and added to the melody. Since all music harmony is subjective by nature, the resulting harmony of this "trio" combination can be "musically correct" yet



not fully satisfying for all music. Most often, all that will be required to produce very satisfactory harmony is the proper selection of voicing, (either close or wide,) and positioning, (either above or below the melody).

Again, this combination is effective for Rock and Country Western music, especially with "wide voicing" and is quite acceptable for jazz performances with "close" voicing. The creative musician need only to experiment with this set up to derive amazingly satisfactory results.

QUARTET (with three DL-5's)

By adding a third DL-5 to the trio combination the result is quartet harmony. Three separate harmony parts are created. The third DL-5 is automatically controlled by the setting of the VOICING control. In the "wide" position the third DL-5 is committed to an octave lower which results in a trio with a "double lead". When the VOICING control is set to the "close" position, a true fourth part is created. The third DL-5 will then be pitch shifted by a fifth or flatted fifth depending on the input note.

As with the DUET and TRIO combination the "wide" voicing is excellent for Rock and Country Western music. Similarly the "close" voicing is best used with jazz-like music. The ultimate choice is very subjective.

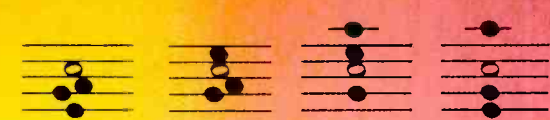
The DL-6 provides you, the creative musician, with a very effective means to personalize your music. The DL-6 is unique . . . only DeltaLab has it.

RULES OF HARMONY

An example of how the DL-6 executes its pre-programmed music theory is illustrated below:

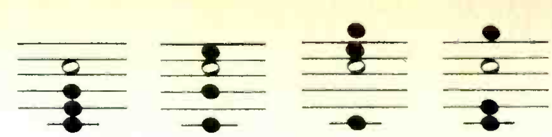
Assume a note "C" is being played in the key of "C". The DL-6 will instruct the first and second DL-5's to pitch shift to either "E", "G", or "A" depending on the Output and/or Voicing controls. The third DL-5 will pitch shift to a lower "C" or "G", again, depending on the voicing control.

"Close" voicing harmonies



DL-5 (1)	I	Lo	"E"	I	Hi	"E"	I	Hi	"E"	I	Lo	"E"
DL-5 (2)	II	Lo	"A"	II	Lo	"A"	II	Hi	"A"	II	Hi	"A"
DL-5 (3)	III		"G"	III		"G"	III		"G"	III		"G"

"Wide" Voicing harmonies



DL-5 (1)	I	Lo	"E"	I	Hi	"E"	I	Hi	"E"	I	Lo	"E"
DL-5 (2)	II	Lo	"G"	II	Lo	"G"	II	Hi	"G"	II	Hi	"G"
DL-5 (3)	III		"C"	III		"C"	III		"C"	III		"C"

Although what is illustrated is a simple C major chord in the key of C, the DL-6 has pre-programmed rules for treating all notes in any key with scale modes that are either major, minor, seventh or Dorian.

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MR&M: Getting back to your own career, when did you actually begin performing?

GJ: In the late 60s. I was performing in high school; I'd be the featured singer in a play or something. I knew I had something, but more important than that I was doing something I enjoyed. After I went through college and the whole art thing, I decided I wanted to play music. In the late 60s it was feasible because there were so many groups forming. You could at least try it, so I did. I started performing all over New York: colleges such as Hofstra and Stony Brook on Long Island; clubs in the city (New York) such as Max's, the Electric Circus, Feenjon, the Cafe Au-Go-Go, the Gaslight, Gerde's Folk City, Kenny's Castaways...

MR&M: What was the reaction like in the early days? Did you experience much racial prejudice?

GJ: It's always been that way, and it's still that way. In fact, it's worse today. Radio doesn't play black music; and the black stations don't play much white music, either. The programmers, the advertisers, the record companies, really want to keep people in a certain world. If a rock station plays a Black record, kids will call up and say, "Get that nigger off the air." I know this for a fact.

MR&M: That really became blatantly true in the disco era; racism was really brought out into the open.

GJ: Yes, and disco wasn't really recognized as a music. OK, some of that music was really trash, but some wasn't.

MR&M: You've always been difficult for the programmers to pin down. They can't call your music Black; they can't call it reggae; can't call it rock, because it's all of those.

GJ: Right, instead of just considering me a musician and checking my music out for its quality.

MR&M: Did your former record company, A&M, have trouble figuring out how to market your music?

GJ: Absolutely. When I came out with the *Ghost Writer* album it was the first of its kind: a mixture of rock and roll and reggae. They didn't know what was on the record.

MR&M: So why would they sign you if they didn't know what you did?

GJ: I think a lot of people get signed before the companies know what to do with them. They sign people and put out a record and then expect the record to happen by itself. There's no real game plan or consideration of how the artist should be marketed. Let's face it: there's no room for artistry in the record business. Certainly not in the States and certainly not during bad economic times. The bottom line is the accounting: credits and debits.

MR&M: Why do you think it's different in Europe?

GJ: I wouldn't say it's completely different. There's just more acceptance of music with more range to it. It doesn't have to fit into a format. There's a lot of music that's very big in the States and is garbage. They don't tolerate that in Europe.

MR&M: How did you come into contact with reggae as early as you did (1970) and why does it appeal to you?

GJ: I just have a love for the music. I have relatives that live in Kingston, Jamaica. I discovered it through them and through this guy who used to play at a gym I go to. It reminded me of that early R&B music that I love. I was inspired by the newness of it: the rhythms and the use of vocal harmonies. I started to investigate it, listening to the Heptones and the Upsetters. Then I went to Jamaica to record part of my Atlantic album. That's when I discovered Jimmy Cliff and Bob Marley, before anybody here knew who they were.

MR&M: Can you talk about your writing style: how you began writing, where you do your writing, etc.?

GJ: I write in different ways. I'm not very successful at writing on the road. I'm very preoccupied with what I've got to do. We did a "live" album on the last tour, so it was very hard to be thinking about new songs. I'm very involved with my show. I write on guitar and I write in my head. I can remember melodies easily. I write my lyrics in any particular way. I've been taking notes on the road for months, so I've got a lot of different ideas for songs. I've got titles, too.

MR&M: So, you like to get a jump on your next project, right? You're already thinking about the next album although the "live" new one is new?

GJ: Absolutely. Once we finish the tour we're currently on, I'll just write incessantly; I'll be writing for two months.

MR&M: Your albums have been well-received by critics, but haven't done overwhelmingly well by commercial standards. Does that bother you or are you satisfied to know you've made good records?

GJ: It's disappointing. For example, *Ghost Writer* is, I think, one of the great records, but nobody knows about it. I don't think I've been given my due. I must say that the fans who come and see me get satisfied with what I do and I'm satisfied with the way they treat me. I almost feel that that's all I can rely on: my fans. I do my "live" performances; I put out the albums. But from there I have no idea what's going to happen. I wish I had more acceptance. I'm not crying over spilled milk or any of that bullshit; I'm just talking facts. I know my record is as good as Bob Seger's "live" album, but he's getting the airplay and I'm not.

MR&M: It seems that the one track they do play, at least on New York radio, is "96 Tears." And that's not even one of your songs. Does that bother you?

GJ: Not really, but they only stick to what they know. They should play "Ghost Of A Chance," though. That, to me, is a really good song. "Graveyard Rock" is a powerful song [Both songs are on *Escape Artist*]. But they don't know anything about playing those kinds of things. They don't know about "True Confessions," "Miami Beach," "We The People."

I feel like a lot of artists feel today. And I feel close to a lot of the new artists; they don't get played on the radio. Elvis Costello has a country album out and it ain't gonna get played on the radio 'cause they're not flexible enough. Graham Parker is a giant and should be recognized. But he's not. It goes on and on. I just know that I have to keep going. I get a lot of pleasure out of what I do, but I get frustrated from time to time. I don't know if I'll tour again after this current one, though. I'm going out and playing my ass off, and people aren't hearing it.

MR&M: Why did you choose to release a "live" album now?

GJ: Why not? It fits into the program. You know, should it come out after the big hit album? Should it *be* the big hit album? I feel that it's a great "live" album; it's a different "live" album. Most "live" albums aren't great.

MR&M: Most "live" albums aren't even "live."

GJ: Exactly. I enjoyed doing this. I had a chance to work with the Rumour on this last tour and it sounded great. So why not put it out as a record? I think the songs were improved upon from their original versions. You can tell from my mood today, though, that I don't want to give a false sense of optimism. I would say that my career is in better shape these days, and I do feel better about that.

MR&M: How has your current record company, Epic, handled you? Are they behind you?

GJ: Certainly more than my other record companies. I must say, though, that there are people at A&M who I respect, like Jerry Moss (president). I don't want to over-complain, though. I'm happy with my life; things are better emotionally, romantically, work-wise; I'd just like to be further along.

MR&M: How did you hook up with the Rumour for the tour that was recorded?

GJ: Well, they've split up now, but originally, Stephen Goulding (drums) and Andrew Bodnar (bass) worked on the *Escape Artist* album. That led to Brinsley Schwarz (guitars) and Martin Belmont (guitars) joining in.

MR&M: Why did you choose to record the "live" album at the Ritz in New York and in Lyon, France?

GJ: I actually wanted to do it in Amsterdam, Hamburg and Paris. But those dates were two weeks apart from each other. And you have to figure out economical ways of doing things, so we decided in the end on doing Lyon, Paris and Montpelier. As it turned out, those were the dates on which we really learned something. Those recordings didn't work out very well, except for the cuts "Bound To Get Ahead Someday" and "Matador."

Once we got to the States we used the Record Plant Mobile, and Bob Clearmountain came over to engineer the French stuff. We had the Rolling Stones Mobile, which didn't work very well for us. It was shoddy and didn't give us what we wanted. I have all good things to say about the Record Plant truck. Dave Hewitt and his organization have done a great service to "live" recording. Clearmountain oversaw that, and we used what we learned—we kept two tracks—and then we recorded at the Ritz and in L.A. Those Ritz shows came across best, not only because we had a great audience and the shows were real consistent, but because you can really hear the bass drum and the bottom. Those three Ritz shows were probably the best I've ever done, consistently, at one time. The stuff we recorded in L.A. didn't work. There were some comparable tapes, but it didn't have the sound quality of the Record Plant Mobile. The Ritz had a big sound—there was a bottom, a top and whole range in between. I have to thank Bob Clearmountain for organizing that.

MR&M: He's certainly a busy man these days. How'd you become involved with him? He's also listed as co-producer of the album.

GJ: Sort of by accident. We co-produced both the "live" album and *Escape Artist*. I've always worried about what the sound in the [control] room would be like while I was working. With Bob, I found out very quickly I never had to worry about that. He's a humble guy, a pleasure to work with. If he doesn't know something, he asks. He doesn't have an attitude of snobbery. And in terms of speed, he's not there to waste your money or to make money for the studio. He's there to do the project. We were both satisfied with the results of *Escape Artist*, although we hoped and expected it to do better.

MR&M: Where did the title of the "live" album, *Rock & Roll Adult*, come from?

GJ: A friend of mine suggested it. He read somewhere that Garland Jeffreys makes the kind of music you might consider adult rock and roll. It meant that the kind of issues I'm talking about aren't the same as those talked about by your basic rock and roller. It's music with an intellectual side. So the idea stuck in my head. "Cool Down Boy" talks about my childhood and about how I am today as a 38-year-old. I felt that *Rock & Roll Adult* was both a serious and a funny title. I thought it had a lightness to it, but at the same time, it represents to me where rock and roll is right now, which is, that 38- and 40-year-old people are making it and making great music. It erases the whole myth of "Is there rock after 30?" There shouldn't be a time limit. There are plenty of people over 30, like Bob Seger, making great music. That's not to say there aren't 18-year-olds and people in their twenties, like the Clash, making great music, too. I was born in the 40s, the music of the 50s is my music, and I'm bringing it into the 80s. And I want to bring it into the 90s, too. The people who grew up with Elvis and Frankie Lymon are into their forties now.

MR&M: Did you ever wish you could go back and record in mono like Frankie Lymon did?

GJ: Yes. I think one of the problems now is that so much music is over-produced. It's over-overdubbed. The technology takes a lot away from the feeling. It's an art form in itself to make the technology work for you. I try to get as much emotion as possible out of the session. I'm not a big fan of overdub sessions. I like to do the vocals "live." I think eleven of the fourteen [vocal] tracks on *Escape Artist* were cut while the basics were being put down. The other three were put on afterwards.

MR&M: Do you ever feel that the technology gets in the way of what you do?

GJ: Well, Clearmountain is into technology, but he hasn't lost his sense of spontaneity.

MR&M: Have you ever studied reggae recording techniques?

GJ: I haven't actually studied it because I feel I have a natural feeling for it. One of my strengths is that I know what not to put on a record. I have a sense of the purity of something like that (reggae), and the success of it relies on the "roots-ness" of it.

For the reggae tracks I used different producers, though, because Bob Clearmountain is not going to become a dub-mixer. He's not even going to try. And he can appreciate that, knowing it's not his work. So on those tracks on *Escape Artist*, Dennis Bovell produced. I look forward to doing more of that kind of music and there may be some surprises on the next album.

MR&M: Can you elaborate on your in-studio working relationship with Bob Clearmountain?

GJ: I fine-tune what he does. I like highs and a brighter sound, and so does he, so I'm always trying to get him to make the sounds brighter for me. He cautions me. He'll move in my direction and then he'll caution me and say it's too thin. He likes bass and he's taught me that you can have bass and still keep the record bright. We generally agree on direction.

MR&M: Were there any problems with the "live" recordings?

GJ: Not really. We taped eight shows; there was no remixing. We just chose the best of what we recorded. I just hope people will get to hear it.

The following day, Steve "Siggy" D'Ambrosia, the stage sound technician for Garland Jeffreys' Ritz performance, explained the stage setup used by Jeffreys and his band. D'Ambrosia began by detailing his own involvement in the music business.

MR&M: How did you get started as a sound technician?

Siggy D'Ambrosia: I studied under Eddie Offord for a few years. He produced acts like Yes and Emerson, Lake & Palmer. I've worked in the Asbury Park, N.J. area since 1975, and worked with Springsteen for awhile. So I was always in touch with Springsteen's people and the Asbury Jukes' people. What happened was Bruce's road manager left him after five years and went with Garland. When Garland needed some people, he called me up. Now I'm Garland's tour manager and sound engineer.

MR&M: Who's playing with Garland at tonight's gig?

SD: Garland, Brinsley Schwarz, Brian Stanley, Stephen Goulding and Carter Cathcart, who used to be with the Laughing Dogs. Danny Federici from Springsteen's band is sitting in tonight.

MR&M: How would you compare working an act like Springsteen to working with Garland?

SD: They're totally different. With Bruce, though, I was just a general helper, while here I'm completely in charge. So with Bruce I helped fix everybody's mistakes and with this I make all the mistakes.

MR&M: You were involved with Garland's "live" album. Why was that recorded at the Ritz and what went into the making of it?

SD: We were here [the Ritz] for three nights, so the conditions stayed the same for three nights in a row. That makes it a lot easier than setting up all over again somewhere else. We had the Record Plant recording truck outside. Bob Clearmountain produced and engineered it, which helped a lot. Plus, this is Garland's home town so it's the best audience response.

MR&M: Did you choose the Ritz because of its sound system?

SD: No, but the system doesn't make a "live" recording sound really "live." It's a really full-range sound. The room mics pick up a very "live" sound.

MR&M: Were there any problems recording here?

SD: We had a problem with New York's power supply. We lasted fine until the last note on the last night, and then we blew out the whole street. That was fun. That was the only problem.

MR&M: Did you work in conjunction with the Ritz sound crew, or did you take over the whole room?

SD: They just give us the main speaker system. I always cover the whole stage setup: the mics, stands, cables, monitors; I do the "live" mixing. The band's very easy to work with; they mix themselves well onstage. They're professionals.

Garland on stage in New York: a question of solving the problem.

MR&M: Can you run through the equipment used at the Ritz shows, including the board and other non-stage equipment?

SD: The board is a Yamaha PM 2000, 32 channels, four effects sends. Also four effects returns for them. Two tape echo units—one is a Shure echo and one's a dub echo, and Eventide 1745 audio delay. Lexicon Primetime for a real long delay, which I use in place of a second tape echo. We use an AKG BX-10 portable reverb unit, which is the closest you can get to a real reverb sound.

Brinsley's guitar setup is the most unique thing onstage. He has three amplifiers—one is very low volume only for the guitar and everything it's doing. That's a little Boogie. Then there's a cabinet on the left for nothing but effects. And the cabinet on the right does all the dry guitar sound. If he puts echo on the guitar it comes out of one amp, while the signal from the other amp still comes out straight. He can also go into stereo and the signal will go back and forth between them. He's got four channels of Harmonizer™, two echo-reverb units, two digital delays, two compressors, distortion boxes, phasers and flangers. He's the only guitar player I've ever heard who has no electronic noise coming through his amp at all.

The bass amp is just an Ampeg SVT setup. The drums are white Ludwigs, the rhythm guitar uses a Boogie head with an Acoustic speaker cabinet. All Brinsley's amps are Boogie amps with Hi-watt slave amps driving the extra speaker cabinets for the dry and effects. There's a Hammond B3 organ with 122 Leslies, and a Farfisa. We usually put one Leslie backstage and mic it for a chamber effect. All the other effects are done by me.

The monitors are supplied by Connecticut Sound Systems in Norwalk, Ct. The side fills are 4520 bass and 4560 midrange, 2482 horn with a slant plate for high and midrange and a 2441 driver on a JBL horn for the highs. The 15-inch speakers are 2420 JBL speakers. The floor monitors are 2420 15-inch speakers with a horn, with a 2482 driver on it, JBL 90°. The drum monitors are two 15s and a "potato masher" with 2482 driver, and two bullets, beaming at your face, ripping your ears off.

MR&M: Any special stage miking techniques?

SD: The only special thing is that we're using Countryman pickup mics, inside the drums, wide out of phase with the microphones actually on the drums. I only use one channel, which helps, because I'm not bringing my own sound system. I only use the one channel, but I still get a powerful shot at it. It cancels out the lingering boom.

MR&M: What is the house system at the Ritz?

SD: It's by Merlin Sound. The speaker cabinets are full-range cabinets. I'm pretty sure it's two 18s, four 10s, two horns and two tweeters in each cabinet. It's a three-way cabinet and the fourth way is passive. There are four cabinets downstairs, two in the balcony and there are horns added on farther back in the balcony on delay.

MR&M: How would the setup differ if Garland were playing in a smaller club?

SD: The stage setup is always the same, except for having problems finding a good place for a Leslie.

MR&M: This room has a cavernous sound when it's empty. How much do you have to adjust when it's filled with people?

SD: It changes completely. There are no seats here, and it's a hardwood floor and cement walls. Also, it's under a balcony, so it [the sound] kind of keeps itself under the balcony and bounces back and forth constantly. When people enter, it shuts down the reverberation almost completely, which is great. I usually don't do much with the sound checks here. I can get an idea if I'm going to have a problem with stage volume, but that's the only thing I really check for. After that I wait for the show to start.

MR&M: What is the house board at the Ritz?

SD: It's a brand new Audioarts Engineering board, 32 x 12. It has twelve mix outputs. It also has two effects sends, which is really unusual for a monitor board. We bring the monitor system; I just supply the effects for the house board.

John Blake,
violin

Avery Sharpe,
bass

McCoy Tyner's string section on Bose.

When John Blake and Avery Sharpe aren't performing with McCoy Tyner, one of the world's foremost jazz pianists, they conduct seminars and master classes on theory and technique at major colleges and music schools nationwide. Here's the advice John and Avery give on amplifying acoustic instruments:

John: "My violin has a naturally sweet and mellow tone, and it can also be very dark. Ordinary speakers don't do this instrument justice—they add an unpleasant edginess, a kind of glare that makes it sound like an electrified instrument.

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Avery: "The first time I used a pair of 802 speakers, I was really surprised by the overall tightness and clarity of the sound. It was a big improvement over what I was getting with my usual 15" bass cabinet.

"Bose 802s give me a very consistent timbre and even response on the fingerboards of both my acoustic and electric basses. I can move up or down an octave or more without sounding like I switched to a different instrument."

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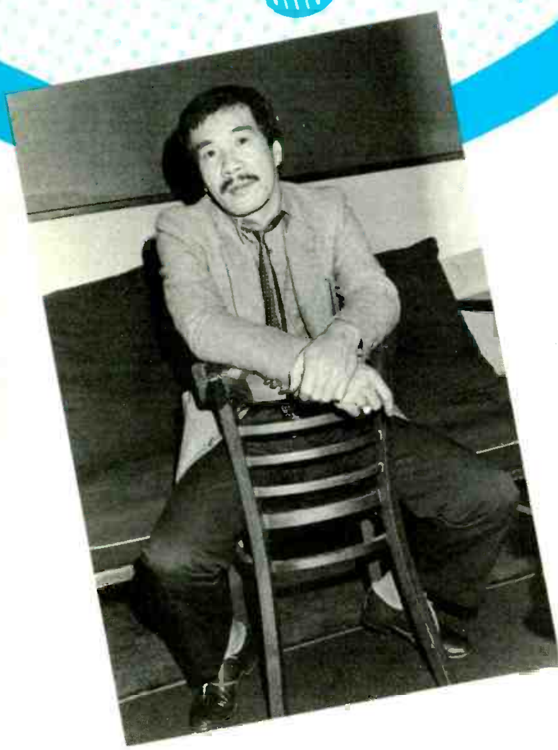
Profile:

Bassist Teruo Nakamura

by Gene Kalbacher



Teruo Nakamura has called himself a "dropped-out artist," but his music with The Rising Sun Band is an integrated palette of tonal colors combining jazz and rock with funk and Latin textures. Born in Tokyo, Japan, Nakamura rejected a career in art (everyone in his family is an artist) and from the age of 15 has set his sights on the world of sound. Since arriving in the United States in 1964, he has amassed impressive credentials as a bassist, composer, bandleader and producer—interdependent roles that qualify him, one might say, as a "dropped-in musician."

Trained in music theory at the Nihon University College of Art in Japan, Nakamura, 39, studied bass with Reggie Workman, a John Coltrane alumnus, upon his arrival in the U.S. and began his professional career in 1969 with the Jazz Samaritans, an early fusion band that included drummer Lenny White, pianist George Cables and saxophonist Steve Grossman. Six months later he caught the ear of drummer Roy Haynes, who recruited the bassist into his Hip Ensemble. Between 1972 and 1975, when Nakamura formed his first Rising Sun Band, he worked with George Benson, Bobbi Humphrey, the late Mary Lou Williams and Stanley Turrentine, the Turrentine association lasting two years.

That experience has proved invaluable. Nakamura has not only recorded six albums as leader of The Rising Sun Band (the most recent is Route 80, *released in Japan on Canyon Records), but also produced a handful of musicians (including Haynes and guitarist Jimmy Ponder) and performed as a bandleader at the Newport and Kool Jazz Festivals and at such prestigious venues as Carnegie Hall and the Bottom Line.*

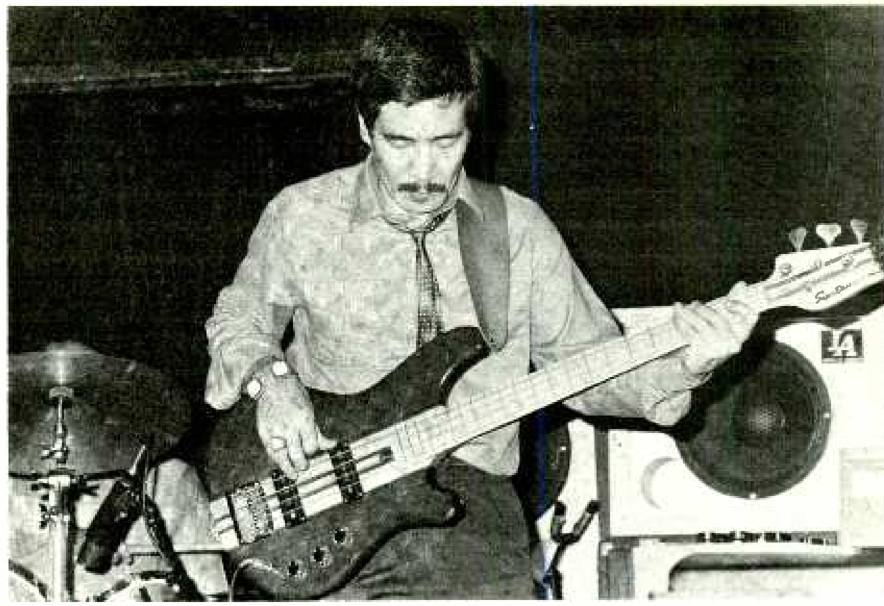
A triple-threat musician, Nakamura's current edition of the Rising Sun features: Bill Washer, electric guitar; Bob Mintzer, flute and saxophones; Gerry Niewood, saxophones; Brian Brake, drums; Leslie "Chuggy" Carter, percussion; Jay Byalick and Mark Gray, keyboards and synthesizers. In addition, the bassist produced Washer's Opening Day *solo LP and co-produced (with engineer Jeffrey Kawalek) Mintzer's* Horn Man, *both for Canyon. At press time, the bassist's plans call for a tour of Japan, his first performing visit since 1964; negotiations are currently underway with a major U.S. label for the domestic release of the above Canyon albums.*

The following interview was conducted in Nakamura's loft in lower Manhattan.

Modern Recording & Music: Both on stage and on record, Teruo Nakamura & The Rising Sun Band creates music that is multitextured yet never cluttered. Although each of the eight musicians is proficient and capable of interesting solos, virtuosic displays are downplayed in favor of an ensemble sound. Unlike other leaders of so-called fusion bands, you rarely take a solo. Why is that?

Teruo Nakamura: Because that's not me. When I was working with Stanley Turrentine and George Benson, they taught me what the bass has to do. One time I met one of my favorite bassists, Leroy Vinnegar, who used to work with Andre Previn and Phineas Newborn. He plays great accompaniment, but he never takes a solo. He influenced me the most.

He and Ray Brown and Paul Chambers all sound good behind the musicians. That's how I want to hear myself. It's hard for me to do things sometimes because I don't like to be out front. I'm very shy, and I just like to make good music.



Teruo Nakamura on stage recently at the Bottom Line in New York City. He is playing one of the three Yamaha basses he uses in "live" performances.

MR&M: While we're on the subject of influences, let's start with your studies under Reggie Workman. What were the principal benefits you derived from that relationship?

TN: When I was in Japan, I thought that all [American] musicians were rich, that they all had two or three Cadillacs. I learned more than just the bass from Reggie. He taught me how to survive.

He took me to restaurants where I could eat nice food; he took me to places where I could listen to good music; he taught me how to relate to musicians. I never knew black people or white people [in Japan].

MR&M: Why did you decide to leave Japan in 1964 and come to the United States? Was it the prospect of owning three Cadillacs?

TN: No. My interest was simply to hear the music, not to play the music. I wasn't thinking of becoming a professional musician. I just simply liked to listen to the music.

MR&M: But surely you could listen to the music in Japan.

TN: You could buy albums, but it's not the same thing as hearing it in a "live" situation.

MR&M: Japan has an amazing communications network and the people are said to appreciate many different kinds of music—everything from fusion and straightahead jazz to rock and pop. From listening to American jazz musicians recently returned from Japan, one can easily get the impression that jazz, a native American artform, is more popular over there than it is here. Is that so?

TN: That's true. Today's Japan is a creation of the United States. They're influenced by good United States culture. They idolize the good things this country has to offer. That's why you can buy so many different albums in Japan that you can't buy here. Still, the greatest music in the world is created by American people.

MR&M: Together for just six months in 1969, your first professional band, the Jazz Samaritans, broke up before it had a chance to accomplish anything substantive, yet each member made his mark with a name musician—yourself with Roy Haynes; pianist George Cables with Joe Henderson; and drummer Lenny White and saxophonist Steve Grossman with Miles Davis. One gets the impression that the Jazz Samaritans were raided, leaving nothing behind.

TN: We did a concert across the street from Carnegie Hall. Miles showed up; Joe Henderson showed up; Roy Haynes showed up. They took the whole band. "I'll take this guy" and "I'll take this guy." [*Laughs*]

MR&M: Your next gig was with Haynes' Hip Ensemble. Your all-instrumental music with The Rising Sun Band, like much of Haynes' work, draws from the harmonic revolution wrought by bebop. Do you see a direct link between the bebop of the late 40s and early 50s and today's fusion music?

TN: Roy Haynes is responsible for the drumming today. Max Roach is outasite too, but Roy is more like the fusion, new type of musician. Music was all fusion, anyway. Bebop was a fusion, too. They just put new names, new labels on it.

MR&M: Having discussed your background, let's turn to your instruments. Let's start with the electric bass.

TN: I play three different Yamaha models: the Super Bass 800; SB 1200; and the BB 2000.

MR&M: What are the essential differences between them?

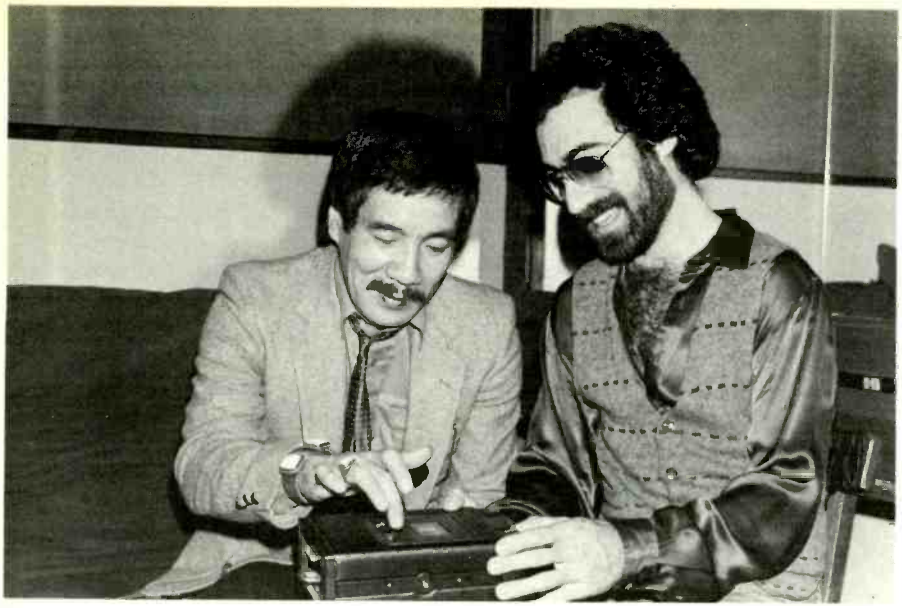
TN: The pickups are different. The BB bass pickup is similar to a [Fender] Precision-type pickup. The 800 and 1200 are similar to the Fender Jazz Bass.

MR&M: You were introduced to your acoustic bass, I understand, by Ron Carter.

TN: Yes. It's an old Czechoslovakian three-quarter bass. With it I have an E-string extension. The low E-string goes down to low C, instead of low E. Ron Carter showed me this bass.

The two pickups I use for the acoustic bass are a Barcus-Berry and a handmade Underwood; I EQ them. The Barcus-Berry is a very old model and it's built into the bridge. I find that the Barcus-Berry has a better bottom and the Underwood a better top.

MR&M: You play the upright through a biamped system, which is rather un-



Bassist Teruo Nakamura relaxing backstage at the Bottom Line with sound engineer Jeffrey Kawalek.

usual. How does this help your music?

TN: I've got control by different sources. I can get clear sound and better control, because I don't manipulate the bottom. When I want to change the midrange, I adjust the midrange; I don't have to mess with the bottom range.

My preamplifier was built by Bill Beaty, a studio engineer who has built stuff for Al DiMeola and Tomita. My power amp is a Yamaha P2200. And for the upright bass I've got a Sansui parametric equalizer. With the biamp system I have four speakers—two are 18-inch and the other two are 12-inch. The 18-inch folded speakers are Electro-Voice; the midrange is also Electro-Voice. The speakers are in cabinets built by Impact Audio. And, man, those speakers give me great punch.

MR&M: What about effects?

TN: Built into my amp rack is an Envelope-Controlled Filter, which was designed by Biegel Sound Lab. This lets me get wah-wah effects and slow sweep sounds. Also, there's a Yamaha E1010 delay system built into the amp rack. In addition, I use a Yamaha System Board 2000, which is a master control center, operated by a foot pedal, which allows me to use chorus, flanging, phase shifting, fuzz and other effects.

MR&M: On stage, I've noticed, you also make judicious use of the Taurus Moog Bass Pedal. How is that operated and what purpose does it serve?

TN: If I want to hit the lower sound, I can get that sound without using my hand. If I'm playing C notes, for example, and I want to get lower C notes, I just press the pedal.

MR&M: Are you the same musician on upright bass as you are on the Yamaha electric?

TN: I try to look at it as just *bass*. But I've been playing the upright much longer, so it's obvious that I feel much more comfortable on it. Plus, who else plays the upright bass with a biamp system?

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MR&M: One of the principal strengths of your band—aside from the players' versatility—is the wealth of songwriting talent. Your two most recent band albums, *Big Apple* and *Route 80*, feature your own writing as well as tunes by Jay Byalick, Bob Mintzer, Mark Gray and Bill Washer. Your leadership role is clearly manifest, yet no one can accuse you of musical intransigence.

TN: I try to be fair with everybody, and I try to listen to their suggestions.

MR&M: Are you more fair with your band members than, say, Stanley Turrentine was with you?

TN: The same. I just want to have a nice time with music. At the moment I'm playing music, I can forget about paying the rent or buying food tomorrow.

I don't like to be *working*. When I'm playing, that's not work. In our band we have some underneath rules—you know, show up at a certain time—but I don't like to tell people what to do. I don't want to hear anybody tell me what to do.

MR&M: When you were with Turrentine, for instance, was it work or play?

TN: That was a pleasure. Stanley Turrentine is one of the virtuosos of the saxophone. He plays his ass off all the time. He gave me things I couldn't buy with money. But after awhile he got tired of me, I got tired of him and I decided to do my own thing.

MR&M: If recording and performing aren't work, what is work?

TN: Moving equipment—*that's* work! Playing music is the easiest thing. You sometimes go through psychological things with musicians or club owners, and maybe a cab driver messes with your head before you get to the job.

But musicians speak our own international, universal language. We don't need to speak in English or Japanese, we just come out with sound. All the musicians in the band are my friends, my partners.

MR&M: Before your band enters the recording studio, do you rehearse the material extensively?

TN: Definitely. It's to my advantage to have a "live" band. Before we go into the studio, we experience the music "live" and see people's reaction. And I think the reason why radio stations across the country are still playing my older albums is because they're very personal.

MR&M: Why did you choose the name Rising Sun for your band?

TN: My name, Teruo, means *rising sun* in Japanese.

MR&M: How much has The Rising Sun changed, if at all, from your original concept?

TN: In the past, every time I played this music I felt pressure because I'm Japanese. It was almost rejected in this country. But now, since the movie *Shogun* was shown in this country, it's changing. Not only that, but there's also a lot more movement of cultures. If you walk around Soho, you can see so many influences from Japan. So many artists get their influences from the Japanese.

MR&M: You have an interesting thumb approach with your right hand on the Yamaha electric. Has that always been your style or is that a recent development?

TN: I started that recently because it sounded good to my ear. Every instrument has a purpose. Each instrument has its own meaning. If you use the thumb, I find that you get a more propulsive sound.

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David Wolford
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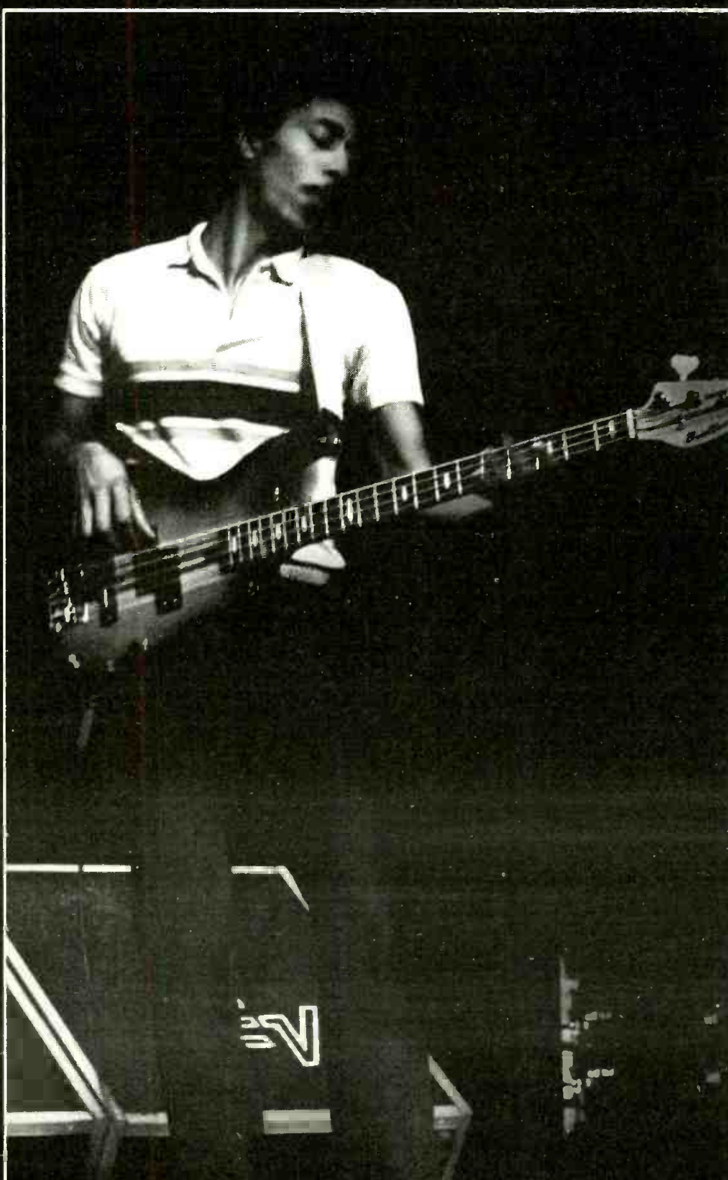

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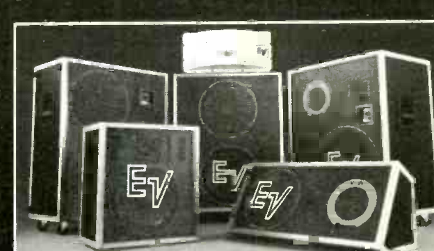
Don't think that good equipment isn't important just because you may be playing for 300-seat clubs instead of 5,000-seat auditoriums. I've played clubs too. Believe me it's just as important. When your bass speakers are *the* bass speakers, you need the best you can get.

With Electro-Voice speakers I can hear it tight and play it loose. That's what the groove is all about.



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MR&M: What, to you, is the purpose of the bass?

TN: The bass is the heartbeat. The saxophone is like a mouth, like a singer singing a song. The drums are the legs. But without heartbeat, the music won't make it.

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For Jeffrey Kawalek, sound engineer for Teruo Nakamura & The Rising Sun Band, "the whole world is rock and roll." On the surface, that seems a strange statement from an engineer-producer who has worked with a wide-ranging, and seemingly disparate, array of musicians ranging from Englebert Humperdink and Dr. Buzzard's Original Savannah Band to George Benson and Blackfoot.

"My approach," says Kawalek, who served as chief engineer at the House of Music in West Orange, N.J., for six years before starting his own Boogie Hotel Studios in Port Jefferson, N.Y., "is that it's all rock and roll. If it's classical, it's rock and roll; if it's jazz, it's rock and roll. It's all just wide-open, happening music."

Classically trained on the piano and later on guitar, the effervescent 31-year-old formed his own band, the Epsilons, as a teenager with Max Weinberg, now the drummer for Bruce Springsteen's E Street Band. After he was graduated from Tufts University, where he served as program director of the radio station, he worked for several studios in the New York area before taking the 24-track responsibilities at the House of Music. The following interview took place at Nakamura's loft.

MR&M: Working with Teruo Nakamura & The Rising Sun Band, what special challenges do you face?

JK: The fact that Teruo plays both electric and acoustic bass is somewhat of a technical challenge. The acoustic bass is one of the most difficult instruments to record because you have to have all the notes projecting with the same quality and the same impact and punch.

As far as sound quality, with the slightest whisper you can get this huge vibrating C note. The extension on Teruo's E-string takes it down to a C.

The biggest problem is that we're dealing with state-of-the-art technology. It's difficult to deal with a process where the final result, the record, is the most primitive step in the whole chain; it's the weakest link. You're taking this marvelous, incredible FET technology and microcircuitry and this fabulous electronic equipment that can capture practically every signal from DC to light.

"Today's Japan is ... influenced by good United States culture. They idolize the good things this country has to offer."

MR&M: For that reason, must you overcompensate when recording?

JK: Sure. You can get incredible things on tape, or capture it digitally, but you still have to deal with getting it on a record. You have to limit it down, keep it in proportion and keep it sounding natural and open.

MR&M: When you're working with the band in a club or concert situation, what are your major concerns?

JK: In a "live" situation, it's a big band and there are a lot of people onstage. They cover so much territory levelwise; they can go from real quiet and subtle to wide open, with everybody screaming and wailing. It's not real structured music. A lot of it is free and spontaneous. So it's not just matching parts.

MR&M: When the music is free and spontaneous, can you, as an engineer, also be free and spontaneous?

JK: No. You have to be the one in control because you're holding it all together. If the sax player is freaking out, making sound effects and working something against the percussionist, you have to be aware of that and keep everything in its perspective.

MR&M: How do you translate Teruo's sound from the stage to the studio?

JK: I treat the studio experience as a "live" experience. For me, it's like painting a picture. "Make that a little more blue." [*Laughs*]

The band is getting to the point where it is really aware of being a band. There's a big difference between a band and studio musicians playing "live." In the studio, you try to capture the energy, the excitement and all the sound of a "live" performance, and then put it on a record. That's what makes it special.

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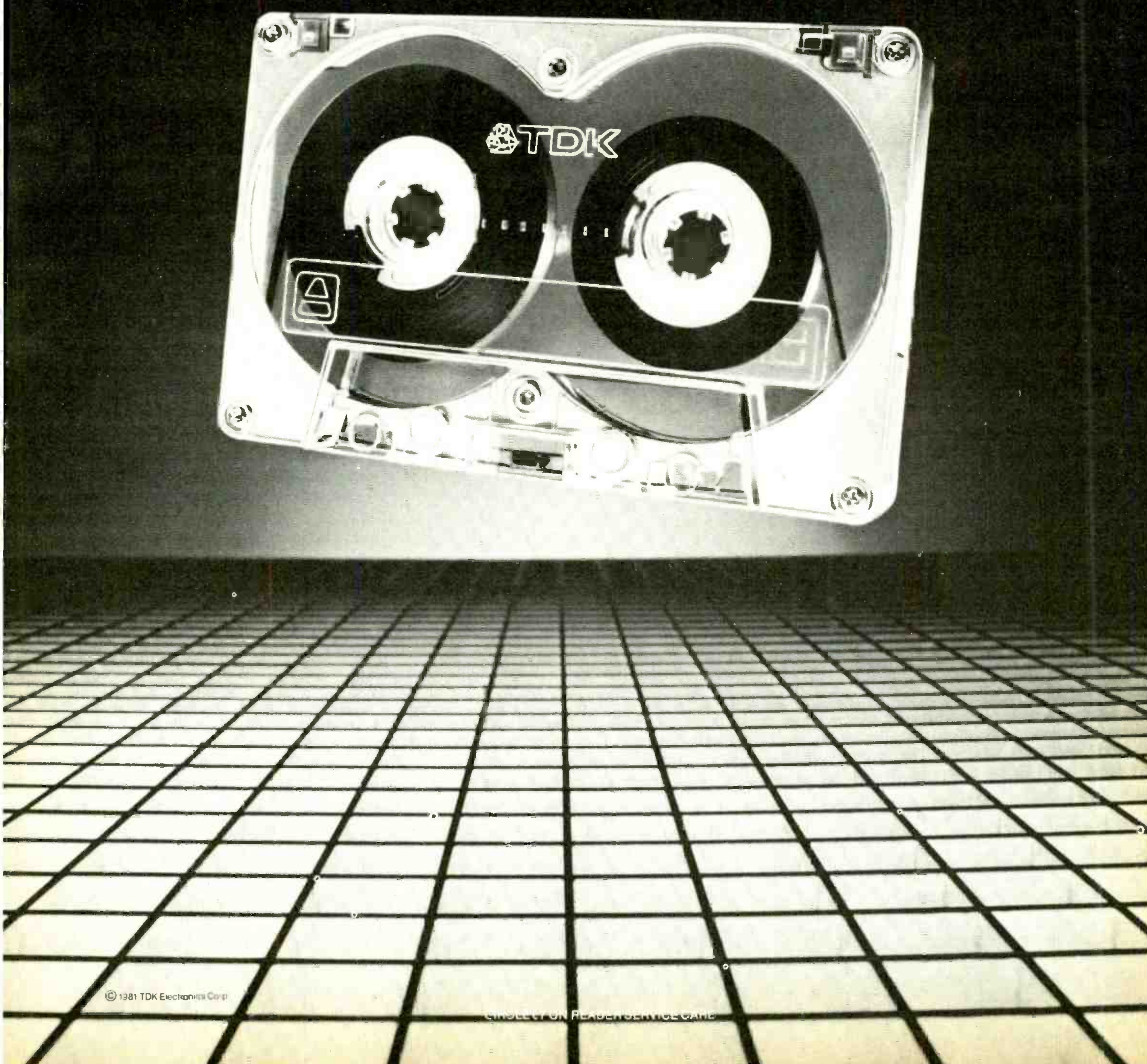

Tascam System 20

By Craig Anderton

You're probably wondering why "Notes" is reviewing a product that might seem more appropriate in another section of this magazine. That should be your first clue that the Tascam System 20 is not your average, everyday mixer. It embodies a specific philosophy towards mixing consoles which I happen to share, but which other musicians might find intimidating or even confusing—especially if you think you already know all there is to know about mixers. The System 20 forces you to think about how the actual process of mixing works, from the moment a signal leaves a microphone (or tape recorder track) to the moment it shows up on your half-track mixdown deck (or P.A., for that matter). This is no plug-in-and-forget black box, but rather...well, read on for the whole story.

WHAT is IT? Forget about mixers for a second, and consider the synthesizer. The first synthesizers were experimental modular systems, designed so that musicians could define sonic parameters such as pitch, harmonic content, instantaneous level changes and so on. At the synthesizer's infancy, no one really knew what direction this instrument would take; some felt it would remain a laboratory curiosity for academicians, while others felt that only big-deal rock stars would have the necessary bucks to actually use a synthesizer in "live" performance or in the studio.

These first synthesizers were not musical instruments in the usual sense; rather, they were more like an audio Erec-

tor set, where you had a large number of basic building blocks strung together with patch cords. By altering the way you patched the modules together, you could achieve virtually any sound you could conceptualize.

Of course, these giant modular systems—while amazingly flexible—had certain limitations. The biggest problem was set-up time, since it could take an hour or more to obtain a particular sound patch and tweak it up for the "right" effect. "Live" performance in particular was difficult, what with trying to go from, say, a cosmic explosion to a dulcet string sound in a matter of seconds. In fact, it just couldn't be done except under the best of circumstances.

Eventually, certain synthesizer patches (trumpets, strings, electronic percussion) became far more popular than others. As a result, many manufacturers ditched the modular concept completely, and went for a *normalized* approach where the synthesizer still had many of the same basic blocks used in large modular systems, but they were all internally pre-patched in a specific, non-alterable way. Some synthesizers hedged their bets by including patch points that allowed for a certain degree of patching flexibility, but still, these normalized instruments were more like the public's idea of a "musical instrument" than the modular "audio laboratories" which preceded them.

How does this relate to mixing consoles? Simple. Virtually all the mixers you encounter these days are normalized with fixed functions. In an 8-input board, you're going to hit eight mic preamps, equalizers, faders, aux busses and so on in a fairly predetermined way. Like the normalized synthesizer, these mixers offer rapid set-up time, easy usage and a limited flexibility. In situations where studio time costs a zillion dollars a minute, a mixer *has* to be normalized to allow for rapid operation. Sure, maybe you're going through some preamp stages you don't really need in the process; but what with professional tape recorders being so quiet, and the use of noise reduction, a couple of dB here and there aren't very significant.

Which brings us to the System 20. Tascam has taken a bold, perhaps even controversial, step by totally rethinking the mixer and breaking it down into a modular mixing system. Forget about switch assignments—there aren't any; all assignments are made with patch cords. In fact, the whole setup is strung together with patch cords, just like a modular synthesizer.

I should mention right now that to adequately describe the many ways in which the System 20 works would be an impossibility, given the space limitations. Instead, this review will concentrate on the conceptual aspects of the System 20; if you want an in-depth look at applications go to your Tascam dealer and check out the (excellent) documentation.

The System 20 compromises four basic modules. A typical system could use anywhere from one to four of these modules. The four modules are:

- The MM-20 Main Mixer: This is a 6-input (two mic inputs, four line inputs) basic console with six output busses and two line outputs. *Figure 1* shows the block diagram. The MM-20 lists for $395.
- The MU-20 Meter Bridge: This contains four VU meters which can monitor various points in the system; it derives its power from the MM-20. Again, you patch these in where required rather than committing them to one particular section of the signal path. The MU-20 lists for $150.
- The PE-20 Parametric Equalizer: This module includes four 3-band "parametrics." (I put parametrics in quotes because there is no resonance control for any of the bands, and the upper frequency stage only has a boost/cut control whose frequency is fixed at 10 kHz.) The midrange section is variable from 1 kHz to 8 kHz and has a boost/cut control, while the low frequency section is variable from 60 Hz to 1.5 kHz and also has a boost/cut control. There is no bypass switch for any of the four parametrics. The PE-20 has its own power supply, making it suitable for stand-alone applications as well as being part of the System 20. The PE-20 lists for $350.
- EX-20 4 Microphone Expander: Since the MM-20 only includes two mic transformers, if you're into multi-miking this expands the MM-20 for four extra mic inputs. List price of the EX-20 is $325.

Ground Rules

Before you read any further, we need to get several things straight. The System 20 is designed for a specific type of recording situation, so if the following doesn't describe the kind of recording you're into, skip ahead to the next article.

Rule #1: The System 20 is for home/budget recording. Due to the fact that all these components patch together, it takes more time to use the System 20 than other mixers. With other mixers, you often just flick a switch to get a signal from point A to point B; with the System 20, you have to actually patch the things together. Since home recording doesn't have the time vs. money constraints of commercial recording, the System 20 is most suited for this type of recording.

Rule #2: The System 20 is optimized for a single artist working in a 4-track environment. There is a new breed of musician who is not just the artist, but also the engineer and producer. For these people, the conventional mixing console wastes a tremendous amount of money. For example, what good is a 6-input mixer where each channel has a mic preamp? If you have six musicians, those preamps might come in handy. But if you're a single musician, the odds of needing more than two mic inputs at a time are remote. So, the System 20

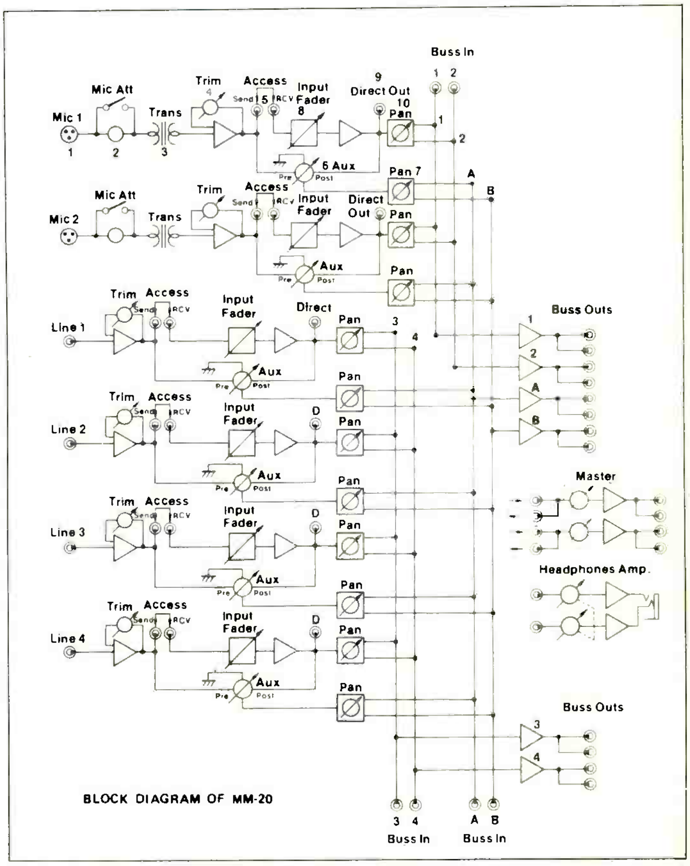

Fig. 1: Tascam System 20: Block diagram of MM-20.

"The System 20 saves bucks by eliminating mic preamps and mic transformers. . ."

saves some bucks by eliminating all those unnecessary mic preamps and, more importantly, mic transformers.

Rule #3: The System 20 is for the musician who demands the ability to squeeze every last dB of performance out of "budget" equipment. The System 20 makes it easy to bypass the mixing circuitry entirely when recording, allowing you to patch the input preamp's output directly to the tape recorder. This saves you several dB of unnecessary noise build-up. Or, take the equalizer: you don't always need EQ on each channel, so with the System 20, you patch in EQ only when required.

Rule #4: The System 20 is for the musician who finds flexibility more important than convenience. For example, you can set up the aux bus as a stereo cue mixer while recording, and as a special effects bus (say, echo and reverb) during mixdown. Again, this saves you bucks since you don't have a lot of redundant, dedicated circuitry; but it does require more time to patch the System 20 into the desired configuration.

Rule #5: The System 20 assumes you've got a certain amount of brainpower and are willing to experiment. The System 20 is by no means an "obvious" mixer. It will take you some time to learn how to use it, there will be times when you get lost in the patch cords, and there will be times when you'd just love to replace some of those patch cords with switches. On the other hand, like any good musical instrument, as you master your instrument you can make it do your bidding in your own unique way. This is what we call "style," and the System 20 is one of the few (perhaps the only) mixer where you can actually impress your "style" on the mixdown hardware. I could very easily see two musicians comparing System 20 patches, for example, and learning different things to do with the aux busses, line outputs and bus inputs.

So much for ground rules; now it's time to evaluate the hardware.

HARDWARE EVALUATION: Like all Tascam produces, the System 20 uses RCA phono jacks. The styling is similar to their Model 144 "Portastudio," with hi-impact plastic construction and intelligent use of color. While I wouldn't drop any of the System 20 components out of a 14-story window, if you're even partially nice to your equipment the construction is certainly adequate (and less expensive than metal enclosures).

I'm going to keep the hardware evaluation fairly short, because these days, the state-of-the-art is such that most mixing consoles within a given price range have reasonable specs and offer useful functions. For example, the System 20 is not the quietest mixer in the world, nor is it the noisiest. It is, however, totally compatible with budget four-track equipment in the sense that the performance of the tape, tape recorder and artist/engineer—*not the performance of the System 20*—will tend to be the limiting factor. The MM-20, EX-20 and MU-20 will give you no unpleasant surprises when you use them, and as far as I'm concerned, give performance that is commensurate with the price. The PE-20 is more suitable for a subjective evaluation, so let's proceed to that.

PE-20 EQUALIZER: Equalizers, by their very nature, almost invite a pro or con opinion. In my own studio, I rely on a variety of outboard equalizers rather than multiple equalizers that are all the same. Although my mixing board has a general equalization shaping stage for each channel, for critical equalization I use either the PAIA Parametrics, synthesizer filters or phase shift networks. Because of this particular bias, I applaud Tascam for breaking the equalizers away from the mixing console and making them patchable. This allows you to use their equalizer, or any other equalizer that tickles your fancy, in conjunction with the MM-20.

As far as the equalizer itself goes, it's good for some applications and not so good for others. I tried the EQ with several instruments and types of program material; it seemed most suited to general sound shaping. The PE-20 makes a particularly good EQ for voice. Most of the time, my main variables with voice are the high frequency sheen, upper midrange definition (for greatest intelligibility) and bass formant. As it so happens, the PE-20's controls are perfectly designed to cover these three "magic spots." The EQ also seemed well suited to guitar and bass for general sound shaping, but lacked the ability to do deep, sharp cuts (often needed with guitar), which limited its use for special effects. With program material, the PE-20 was fine for general tonal alteration, but again, the lack of resonance didn't allow for such things as notching out hum, putting one drum back further in the mix, notching out a resonance with an acoustic instrument or over-zealous synthesizer filter and the like. So, my overall opinion is that the PE-20 is a definite step up from the bass/treble, boost/cut kind of control, but you might want to have a couple stages of true parametric EQ around as well for situations that require more specific equalization. Of course, this just goes to show the wisdom of the modular approach: you can patch in any EQ, or mic preamps, or metering circuits you want.

TECHNICAL EVALUATION: With respect to noise, the System 20 meets "musician's specs." From line input to bus output, the S/N ratio is quoted as 75 dB weighted, 70 dB unweighted, while the mic preamp inputs are 65 dB and 60 dB, respectively. Cross talk in each of the units exceeds 60 dB

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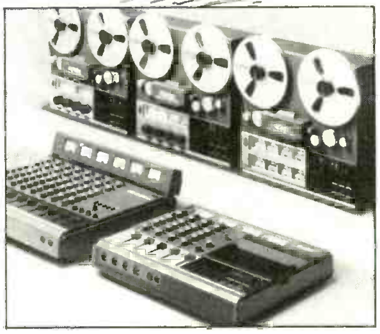

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at 1 kHz. Input and output impedances are well chosen to allow you to do lots of bizarre patching and Y-ing if that's your thing, and there really don't seem to be any "loose ends" in the System—everything works well with everything else. While the System 20 is clearly not the ultimate expression of mixer technology, for the price and the intended application it does its assigned job well.

Also, the documentation supplied with the System 20 deserves special commendation. First of all, it is friendly; it feels like somebody is sitting there next to you. While some of the parts are tough going, and will require some concentration on your part, the effort is well worth it as it will educate you in the process. All that text may all seem a bit confusing at first, but something like the System 20 is a device you have to *learn* to use. Reviewing the owner's manual while fiddling with the dials should get you up to speed in the shortest amount of time; don't *only* read the manuals or *only* fool with the knobs, hold the documentation in one hand and a knob (or patch cord) in the other.

Before getting off the subject of documentation, I'm sure the more technically-minded *MR&M* readers will appreciate the inclusion of complete schematics. Other companies tend to be positively silly about this; c'mon folks, let us know what's in these boxes. We paid for them and should have the right to modify them for our own purposes. Thank you, Tascam, for recognizing that some of us *like* to know about our equipment.

OVERALL EVALUATION: If you're afraid of patch cords, don't like to think about input and output impedances, and sold your modular synthesizer to get one with push buttons, then this is *not* your mixer. (Come to think of it, in that case *MR&M* probably isn't your magazine either.) On the other hand, maybe the System 20 *should* be your mixer so that you'll learn to not be afraid of patch cords and experimentation.

Personally speaking, I don't hold out much hope for the System 20 achieving mass acceptance; a lot of people are simply too lazy to get into something as undedicated as the System 20. Modular synthesizers don't sell as well as the push-button types; many musicians are stymied by the modular nature of rack systems, preferring to buy one or two effects boxes instead; and to top it all off, in this age of instant gratification people seem to "want it all now," and the System 20 is not all about breaking speed records in the studio.

But for those of you who work mostly alone in a 4-track environment and would prefer to squeeze every last possibility out of a mixing setup, the System 20 was obviously designed with you in mind. It's also good for those on a budget, since you can buy it a piece at a time. The MM-20 is a perfectly respectable 4-track mixer, and you can add the other modules—or even build your own peripherals and make them part of the system—as you go along. This is definitely a mixer for people who regard the studio as a collection of modules, each of which can be connected together in almost infinite ways. It will be interesting to see if the public agrees that a modular approach, while less convenient, is still the most flexible way to deal with systems involving musical electronics.

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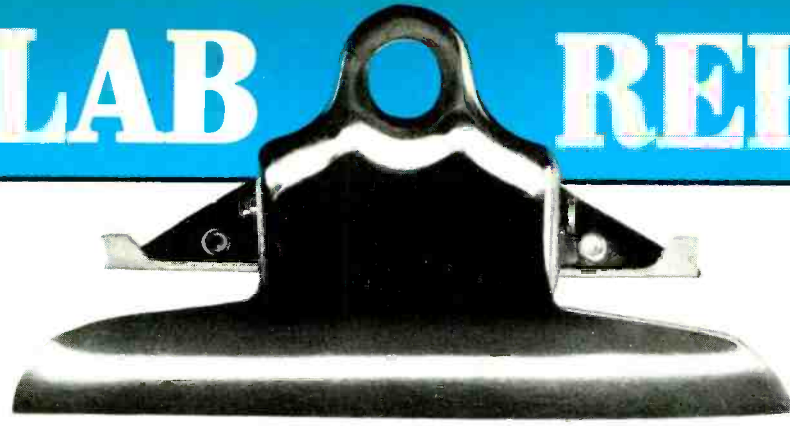
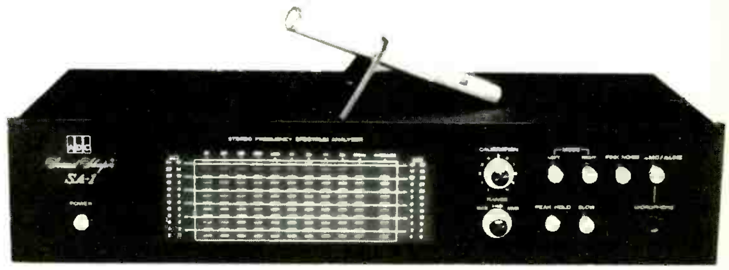
MODERN RECORDING & MUSIC **LAB REPORT**



NORMAN EISENBERG AND LEN FELDMAN

ADC SA-1 Spectrum Analyzer



General Description: The Sound Shaper® model SA-1 from Audio Dynamics Corporation (ADC) Professional Products Group is a real-time spectrum analyzer with a built-in pink-noise generator. It is supplied with a compensated microphone. Probably the lowest-priced real-time analyzer on today's market, it is suited for "voicing" a listening room, and for measuring both program sources and direct sound sources.

The display consists of eleven vertical rows of LEDs, each row containing twelve LEDs. The first ten rows correspond to bands on frequencies of 31, 62, 125, 250, 500, 1 K, 2 K, 4 K, 8 K and 16 K Hz. The last row provides an average band for all audible frequencies. The vertical ranges may be selected for total spans of 12, 24 or 36 dB with each LED representing 1, 2 or 3 dB, respectively. Above the range selector knob is a calibration control which—when used in conjunction with the range switch—helps keep the display conveniently centered for easy interpretation and for associated adjustment on a separate equalizer. Additional controls include peak hold and slow-display action switches. The peak hold may be used to hold the display for about ten seconds; the slow display switch enables a study of averaging effects for facilitating measurements of musical signals. For all input signals other than a microphone source, there are two mode switches that can choose left or right channels or both simultaneously. Also on the front panel are the pink-noise switch; a mic/line selector switch; and the microphone jack. The rear of the SA-1 contains the line inputs, the pink-noise outputs, the unit's AC power cord and an AC convenience outlet (unswitched). The microphone is an electret condenser type with rated impedance of 600 ohms, omnidirectional pattern and is powered by a 1.5 volt UM-5/size N battery.

Test Results: Although the ADC SA-1 is the lowest-priced real-time spectrum analyzer we know of today, there is nothing "cheap" about its performance which either met or exceeded published specifications and in general proved to be accurate and reliable for the device's intended uses. Our lab is equipped with a much more expensive real-time analyzer that operates in either third-octave or full-octave manner, and when we compared the results obtained with the ADC SA-1 against those obtained on the costlier analyzer in its full-octave mode, the two sets of results were virtually identical.

For those readers who may not be familiar with just how a real-time analyzer may be used for adjusting an equalizer, *Figure 1* shows how the SA-1 would have its

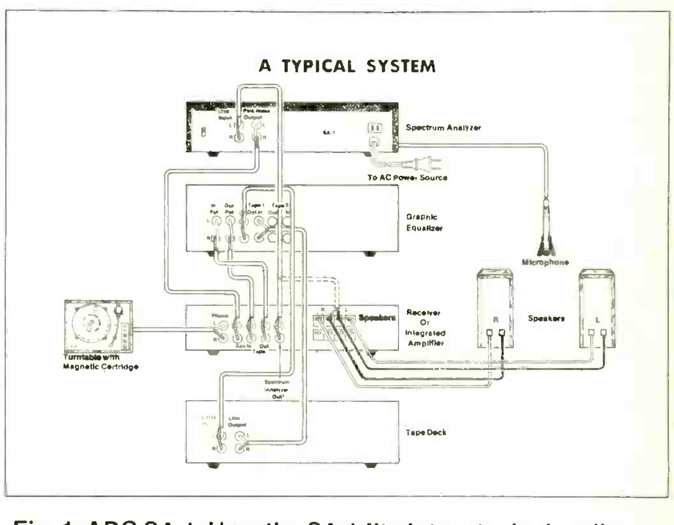


Fig. 1: ADC SA-1: How the SA-1 fits into a typical audio system.

pink-noise output connected to a typical home audio sound system. *Fig. 2,* in turn, shows how the device would be used to "voice" a listening room. With pink noise heard over the loudspeakers, and the SA-1's microphone positioned where one would normally listen to the sound system, the graphic equalizer of your choice is adjusted—band by band—until all the rows of LEDs on the display panel of the SA-1 are illuminated to the same vertical level. A helpful feature in this application is the three-position range switch that changes the LED sensitivity from 3 to 2 to 1 dB, so that the user can easily adjust the detail of the display and thus more readily zero in on flat response. Centering the display via the calibration level control also helps in this regard.

General Info: Dimensions are 17 inches wide; 3 5/8 inches high; 9 inches deep. Price: $230.

Individual Comment by L.F.: The SA-1 is both reliable and low-cost. My only reservation regarding it has to do with its lack of any meaningful sound pressure calibration. That is to say, while it has an "averaging" LED row that provides a single indication of overall sound level, there is no way of relating that indication to actual SPL readings in dB. Pink noise can be deceiving, and I have found that in analyzing and voicing a system it is a good idea to do so at absolute sound-pressure levels that are typical of those that will be used when listening to music via the same system. In the case of the SA-1 I had no way of knowing whether I was listening to an absolute SPL of 70, 80 or 90 dB or even more. It would have been a relatively easy matter for ADC to have provided a rough calibration printed around the already provided "Calibration" control in dB/SPL instead of in arbitrary numerals from 0 to 10. The calibration could have been related to the microphone input only, if necessary, since there would be no need for it when using the line output.

But perhaps I am forgetting that we are talking about a product that costs only a little more than $200. I guess we're all getting spoiled as electronic technology advances, and we keep getting more and more for our money, countering the inflationary trends that pervade all other industries.

MEASUREMENT OF LISTENING ROOM ACOUSTIC CHARACTERISTICS

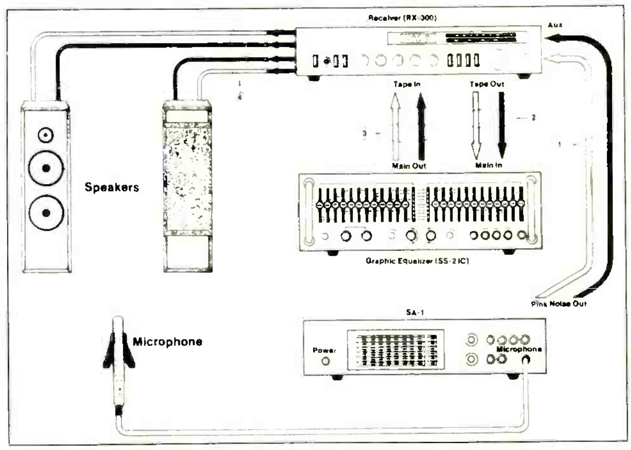


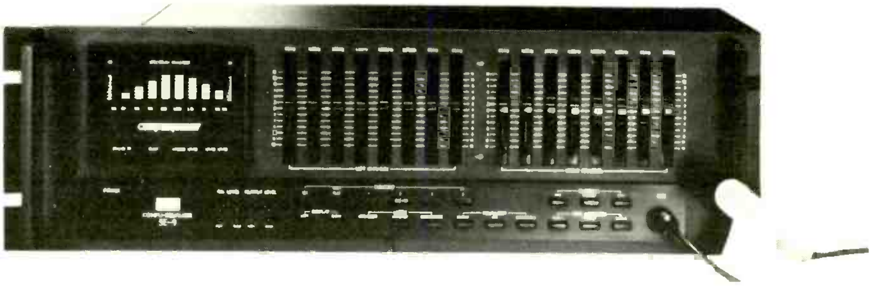
Fig. 2: ADC SA-1: Connecting the unit to allow for proper room voicing.

Individual Comment by N.E.: Contemplating the ADC SA-1 it takes a bit of mind-jogging to recall that when real-time analyzers first appeared some years ago they were priced near the $1,000 mark and were regarded as fairly esoteric devices for use not only by professional sound persons but well-heeled pros at that. Here is a unit that does the job remarkably well, is priced well within the reach of a wide market of sound enthusiasts (pro, semi-pro, home audiophile), is easy to set up and use and should encourage a lot of owners of equalizers to the proper and correct use of those equalizers. I give this product a very cordial pat on the back, although I do agree with Len that the SA-1 would be more useful if it included actual dB/SPL markings. Without these, you still need a dB/SPL meter to really know where you're at in terms of actual sound levels, and especially for checking speaker performance.

ADC SA-1 SPECTRUM ANALYZER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPECIFICATION	LAB MEASUREMENT
Display frequency accuracy:		
from 31 Hz to 1 kHz	± 10 %	± 6%
from 2 kHz to 16 kHz	± 5%	± 5%
Display tolerance at 1 kHz:		
12 dB range	± 1 dB	confirmed
24 dB range	± 2 dB	confirmed
36 dB range	± 3 dB	confirmed
Microphone input sensitivity	0.5 mV	0.4 mV for full display
Line input sensitivity	150 mV	150 mV for full display
Pink noise output level	150 mV	160 mV
Frequency response at line in	± 0.5 dB, 31 Hz to 16 kHz	confirmed
Frequency response at mic in	± 3 dB, 31 Hz to 16 kHz	confirmed

Sansui SE-9 Equalizer/Analyzer



General Description: Sansui's SE-9 is a combined stereo graphic equalizer and real-time spectrum analyzer. Microprocessor controlled, the device can set equalization automatically or manually, and it has a memory for storing four EQ curves. The SE-9 incorporates a pink-noise generator and it is supplied with a calibrated microphone. In addition to its use in voicing a room and analyzing sound fields, the SE-9's connections and front-panel switching permit interfacing with a sound system using two tape recorders, with options for dubbing from either recorder to the other with or without compensation, and for direct recording with both options.

The equalizer itself provides eight bands on each channel, with center frequencies at 80, 160, 315, 630, 1.25 K, 2.5 K, 5 K and 10 K Hz. The sixteen sliders cover a span from +12 to −12 dB with markings for each 2 dB. To the left of the sliders is the display panel consisting of eight rows of LEDs, eight LEDs per row. Each row is marked for a frequency corresponding to those of the sliders; the vertical markings are for +3, 0, −3, −6, −9, −12, −18 and −24 dB. In addition to the spectrum information, this display panel includes four indicators for standby, "busy" (the unit's computer is working); "under level" (the pink noise output is too low); and "over level" (the pink noise output is too high).

Operating controls are grouped along the lower portion of the front panel. At the left is the power off/on switch, followed by two control knobs, one for the pink noise and the other for the unit's output level. Several push-button switches come next. Six handle the device's memory and include a "memory-set" for entering EQ curves into the memory; "flat," for allowing the EQ controls to be automatically adjusted by the computer; and four buttons to recall any of the four EQ curves pre-stored by Sansui and known as the "Sound Menu," or alternate curves derived by the owner and substituted for those curves.

The buttons below these handle additional functions. One pair for the display can choose left or right channels or both simultaneously. Another group of three switches selects the mode: in line/reset, a source connected to the unit's line input will be reproduced; in analyze mode, the pink noise will come through the speakers for pickup by the microphone and spectrum analysis via the LED display; in compensate mode, the equalizer portion comes into play.

The equalizer itself has three mode switches: defeat; on; and recording (pressing the last switch in this group permits recording or dubbing with the equalizing effects added to the taped copy).

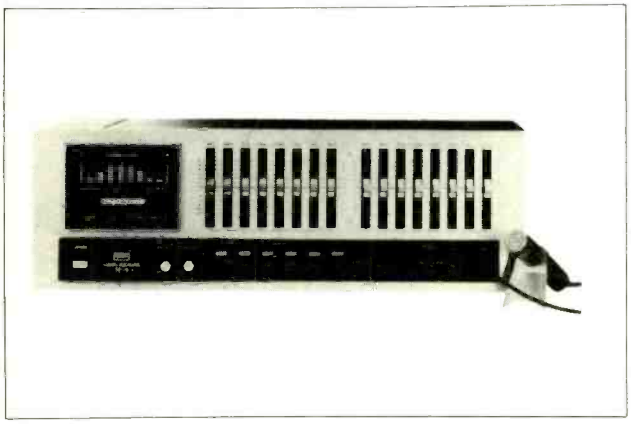
There are three dubbing switches. One permits dubbing from tape-1 to tape-2; another permits doing the opposite (2 to 1). The third switch here may be used for reproducing the signal from the system amplifier. If two tape decks have been connected to the SE-9's outputs, both can be recorded onto at the same time.

Finally there are three tape monitor switches for selecting tape 1, tape 2 or the source. The unit's microphone input jack is at the extreme right.

Rear panel signal connections include stereo pairs of pin jacks for input from, and output to, a stereo amplifier; plus record and play hookup for two stereo tape decks. Also at the rear are the device's AC power cord and an unswitched AC convenience outlet. The SE-9 is supplied in a metal case. SE-9s with silver panels are supplied with four feet for shelf or cabinet installation; units with black panels are provided with EIA-standard rack mounting adapters that fasten to side holes and are fitted with handles. The microphone supplied with the SE-9 is an electret condenser type that is powered by a 1.5-volt size "AA" battery.

In its automatic room-equalizing function, using the pink-noise output, the SE-9 adjusts itself with each of the sliders moving up or down accordingly until the built-in computer "tells" the device that flat response has been achieved.

The "Sound Menu" refers to the four pre-stored EQ curves supplied with the unit. One such curve is supposed to provide suitable equalization for rock and pop; another, for classical music; a third for disco sound; the fourth is for preparing cassettes for playback on car stereo systems and portable cassette players. Any or all of these curves can be replaced by the user with new EQ curves and stored in the unit for recall later. However, should the SE-9 be disconnected from its AC power



Sansui SE-9: Front view of unit with alternate brushed metal finish (minus rack-mounting hardware).

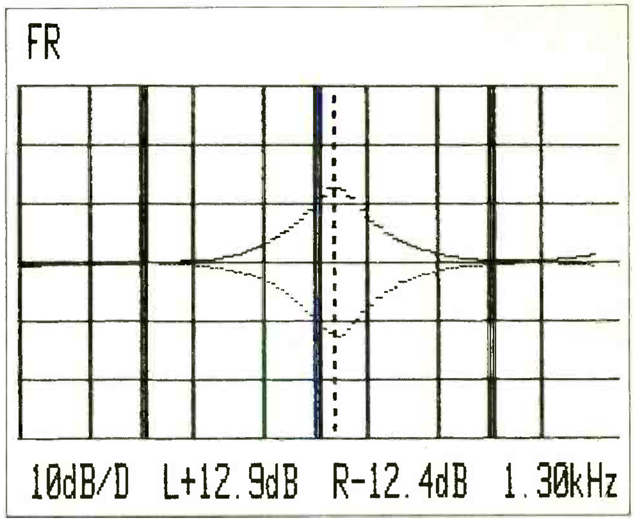


Fig. 1: Sansui SE-9: Typical boost and cut range of single control (1.25 kHz).

source, the new curves will be erased, and the original "Sound Menu" curves will reappear when the device is again turned on. The entire process of analysis and automatic adjustment takes less than half a minute.

Test Results: Lab measurements readily confirmed or exceeded published specifications for the Sansui SE-9. Especially commendable was the unit's extremely low distortion, well below rated level across the audio band. EQ center frequencies were accurate, and the boost/cut range of the EQ sliders was a very generous ±12 dB. S/N, measured at 100 dB for a 1-volt output, was 10 dB below spec, but still very ample. The spectrum analysis display, while not of an order of sophistication or ultimate accuracy as would be found in a specialized real-time analyzer, was adequate in providing an overall visual image of system response and of confirming flat response or deliberate departures from flat response, as applicable.

Considering the equalizer as such, as an 8-band graphic unit, performance was just about superb. *Figure 1* shows a plot of the symmetrical boost and cut characteristics of one of the control bands (the one centered at 1.25 kHz). *Fig. 2* is a 20 kHz spectrum analysis of the pink noise signal provided by the SE-9. Note that the 3-dB-per-octave slope is, by definition, what makes this random noise signal a "pink"—rather than a "white"—source.

Figure 3 (A, B, C and *D)* depict the four EQ curves supplied as the "Sound Menu." Bearing in mind that the vertical divisions in these graphs represent 10 dB each, you may understand Sansui's thinking here. For example, the rising mid-to-high frequency portion of *Fig. 3D* is intended to compensate for the fact that car speakers may be mounted behind the listener, thereby attenuating the relative amplitude of middle and high frequencies.

Figure 4 shows left and right channel correction provided by the SE-9 for a given listening position in one of our studios.

General Info: Dimensions are 16 15/16 inches wide; 5 7/8 inches high; 13 inches deep. With rack-mount adaptors (black panel models only), width is 18 15/16 inches. Weight is 14.6 pounds for silver panel model; 15 pounds for black panel model. Price: $700.

Individual Comment by N.E.: A lot of product is offered for the price here, and it's all very good indeed. To the credit of the SE-9 is the fact that it offers separate equalizer facilities for each stereo channel, with fine symmetry in the shape of the waveform on either side of the center frequency; its utterly low distortion; its versatility for applications in system and room equalization as well as in sound-tailoring as desired for tape recording and

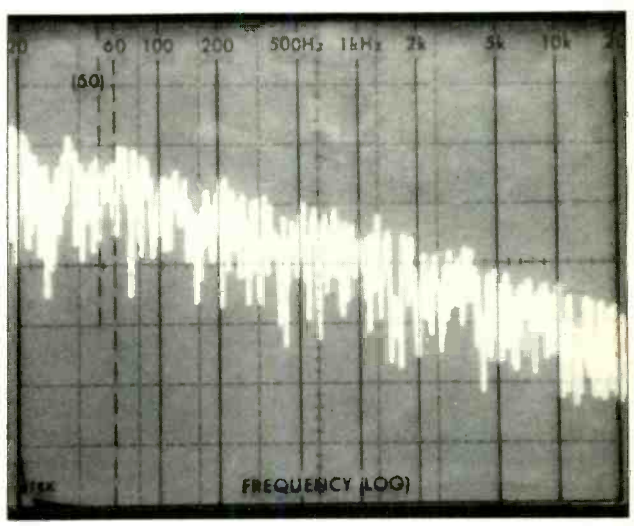


Fig. 2: Sansui SE-9: Spectrum analysis of pink noise signal provided in unit.

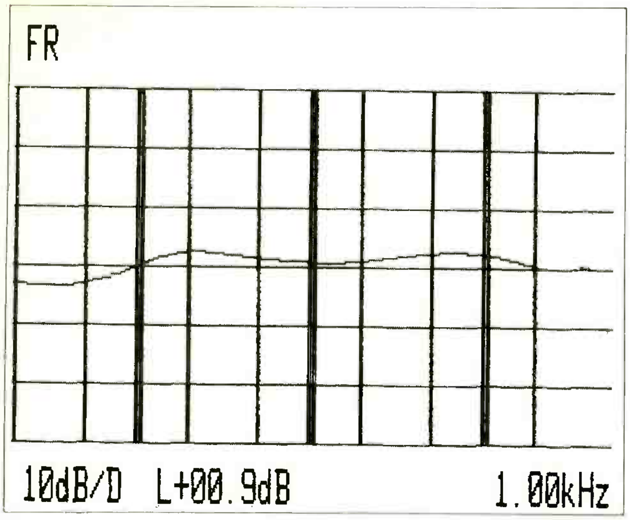


Fig. 3A

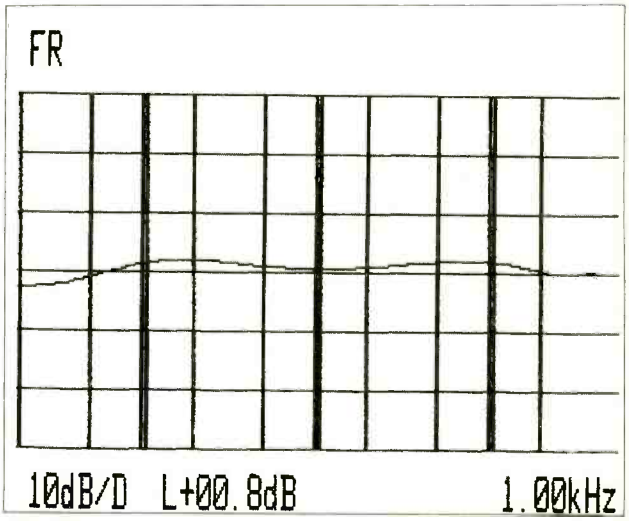


Fig. 3B

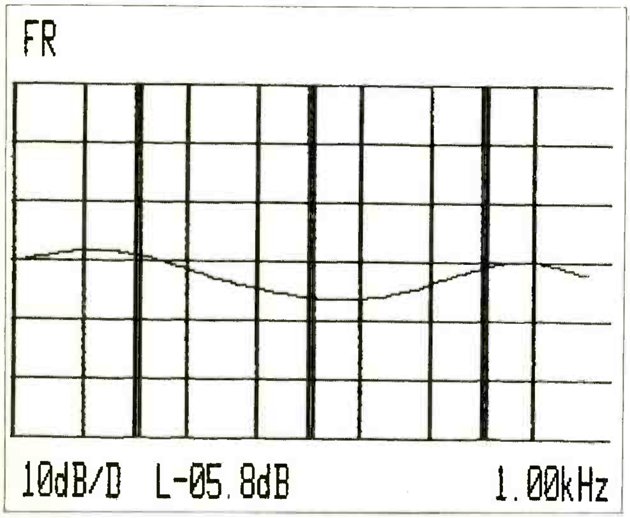


Fig. 3C

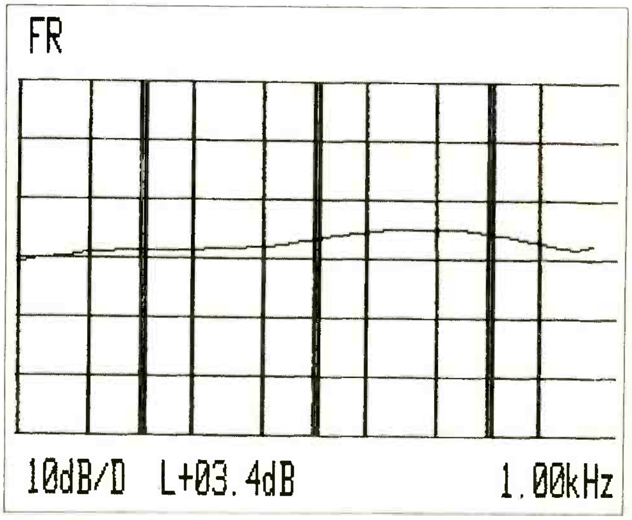


Fig. 3D

Fig. 3: Sansui SE-9: In addition to its analysis and adjustment modes, the SE-9 has four "built-in" selectable curves for (A) rock and pop music; (B) classical; (C) disco; (D) playback over car stereo or portable cassette systems.

dubbing; its hookup options for handling two decks at once with two-way dubbing if desired; its ability to store four curves; its automation and self-adjusting feature which proved in our tests to be as fascinating to watch as it was accurate in achieving proper system response in different rooms.

To be sure, one can buy a separate equalizer with more EQ bands and special low and high pass filters. One also can buy a more elaborate and sophisticated real-time analyzer. But even with the compromises inevitably found in a combination device, the SE-9 stands as a worthy instrument that can serve as an "audio tool" for the semi-pro or even fully professional sound technician, as well as a useful adjunct for the home audiophile and sound hobbyist.

Myself, I could do without the "Sound Menu" feature—most popular music is recorded with less than "flat" response to begin with and, in any case, I prefer to introduce my own preferred curves when dubbing for playback over my car stereo (which has speakers up front as well as in the rear). Of course, you can ignore the "Sound Menu" and program your own EQ curves into the SE-9's memory, but note that should you unplug the AC power cord, your own curves are gone and the "Sound Menu" returns when you reconnect and turn on the unit. I wish Sansui had provided the option for retaining one's own curves, although—at the speed with which a new EQ curve can be derived—this is hardly a major criticism.

The owner's manual, by the way, is complete, clear and amply illustrated. It contains far fewer linguistic awkwardness than many others we recently have seen.

Individual Comment by L.F.: Readers of *MR&M* may recall our report (June 1981) on the dbx equalizer/analyzer that could spectrum-analyze total system response right up to your listening position in the room and then completely electronically adjust overall response via its 10-band equalizer for "flat" response at that listening spot. Even more remarkable, the dbx unit could then store the resulting curve plus up to nine other curves for other listening locations or rooms, and have any of them, or their average, recalled from memory at the touch of a button or two.

If you remember that electronic marvel you will also recall that it cost $1500. If someone had told me then that most (but not all, to be sure) of those miraculous things would be handled by a device costing less than half that amount, I would have said it was an idle dream. But such are the technological advances in our industry that Sansui has, in fact, come up with what has to be the biggest equalizer/analyzer bargain to date.

Okay, so the unit has only eight bands instead of the ten provided on the dbx "rich man's" model. But, on the other hand, it features separate controls for left and right channels, whereas the dbx unit had controls common to both channels. Many equalization enthusiasts argue that it is better to be able to adjust the curves for the two channels separately. If you agree, then the Sansui SE-9 actually has something that the costlier dbx unit lacks. And okay, so the dbx unit had ten curve-memory capability while the Sansui unit can store only four curves for recall. However, in all but a few instances I would think that four curves are plenty, especially since if you need a curve not previously derived and stored in the SE-9, you can get a new one via analysis and adjustment in about 30 seconds time.

I could not help but be fascinated (and you will too) as I watched the SE-9 "doing its thing." There, before my eyes, were those sixteen slide controls moving one at a time, from left to rightmost control, after the system had "listened" to left and right channel speakers pumping out a half-second or so of pink noise each, and then decided how much each slider needed to be moved to achieve flat overall system response.

In one sense I feel that Sansui may have gone a bit overboard with the "Sound Menu" feature. I suppose some users may find those four preset curves useful, but most users—I believe—will probably prefer to create EQ curves more individually suited to their own tastes and listening environments.

Since Sansui always has prided itself on being primarily a "hi-fi" company, it is no surprise to find that some of the basic specs of the SE-9 are as excellent as one would expect to find in a top-of-the-line preamp or basic amplifier. Distortion and residual hum, for example, contributed by the SE-9 are likely to be lower than that contributed by the major elements of the sound system itself. I also liked the fact that the controls and switches permit the user to pre-equalize signals before they emerge from the record-out jacks. This particular feature makes it practical to pre-equalize your taped cassettes for their intended application.

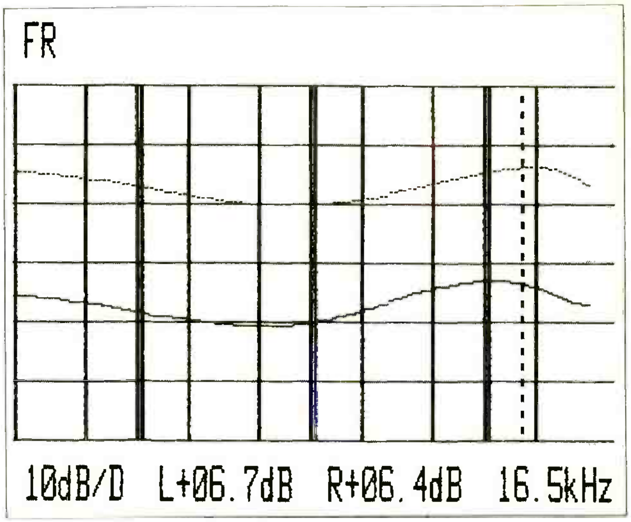


Fig. 4: Sansui SE-9: After analysis with the SE-9, Mr. Feldman found that his favorite listening position in his studio required left and right channel "corrections" shown above, for true, overall flat system response.

Incidentally, the two curves shown in *Fig. 4* are what the SE-9 indicated I should apply to my own system to get flat response. Note that they call for a boost of over 6 dB (maximum) at around 16 kHz. Considering the fact that for the sensation of a concert hall one doesn't want a flat high-end response (but rather a gentle rolloff above about 2 kHz to provide the so-called "house curve" that is suggestive of a concert hall environment), and bearing in mind too that my own listening room is only 14 feet by about 20 feet (and therefore cannot faithfully sustain ultra-low bass tones without standing-wave effects—hence the need for moderate bass boost where the microphone happened to be positioned), I am delighted to learn that my own reference system is as good as it is. For that alone I am grateful to Sansui for having come up with such an elegant instrument at such low cost. More seriously, I suspect that many audio enthusiasts and professional sound people will "voice" their systems now that there is a one-piece equalizer/analyzer that can do the job effectively at a reasonable cost.

Quite possibly, Sansui opted for an 8-band unit instead of an octave-by-octave 10-band system because the microprocessor needed for those four additional bands (two per channel) might have pushed the price beyond Sansui's targeted level. Or, perhaps, since the adjustment process takes place by actual mechanical motion of each successive band level, they may have felt that people would not want to wait for the adjustment sequence to be completed. If that was the reason, let me assure Sansui that I'd be content to watch those levers move up and down even if it took twice as long as it does now. In fact, I can't tear myself away from this fascinating product. Kudos to the designers!

SANSUI SE-9 EQUALIZER/ANALYZER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPECIFICATION	LAB MEASUREMENT
Nominal in/out level	1.0 volt	NA
Maximum output level (for 0.5% THD)	5.0 volts	6.3 volts
Input impedance, mic; high level	47 K; 30 K ohms	confirmed
THD, 20 Hz to 20 kHz	less than 0.008%	0.004% at 1 kHz 0.006 at 20 Hz 0.0065% at 20 kHz
Frequency response, controls flat	−1 dB, 10 Hz to 100 kHz	confirmed
Equalizer center frequencies	80 Hz, 160 Hz, 315 Hz, 630 Hz, 1.25 kHz, 2.5 kHz, 5 kHz, 10 kHz	confirmed
Boost and cut control range	±12 dB	+12.9; −12.4 dB (Typ.)
Signal-to-noise ratio, "A" wtd	110 dB	100 dB re: 1 volt
Gain, controls flat	0 dB	confirmed
Power consumption	30 watts	20 watts

Technics RS-M280 Cassette Recorder

General Description: The Technics RS-M280 is a stereo cassette deck whose transport uses three motors, the one for capstan drive being a quartz-locked direct drive type. A closed-loop, double-capstan arrangement is employed, as is a three-head (separate erase, record and play) configuration. The light-touch transport buttons operate through a microprocessor logic-control system. The machine's tape selector is set automatically for bias and equalizer via the tape-type detection holes found along the rear edge of a cassette housing. In addition, however, there is ample provision with the RS-M280 for making these adjustments manually, and deliberately varying them for desired effects. For this option, the deck incorporates a two-tone signal generator (400 Hz and 8 kHz), and the requisite adjustments up front. The signal meters on the deck are

The neat styling of the front panel is due in great part to the hinged cover under the meters. Behind this cover are found the so-called secondary controls for the recorder. These include the record calibration adjustments for each channel; the tape/source monitor switch; the Dolby NR switch; a multiplex filter switch; the tape selector; an input selector (mic, line and the test signal positions); the bias fine-adjust; output level control (handles line and headphones); left and right mic input jacks; dual-concentric input level knobs for left and right channels. The tape selector, when left in the automatic position, permits the deck to detect the kind of tape in a given cassette, assuming that the cassette does have the requisite coded holes. The tape types thus selected are normal, CrO_2 and metal. In manual mode, the tape selector offers the option for FeCr or metal. A sectioned panel above this area contains indicator lights showing various operational characteristics for tape type, monitor, Dolby and mpx filter. To its right is the metering panel which contains a digital tape counter and the memory indicator in addition to the fluorescent signal level markings. Peak levels are shown for recording and for playback, with a two-second hold option. The meters also are used for calibrating recording sensitivity and for bias fine-adjustments.

To the left of the display area are the transport buttons for the functions of record, pause, record muting, rewind, play, fast-forward, stop and counter reset. To their left is the cassette compartment which has a remaining-tape light and a head illuminating lamp. Finally, to the left of this section are the cassette eject button; the unit's AC power off/on switch; a switch for use with an external timer for unattended record or

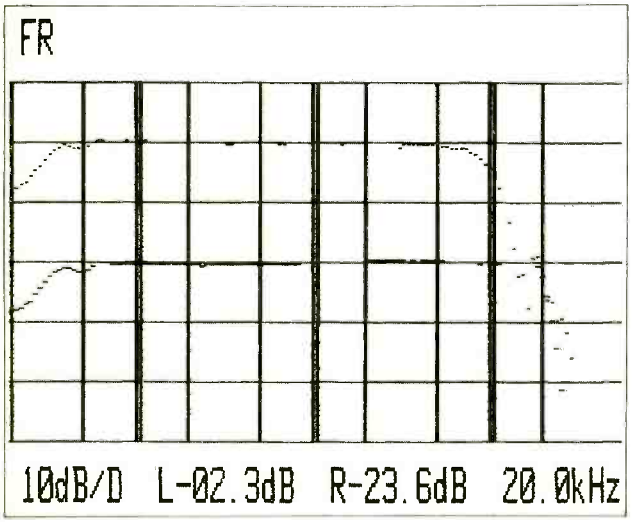


Fig. 1: Technics RS-M280: Frequency response at 0 dB and −20 dB record level (using Maxell XL-IS tape).

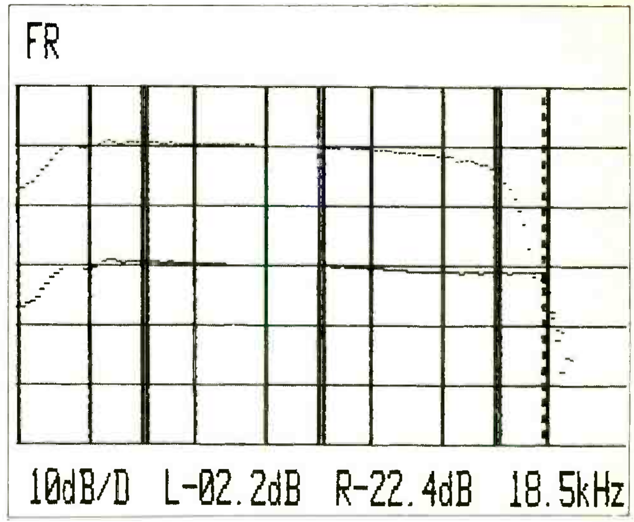


Fig. 3: Technics RS-M280: Frequency response (using Technics metal tape).

playback; the memory switch; and the headphone output jack.

The rear of the deck contains line input and output jacks; a socket for connecting an optional remote control unit; and the machine's AC line cord.

Test Results: Tests of the Technics RS-M280 were run with three types of tape: normal bias (we used Maxell XL-1S); high-bias or chrome equivalent (Technics XA); and metal (Technics Metal). While the RS-M280 can also handle FeCr tape, there are so few manufacturers offering this double-layer formulation these days, that we felt the evaluations with the other three types would serve to tell us (and you) everything about the deck worth knowing.

Figures 1, 2 and *3* show frequency response plots for the three tape types tested. Each plot shows response at a 0 dB record level (upper trace), and at the normal −20 dB record level. We should point out that Technics calibrates its record-level meters so that "0 dB" corresponds to around 165 nWb/meter, rather than to the more common 200 nWb/meter which also corresponds to Dolby calibration level. Referring to the curves and to the −3 dB points listed in our "Vital Statistics" table, it is interesting to note that the Maxell sample—using the so-called normal bias formulation—actually did a little better in frequency response than either of the other two tape samples, both of which bore the Technics label. We used these samples simply because they were supplied with the deck, and we assumed that the deck would work best with them. The results were hardly "bad"—in both instances while the low end fell a trifle short of spec'd

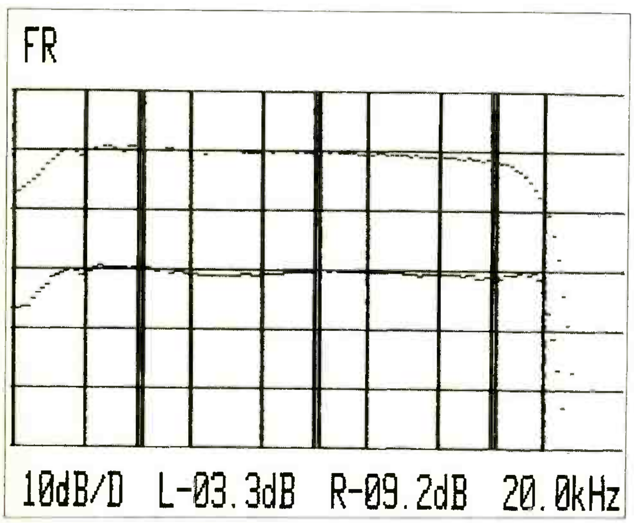


Fig. 2: Technics RS-M280: Frequency response (Technics XA high-bias tape).

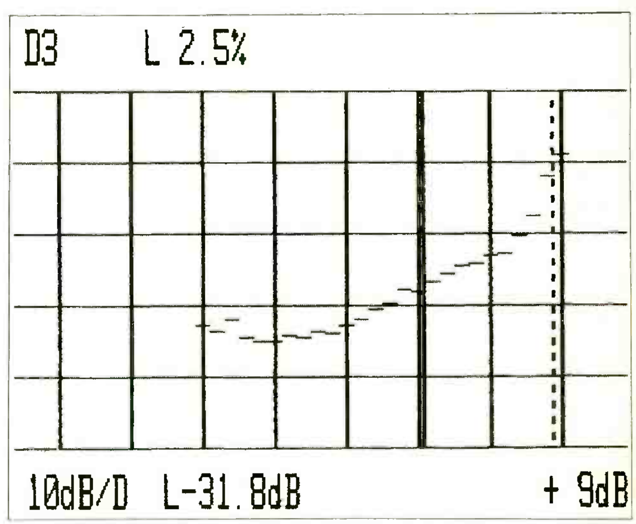


Fig. 4: Technics RS-M280: Third-order distortion vs. record level (Maxell XL-IIS tape).

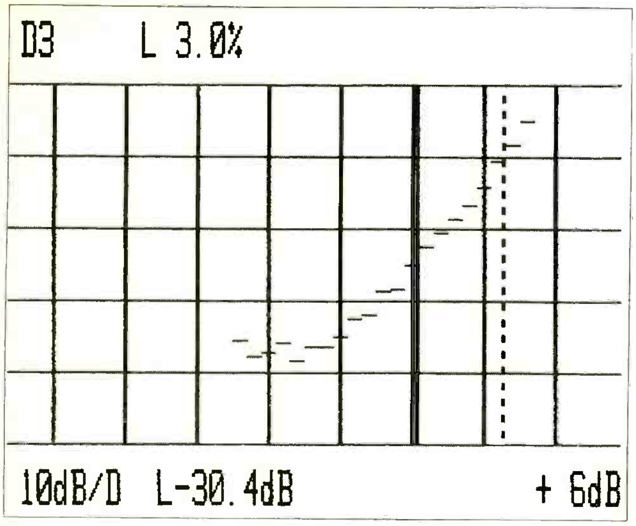


Fig. 5: Technics RS-M280: Third-order distortion vs. record level (Technics hi-bias tape).

response, the high end exceeded it slightly.

Figures 4, 5 and *6* are plots of third-order harmonic distortion versus recording level. In each case we have set the dotted-line cut cursor to read that recording level (relative to the machine's own "0 dB" reference) at which approximately 3% third-order distortion was observed. Again, the Maxell sample (normal bias) had the highest relative headroom (+9.5 dB) of all three tape samples.

Figures 7, 8 and *9* show overall signal-to-noise ratios obtained for the three tapes, relative to the 3% third-order distortion levels previously measured. In each of these displays, the "bar graphs" at the right correspond to the single numerical reading above each graph, while the spectrum analysis at the left shows the third-octave distribution of the residual tape noise. In these displays, the "A" diagrams were produced with Dolby off; the "B" diagrams with Dolby on. The improvement introduced by Dolby in the high frequencies is quite evident when any pair of displays is compared side by side. All measurements of S/N were made using the CCIR/ARM weighting which is becoming more and more popular for this kind of test.

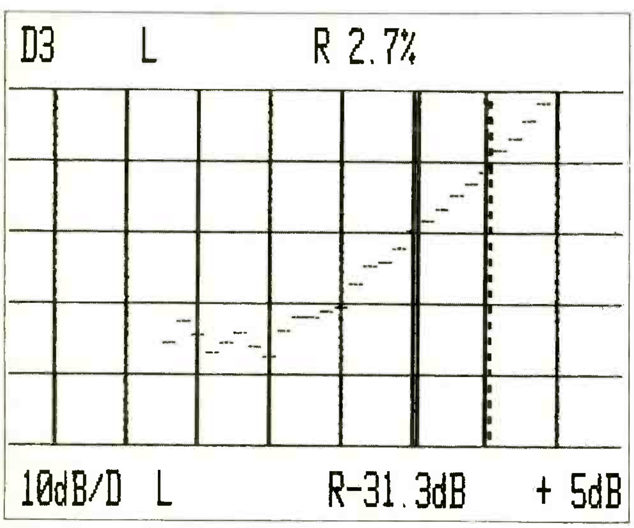
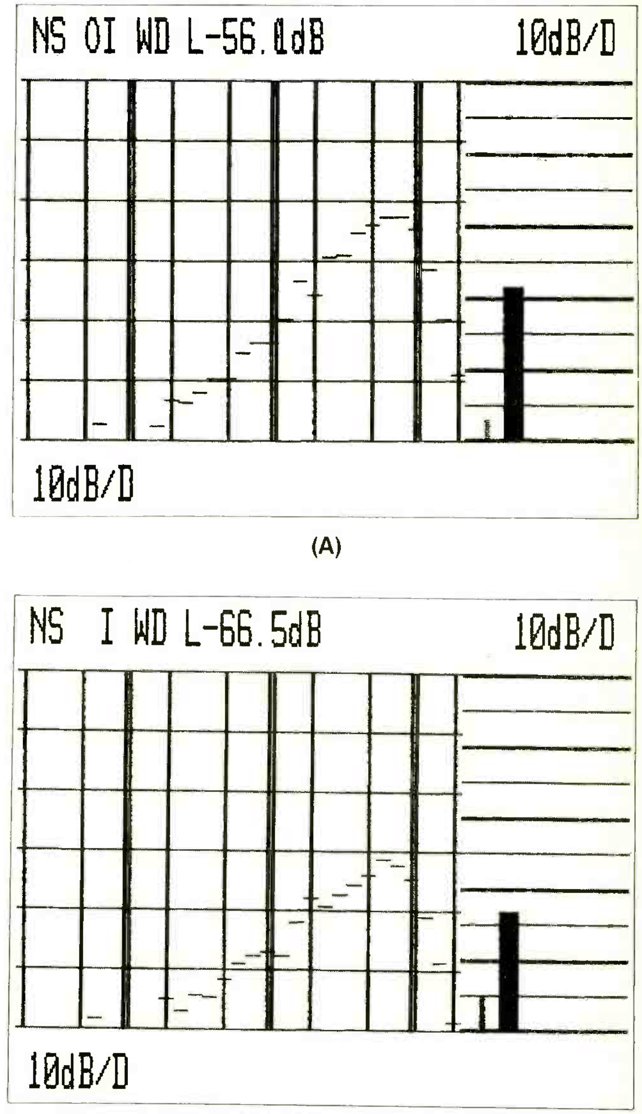


Fig. 6: Technics RS-M280: Third-order distortion vs. record level (Technics metal tape).



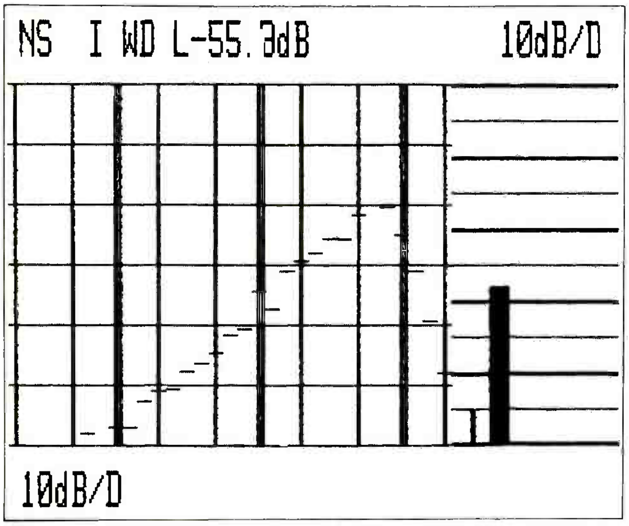
(A)

(B)

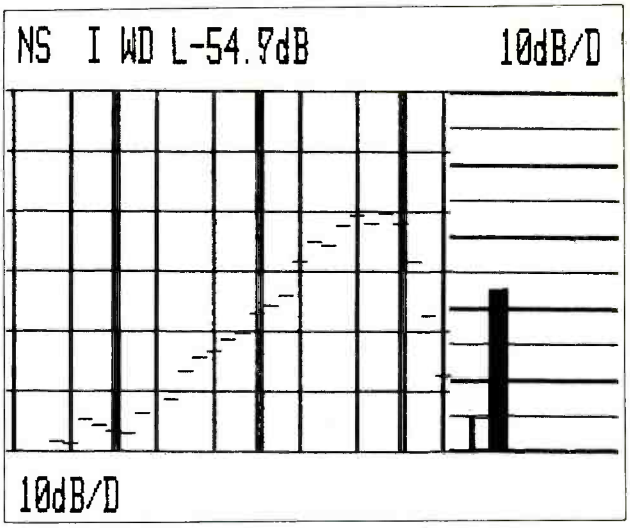
Fig. 7: Technics RS-M280: Weighted (CCIR/ARM) S/N, using Maxell XL-IS tape; (A) without Dolby, (B) with Dolby B.

The wow-and-flutter reading *(Fig. 10)* is, as best we can recall, the lowest we have ever obtained for any cassette deck we have tested. Also commendable was the deck's speed accuracy, being a mere 0.02 percent within nominal speed.

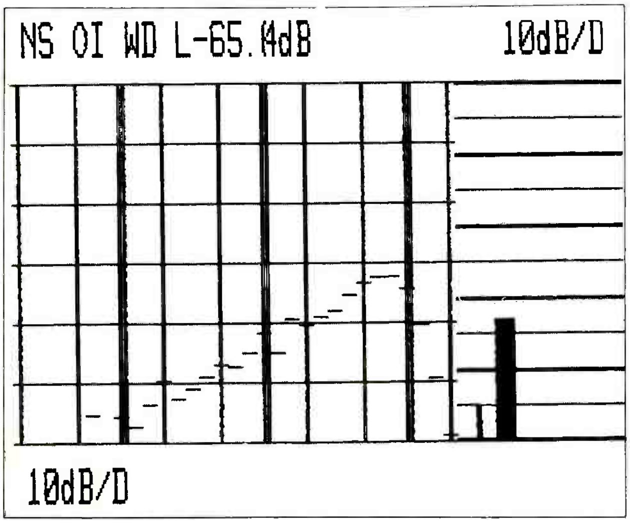
General Info: Dimensions are 16 7/8 inches wide; 3 7/8 inches high; 13 3/8 inches deep. Weight is 13 pounds, 15 ounces. Price: $800.



(A)

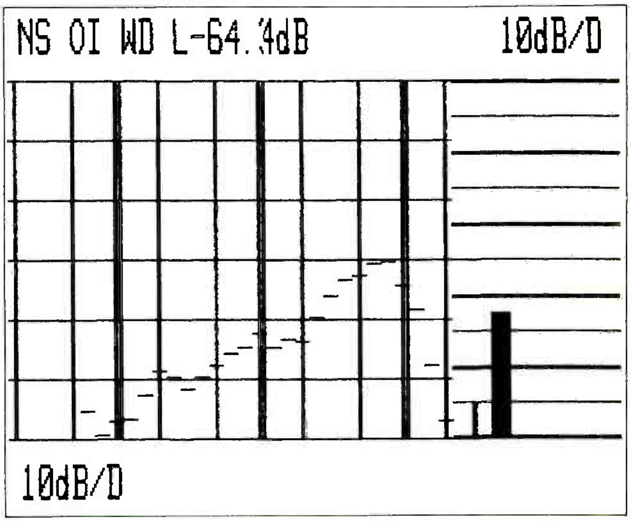


(A)



(B)

Fig. 8: Technics RS-M280: Weighted S/N, using Technics XA high-bias tape, (A) without Dolby, (B) with Dolby B.



(B)

Fig. 9: Technics RS-M280: Weighted S/N, using Technics metal tape. (A) without Dolby. (B) with Dolby B.

Individual Comment by L.F.: In recent months we have tested and written about several high-technology microprocessor equipped cassette decks. These are able to set their own bias levels, sensitivity (to properly track Dolby noise-reduction systems), and—in some instances—record equalization characteristics to suit any tape made. Invariably we have also noted with respect to these machines that they tend to favor "flat" frequency response, even if that means sacrificing some S/N capability, or encountering higher-than-necessary distortion levels. The pro audio recordist, we have always felt, may object to that kind of "locked-in" situation, and would rather prefer to give up some of the microprocessor automation and self-adjustment features if he or she could only tailor a recorder's parameters to suit a given tape and to suit the recordist's own preferences.

The Technics RS-M280 does offer just that kind of flexibility and adjustment capability. Of course, it is one thing to provide a vernier bias adjust, and separate left and right channel record sensitivity controls. It is, however, quite another thing to provide the user with a self-contained means of correctly using those adjustments for a given recording. The visual adjustment method used in the RS-M280 is foolproof. With just a little practice, I was able to come up with ruler-flat response for just about any tape. Better than that, however, if I preferred to overbias slightly (to reduce distortion still further at mid and low frequencies), or to under-bias slightly (so as to extend high-end response and high-frequency headroom slightly), I could do so.

Learning to use the test tones and the meters for setting up any type of tape to your own taste may take a little while, but I feel it is well worth the effort. There is an analogy here, I think, to tastes in automobile dashboards. Some of us prefer those simple red lights (often call "idiot lights") and automatic transmissions. Others of us would rather have oil-pressure gauges, coolant thermometers and a stick shift. Of course, both camps want a smooth, reliable ride. It's all a matter of how you get from "here to there."

Individual Comment by N.E.: Depending on one's viewpoint, it would seem that the Technics RS-M280 can be regarded as a cassette recorder with a "split personality" or as one that is intended to appeal to both the home hi-fi enthusiast and to the more advanced recordist. At that, there is much about the unit that validates both claims for attention. It has the basic requisites for good, reliable transport action with a three-motor, closed loop, double capstan drive system, under microprocessor logic control. Tape motion is flawless, and that 0.02 percent wow-and-flutter figure is remarkable indeed. The deck will adjust itself automatically for different tapes, but it also lets the user make his or her own adjustments, a useful option for the serious recordist. S/N figures seem in the ball park for Dolby-B; one can only wonder how the unit would have done with Dolby-C.

"Hiding" many of the adjustments behind the swing-down door on the front panel seems a cosmetic bow to the "home" market, while providing the adjustments should appeal to the more advanced recordist. It does lend the unit a less cluttered look, particularly with that indicator panel above it. The departures from some of the specs in our test results are, in sum, not terribly significant, although it would have been nicer to report that the unit met or exceeded specs. The machine does, in any case, appear to be a smooth performer with good sound and high reliability.

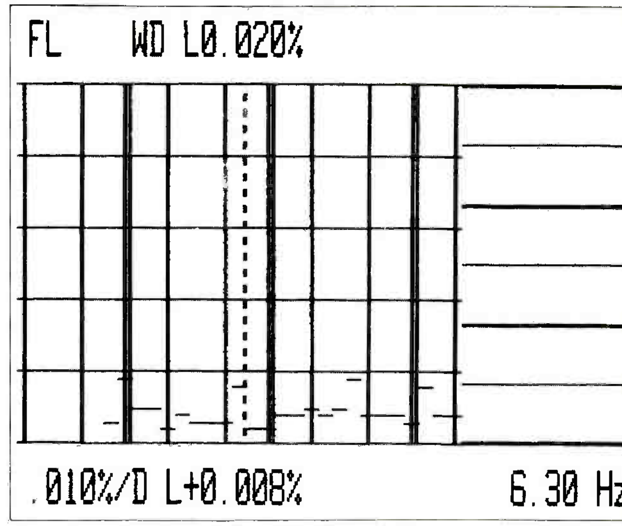


Fig. 10: Technics RS-M280: Spectrum analysis of wow-and-flutter components (from 0.5 Hz to 200 Hz) and overall W&F measurement (0.020% WRMS).

TECHNICS RS-M280 CASSETTE RECORDER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPECIFICATION	LAB MEASUREMENT
Frequency response, normal tape	±3 dB, 25 Hz to 17 kHz	±3 dB, 31 Hz to 20.3 kHz
high bias tape	±3 dB, 25 Hz to 18 kHz	±3 dB, 29 Hz to 19 kHz
metal tape	±3 dB, 25 Hz to 19 kHz	±3 dB, 29 Hz to 20 kHz
Signal-to-noise ratio, Dolby off, re: 3% THD record level normal; hi-bias, metal	NA; 60 dB; NA	56.5; 55.8, 54.7 dB
Signal-to-noise ratio, Dolby on, re: 3% THD record level normal; hi bias; metal	NA; 70; NA	66.5; 65.5; 64.4 dB
Record level for 3% THD (0 dB = 165 nWb/m) normal; hi bias, metal	NA; NA; NA	+9.5; +6.0; +5.5 dB
THD at 0 dB record level normal; hi bias; metal	NA; NA; NA	0.40; 0.57; 1.00%
Wow-and-flutter (WRMS)	0.024%	0.020%
Speed accuracy	NA	0.02%
Line output at 0 dB	700 mV	720 mV
Headphone output level at 0 dB	125 mV (8 ohms)	120 mV
Mic input sensitivity for 0 dB	0.25 mV	0.25 mV
Line input sensitivity for 0 dB	60 mV	67 mV
Fast-wind time, C-60	80 seconds	75 seconds
Bias frequency	85 kHz	85 kHz
Power consumption (watts)	25 watts	22 watts

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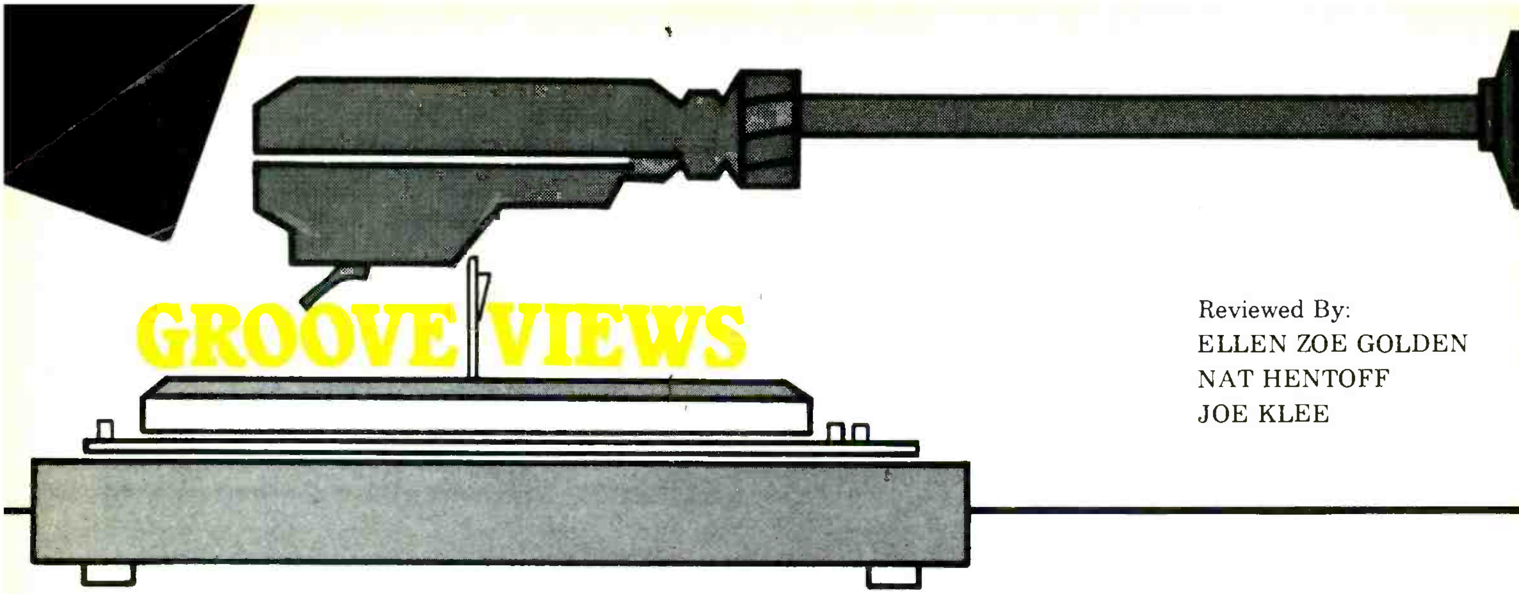

Reviewed By:
ELLEN ZOE GOLDEN
NAT HENTOFF
JOE KLEE

POPULAR

IAN DURY: *Lord Upminster.* [Chaz Jankel, Steven Stanley and Ian Dury, producers; Steven Stanley, engineer and mixer; Harold Dorsett, assistant engineer; recorded at Compass Point, Nassau, Bahamas, April-May 1981.] Polydor PD-1-6337.

Performance: **Grand**
Recording: **Clean as a whistle**

When one thinks of Ian Dury, the words pleasant and comforting don't usually come to mind, but Dury's latest LP, *Lord Upminster,* can be called that and more. With Chaz Jankel, co-writer of Dury's most popular and absurd material back on hand, and a backing band that includes *the* rhythm duo of Sly Dunbar and Robbie Shakespeare, Dury (without the Blockheads) has managed to deliver an album rich with beat and humor, but buffed with a fine coat of polish that actually works well to clean up his previously rowdy sound.

With the fine production mix of Steven Stanley, Jankel and Dury, it is easy to identify the jazzy guitar of "Red (Letter)," the tight-skinned bongo slaps of "The (Body Song)" and "Lonely (Town)," as well as the crazy tongue-twisted lyrics of "Funky Disco (Pops)." The latter is the epitomy of a proper disco song—solid, consistent beat underneath an insightful tale about a frustrated and still man in a world where everybody just wants to dance.

Dury takes frustration and glorifies it

IAN DURY: Glorifying frustration.

in the controversial "Spasticus (Autisticus)," banned on the BBC because of its apotheosis of the spastic human being. It's always been Dury's forte to take a malady or condition and make a dance hit out of it, but the cleaner sound of "Spasticus (Autisticus)" allows the tight rhythm ribbon of Dunbar and Shakespeare to shine through appropriately, while today's dance beat-maker—the synthesizer—only adds flavor-enhancing seasoning.

As before, Dury and Jankel's lyrics are witty, albeit intelligent, and the crazy rhymes of "Trust (Is A Must)," drag an ear to the speakers to try and out-guess the authors. Lines like "Hope helps you cope," "Friends are the trend," and "Time is sublime" are interspersed with a sound that resembles a twanging bent saw. What a clever combination! As are the hip-bone's-connected-to-the-thigh-bone elaborations of "The (Body Song)" that are mixed with a prominent bass guitar and a sensible disco beat.

Don't be mistaken, *Lord Upminster* is not a wild and wooly beat-crazy clash, but rather a soothing rhythmic masterpiece demonstrating a successful new side of the loveably cocky Dury. E.Z.G.

JAZZ

ANNE MARIE MOSS: *Don't You Know Me.* [Bernard Brightman, producer; Charles Leighton, engineer; recorded at J.A.C. Studios, New York, N.Y., Jan., 1981.] Stash ST 211.

Performance: **It's time for Anne Marie Moss**
Recording: **Wonderful**

Anne Marie Moss isn't new on the scene. She came down from Canada with Maynard Ferguson's Orchestra in 1960. She served a stint with Jon Hendricks and Dave Lambert as Lambert, Hendricks and Ross became Lambert, Hendricks and Moss temporarily. She's been singing around town as a single and as a duo with Jackie Paris for the past...(wow, has it really been that long). The few records previously available were either badly recorded, badly produced or badly exploited. It's just about time the kid got a break. Enter Stash Records, J.A.C. Studios and especially a fantastic arranger named Hale Rood and here's the record that's only something like twenty years overdue.

Well, anyway, it gave Anne Marie a chance to gather some great tunes like

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JAZZ DIVAS: CARMEN McRAE AND SARAH VAUGHAN

By Nat Hentoff

Carmen McRae is the very model of the disciplined professional, but there are times when there is so much control that her performances tend to be mechanical. Yet, when Carmen loosens up, she can be one of the most compelling jazz singers in the history of that rare art. One such occasion was a 1970 recording session in London with the formidable Kenny Clarke/Francy Boland Big Band. It has finally been released here as *November Girl* on the new Jazz Man label (First American Records, 73 Marion Street, Seattle, Washington 98104).

The primary reason this set contains such full-strength Carmen may come from the fact that six of the eight tunes were entirely new to her, as were all eight Francy Boland arrangements—and there was very little rehearsal time. Therefore, she couldn't plan every nuance of her performance. Forced into spontaneity, Carmen drew strength from the immediacy of her feelings and the result is what Spanish gypsies call "deep song."

Along with such standards as " 'Tis Autumn," Carmen probes such intriguing originals as Jimmy Woode's "Dear Death." Throughout, there is Carmen's masterful use of space (the power of momentary silence); her authoritative musicianship; and her sinuous beat. This time, all these qualities cohere with minimal calculation and maximum self-surprise.

The engineering is judiciously attentive to keeping Carmen and this juggernaut of a band in balance, and the sound is both crisp and warm.

After her bold improvisatory beginnings at the dawn of bop, Sarah Vaughan also went through a long period of being somewhat too self-conscious in burnishing her professionalism. In her case, the emphasis was on virtuosity and she could so ornament a tune as to muffle her soul in lace. During the past couple of years, at least on her Pablo albums, Sarah has become much less baroque; but it's still a delight to hear her when she was still creating herself anew almost every time she sang. As in the mid-1950s *Sarah Vaughan* session on EmArcy.

These EmArcy dates were directed by Bobby Shad and have long been unavailable, but Polygram is now releasing them on first-class Japanese pressings—as it is also doing with Norman Granz's Verve catalogue. On Sarah's session, she is backed by the floatingly lyrical Clifford Brown on trumpet, tenor saxophonist Paul Quinichette, flutist Herbie Mann (among his best on record) and a superbly integrated rhythm section of Roy Haynes, Jimmy Jones and Joe Benjamin. Sarah soars utterly relaxed, utterly swinging, her extraordinarily textured voice a continual astonishment.

Despite all kinds of technological advances in the years since, the sound here holds up so well—clarity, warmth, presence, balance, naturalness—that it's quite a tribute to whoever engineered the date. But no credit is given. It was probabaly recorded at the Fine Studios at the old Great Northern Hotel in New York.

CARMEN McRAE AND THE KENNY CLARKE/FRANCY BOLAND BIG BAND: *November Girl*. [Gigi Campi, producer; Adrian Kerridge, engineer.] Jazz Man JAZ 5004.

SARAH VAUGHAN: *Sarah Vaughan*. [Bobby Shad, producer; no engineering information.] EmArcy EXPR-1009.

Hoagy Carmichael's "Baltimore Oriole" and Jerome Kern's "I'm Old Fashioned" and to refine her art even further, polishing the perfection. There're some new tunes too. I'm particularly fond of "He's No Good For Me" but the gems of the set are the old standards. Anne Marie sings "I'm Old Fashioned" starting with the last eight bars and then goes back to the top backed by Jim McNeely's delightfully understated piano chords. Anne Marie sings the verse to George and Ira Gershwin's "How Long Has This Been Going On" but she gives the word "Dante's" the correct pronunciation (dahn-tays) thereby correcting Ira Gershwin's most unforgivable pun when he puts the word in to be sung (dan-tees) so as to rhyme with panties and aunties. Anne Marie opens Rodgers and Hart's "My Romance" going around with Eddie Gomez' walking bass and Eddie's never been in better company.

To be frank about it, except for my dislike of synthesizers and electric keyboards and one tune which was picked for the album that I think isn't as good as the others ("Corner Of The Sky") this is a perfect recording. It's already one that's frequently played around our house and I see no signs of weariness setting in. It's just too bad it took so long to arrive. Well, the next one ought to come a lot sooner, so things are looking up. With a talent like Anne Marie's, they should be. J.K.

MONGO SANTAMARIA, with DIZZY GILLESPIE AND TOOTS THIELEMANS: ***Summertime.*** [No producer given; David Richards, engineer; recorded at the Montreaux Festival July 19, 1980, by Mountain Studios, Montreaux, Switzerland.] Pablo Live 2308229.

Performance: **Exciting, invigorating, thrilling**
Recording: **The sleeve says digital, but the sound says terrible**

It is of particular interest to me that Norman Granz is not listed as producer on this album. It may be a printer's error that his name was omitted. I would prefer to think that when he heard the horrible balance and awful fidelity of this recording he threw up his hands and said "there's no way I'll let this come out under my name," and I don't blame him. The whole idea of the Pablo Live series is to preserve important and monumental performances that went down on stage even though this may not happen with the greatest stereo high fidelity digital record equipment on hand. Certainly the great music that Dizzy Gillespie and Toots Thielemans and Mongo Santamaria's house band made that evening justify putting out this record but be aware that the engineering is not up to snuff.

Mongo's band plays the first chart without Diz and Toots and a fine band. It's Allen Hoist whose baritone sax is worth straining your ears to hear. Pianist Milton Hamilton also makes some fine statements in a style that reminds me of the early recording of mambo king Perez Prado who, it was once observed by a critic, played piano like a boxer. Unfortunately, the sound balance seems most favorable to bassist Lee Smith whose specialty seems to be ostinato phrases repeated ad nauseum. Still when Dizzy takes over during John Coletrane's piece "Afro Blue," the Alps come alive with the sound of good jazz.

Dizzy was an early champion of the Latin jazz fusion. When he came to New York he played early on with Alberto Socarras who had a jazz/Latin band and when Dizzy formed his own big band the Afro-Cuban influence was heavy in the band, with the famous Chano Pozo on bongo drums and charts by George Russell and by Chico O'Farrill. Theilemans came up a bit later scoring his early success with George Shearing's group, another group that relied heavily on Latin American influences. Yet the ultimate jazz/Latin fusion came when Norman Granz got the idea of pairing the best of the boppers, Charlie Parker, with the best of the Latin bands (with jazz influences), Machito. It was those records with Bird and Machito that set the tone for what today is such a complete fusion that it's not even thought of in terms of fusion any more. The common ground in which jazz and Latin music meet is today taken for granted. In the days when Bird and Machito did their records together it was a novelty. Dizzy, along with George Shearing and Stan Kenton, had a great deal to do with the acceptance of Latin jazz and it is good to hear him in this context again. It is a context which seems to induce him to play up to (or maybe even better than) the best of his current level of playing. I hope that Diz and Mongo get together again, only this time under more favorable conditions as far as recording is concerned. J.K.

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Ms. Ciani, principal of Ciani Musica, Inc., a New York-based production company, is soon to release her first original electronic album, "New Waves" on the Victor Musical Industries label. She holds a Master's Degree in music composition and has been at the forefront of electronic music for nearly fifteen years.

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