VOL. 2 NO. 8
AUGUST 1977

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MODERN RECORDING

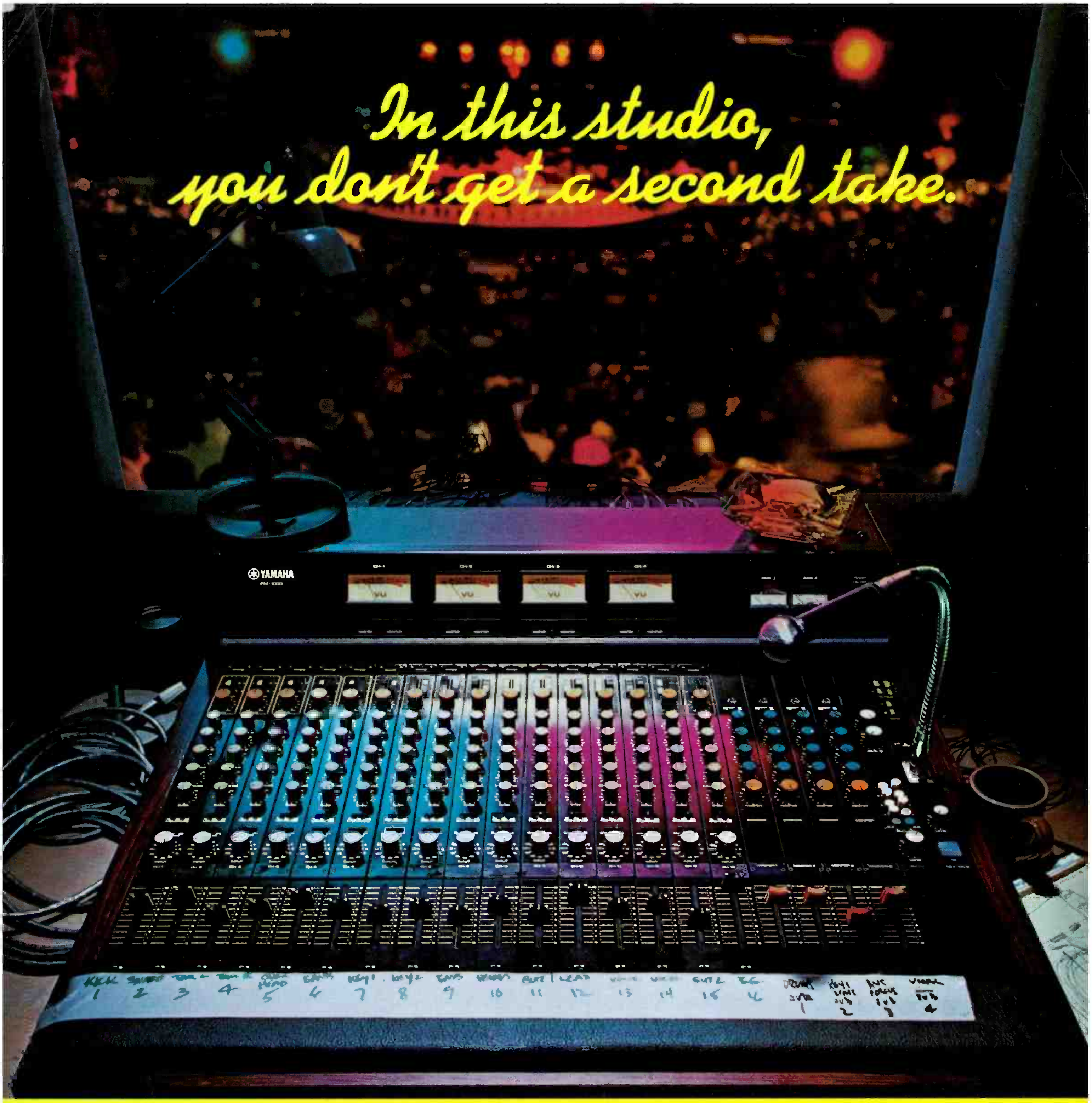
SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

A Session with the Bee Gees

What do Chick Corea, Alice Cooper, John Lennon and Peter, Paul and Mary have in common?

From Tape to Disc—Disc Mastering, Part I

Lab Reports • New Products • Record Reviews



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Monaural power amplifier
- 200 Watts rms
- 20 Hz to 50 kHz response
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- Less than 0.2% IMD
- LED overload indicator
- 19-inch rack mount
- Forced air cooling

CS-400 $424.50 *

Stereo power amplifier
- 200 Watts rms per channel
- 20 Hz to 50 kHz response
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AUGUST 1977
VOL. 2 NO. 8

MODERN RECORDING

SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

THE FEATURES

COMING NEXT ISSUE!
DISC MASTERING, Part II

THE STAPLES

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Cover photo by Bill Deistchle

MODERN RECORDING
SERVING TODAY'S MUSIC/RECORDING-CONSCIOUS SOCIETY

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Letters to the Editor

Some Fancy Figures

The article by Robert Angus on "Equipment: What It Should Cost To Fix" (April, 1977) attempts to tell it like is. However, I believe I have a more meaningful solution for the problem.

Thanks to my trusty little H-P pocket calculator, I now do a statistical analysis of all service charges twice a year. The results are posted as follows:

Statistical Analysis of All Charges
For the Most Recent 6-Month Period

*Avg. Total Charge:	$21.62	†Standard Deviation: $10.34
Avg. Labor Charge:	$15.29	Standard Deviation: $ 6.24
Avg. Parts Charge:	$ 6.33	Standard Deviation: $ 4.10

*Avg.: Arithmetic Mean. †Standard Deviation: 50% are higher or lower than this amount. You will be informed if your cost exceeds the Avg. plus Deviation.

I've found this procedure most reassuring for our customers and we seldom get a request for an advance estimate.

I should also mention that we limit our service to musical instrument amps and sound reinforcement gear. However, I'm sure this procedure is worth the effort for any service business.

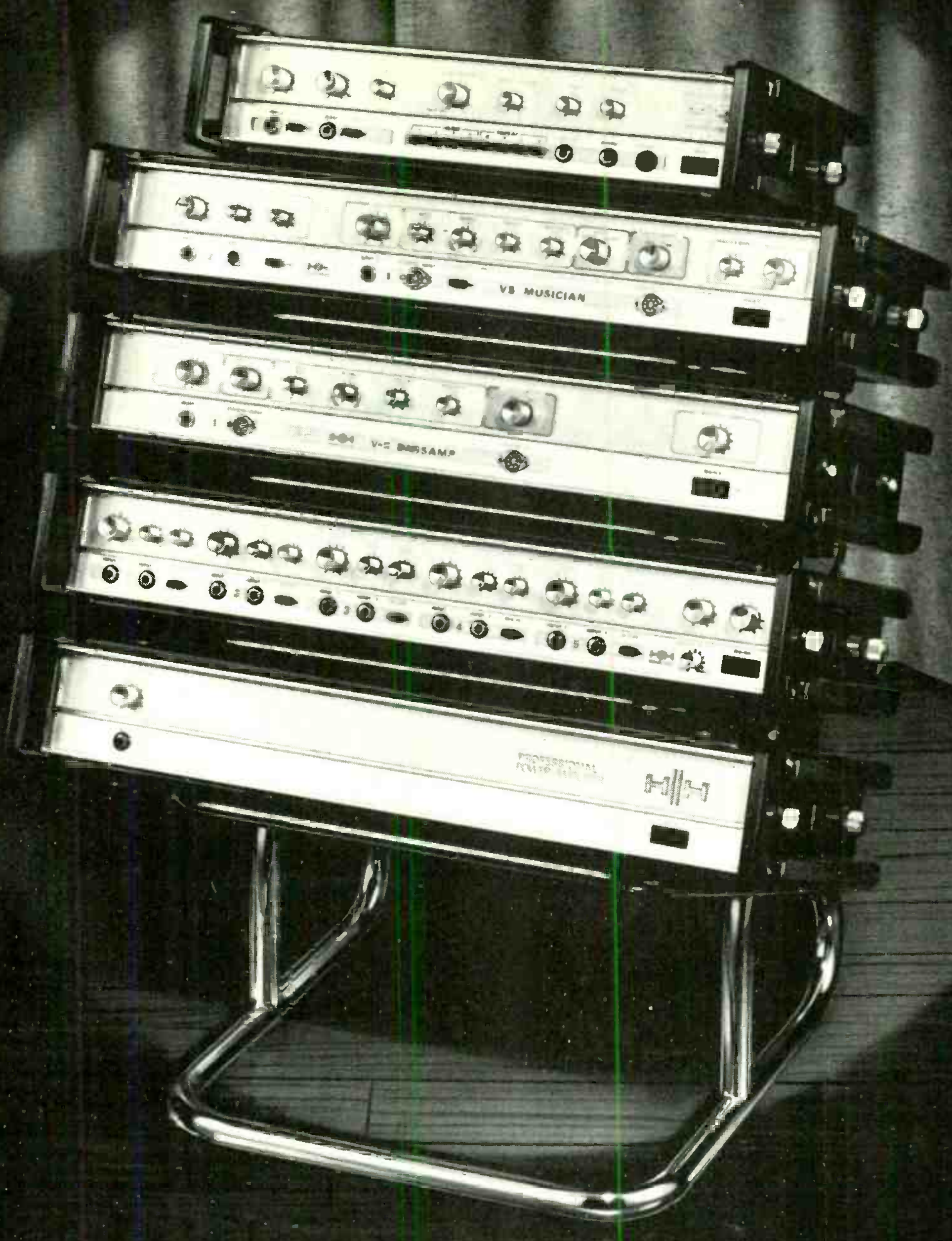
—Ed Griese
President
InterFax
Milwaukee, Wi.

Finding Used Equipment

I have a problem that perhaps you could help me with. I am trying to locate an inexpensive reverb unit manufactured by Gately Electronics, Inc. (model EK-6, sold as a kit or wired; the kit version was priced at $195.00). I have tried unsuccessfully to contact the company by phone and by mail, but have been unable to find out for sure whether they have gone out of business or what has become of them. At any rate, the unit is listed in your current *Modern Recording Buyer's Guide*. Is it still available? If so, where? Anywhere in the Washington D.C. or Richmond area? If I can't obtain a new one, maybe another reader has a used one for sale. Any help would be greatly appreciated as this seems to be the only reverb device made in this price range with a built-in sound mixer. Thank you.

—Ted Kallman
Charlottesville, Va.

After a myriad of telephone calls, we finally found someone who could tell us exactly what the story was. Al Grundy of the Institute of Audio Research informed us that, while Gately Electronics has been dissolved, used equipment can be found in either of two places. You can write for a catalogue of used pieces (including the Gately reverb you need) to: Boynton Studio, Inc., Melody Pines Farm, Morris, N.Y. 13808 (Tel. 607-263-5695) or Equipment Locator, P.O. Box 99569, San Francisco, Ca. 94109 (Tel. 415-397-4623).


VS MUSICIAN
V-S BASSAMP
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Praise for MR

I really enjoy your magazine. It is informative and interesting. My employees read it like a textbook to (hopefully!) learn new things and refresh themselves on techniques and new products on the market.

We look forward to our monthly issues. Your hard work pays off! Rock and Roll!

—David Lowe
Brandywine Sound Company
Downingtown, Pa.

Reader Offers Another Solution

Congratulations on a well-produced, highly useful magazine. We here at Hutch Productions in Houston especially appreciate your precise, well-aimed equipment reviews—they save us a lot of "window shopping."

In regard to your recent and continuing series of readers letters pertaining to the use of ¼-inch four-channel tape machines such as the TEAC 3340 as a master recorder in the production of stereo or mono mixdown recordings, we feel that our method of accomplishing this may be of some value to your readers. We begin by recording four basic tracks on the 3340, which are then mixed down to two tracks on a ¼-track two channel recorder. This mixdown is then rewound onto the 3340, opening up two new tracks without having to "bounce back" to the multitrack recorder as is necessary with wider format machines. This eliminates one generation of tape per mixdown, and we have found that this process can be repeated through three or even four mixdowns with excellent final results.

While this sounds simple on paper, it is somewhat more complex (as you might guess) in real life. Noise reduction units on both machines are a *must* (we use Dolby B units), as is extremely precise electrical and mechanical alignment of both machines. For best results, we run both machines at 15 ips and align them to IEC playback curves. Final mixes are made on a ½-track stereo machine with NAB equalization for mastering to disc (we use a Pioneer RT-1050). Our ¼-track mixdown recorder is a TEAC 2300S which has been modified to run at 15 ips (primarily by changing the capstan to the one used in the TEAC 3340S).

As an example of the final results that can be achieved using this method, we offer our production of an album of progressive rock music by the group Lionhart. It has been receiving rave reviews and heavy airplay from radio stations in the Louisiana/East Texas region,

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and is available locally in that area or by mail directly from us for $6.98 postpaid, plus 5% state sales tax for Texas residents. (Hutch Productions, P.O. Box 10326, Houston, Tx. 77206.) We are including two copies of Lionhart's album for your critical review.

Again, more power to you good folks at *Modern Recording*—your magazine is something many of us have needed for a long, LONG time!

—Craig Bransfield
Hutch Productions
Houston, Tx.

Thank you for your letter and the Lionhart albums. We will see if we can get our reviewers to give you their opinion on your creative recording.

In Response to a Response

The following is the author's response to a letter received from JBL which appeared in the Letters section—"JBL Disagrees"—May, 1977. It was a comment on Mr. Lewis' article, "Monitors for the Recordist," (MR Feb/Mar 1977).

My "positive assessment of narrow-band, peaked-response monitors" was in no way intended to be an endorsement. It was simply an acknowledgement of two facts of life. Namely, that there are a lot of monitors out there with these characteristics, and that a lot of good records have been made with them (albeit with a certain amount of "guesstimated" response corrections). This may in fact be the whole point.

Furthermore, it is hard to see how an ultra-wideband monitor could help much with such problems as cutter overload and turntable rumbles (to which I would add record warp) when in fact all these degradations lie completely outside the "loop" in which the recording engineer works.

Finally, on the question of peaked midrange, examination of JBL's own published response curves for several of its monitor systems reveals precisely what my article claimed: a slight upper-midrange peak!

—Rob Lewis
Los Angeles, Ca.

Eager Reader

I am amazed at all I have learned from your magazine since I became a steady subscriber. Now, I am eager to learn all that I might have missed before I discovered *Modern Recording*. Are back issues available? Also, what can you say about TEAC's Tascam Model 5's and 80-8? Any info you have on these would be appreciated.

—Jack Williams
Rock Tavern, N.Y.

Information about the TEAC Tascam Model 80-8 can be found in the Talkback section of the Dec/Jan 1977 issue of MR, *on page 14. Info about the Model 5 can be found in the July 1977 issue, where it's the subject of the "Hands On" report. Currently, the back issues that are available include Dec/Jan 1977, Feb/Mar 1977, April 1977, May 1977, June 1977 and July 1977. To receive back issues simply write to our Subscription Department and request them at $1.75 per copy.*

No Info Yet

What is the Aphex Aural Exciter? I believe it is used at the Sound Factory in Los Angeles.

—Charles Fargrehan
Bill Case Sound
San Antonio, Tx.

We have tried repeatedly to get an answer to your question. However, our sources inform us that Aphex is hesitant to release any premature or incomplete information about this unconventional device, which was ten years in the making. They feel, since the ideas and theories that the Exciter embodies are so new, any partial information would lead to misconceptions. While the piece has already been used extensively by professionals, Aphex prefers to wait until September to launch a full, public disclosure. MR *plans on having more on this revolutionary piece of equipment, so you can expect a full explanation soon.*

We Answer a Question

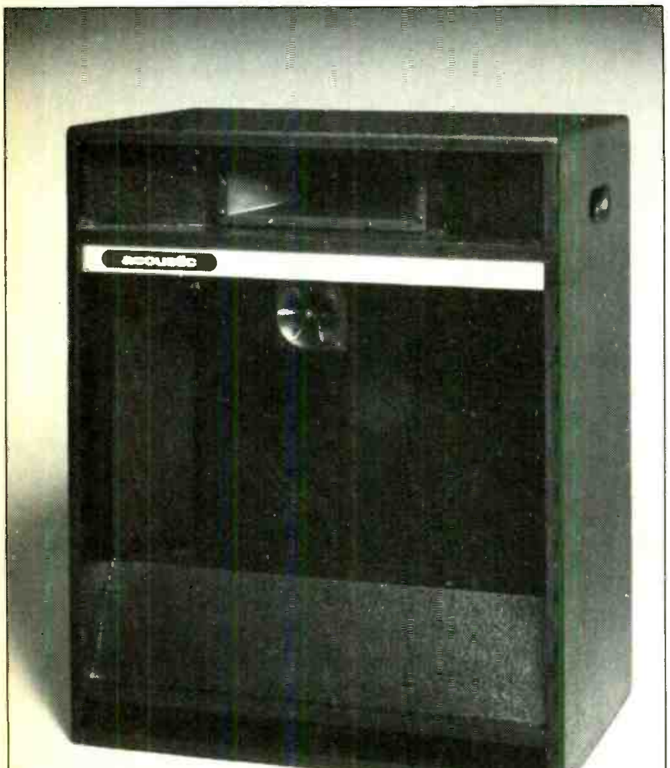
Just thought it was about time that I dropped you a line to say how much I enjoy your fine magazine. It definitely fills a need in the lives of audiophiles like myself. I also have a question for you—which studios use Klipsch loudspeaker systems as monitors?

—Tom Richert
Fort Collins, Co.

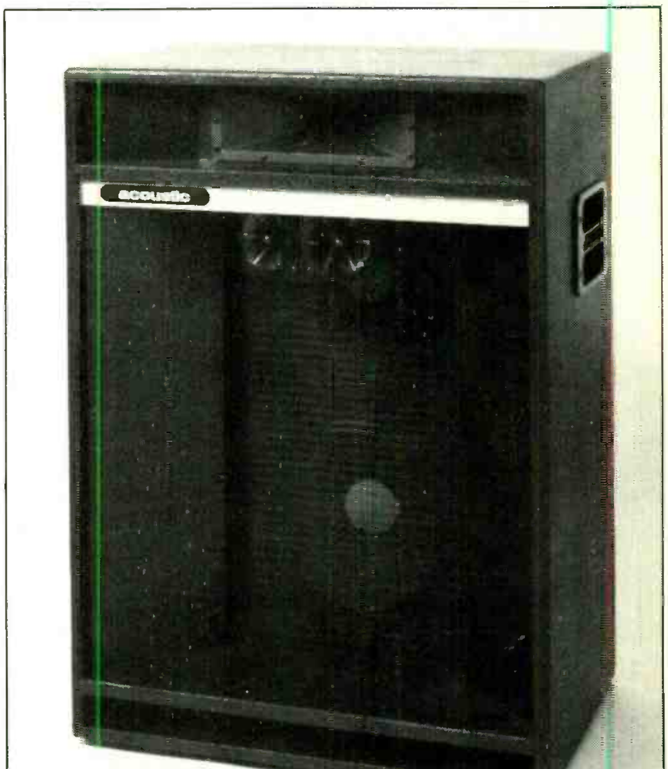
Don Peterson and Don Keel from Klipsch tell us that the following studios are using Klipsch systems: American Studios and Master Control Studios, Nashville, Tennessee, and Audio Graphics, Royal Oak, Michigan.

QUICK,

WHAT COMPANY MAKES THE MOST EFFICIENT P.A. AND DISCO SYSTEMS UNDER $500?



Acoustic Model 806



Acoustic Model 807

WRONG,

IT'S ACOUSTIC!

Introducing the Acoustic 806 and 807, two new speaker designs for maximum performance at a very modest price. Both are three way systems that cover the audio range with incredible efficiency. And efficiency means more dollars in your pocket since they don't require a monster amplifier to get the sound out. Visit your nearest Acoustic dealer, and compare the 806 and 807 to whatever you thought was the best. You'll be pleasantly surprised. Now check the price tags. When we said "Under $500," we meant "WAY UNDER $500." From Acoustic . . . the surprising company.

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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 *reader's technical forum.*

A Clarification of Terms

I have heard the term "re-entrant horn" greatly misused. There is also a bit of confusion with the terms "rear loaded" and "front loaded." Could you clarify these terms for me?

—Paul Lazarro
Waterbury, Ct.

The terms "re-entrant," "rear loaded," and "front loaded" all refer to a horn type of device. A horn (or horn loaded) device is much more efficient (more sound output for the same power input) than vented enclosures (bass reflex). This type of device is required in many public address and hi-fi applications because amplifier power is expensive.

A re-entrant horn provides a longer path length (see Figure 1) for the sound to travel which provides increased efficiency at lower frequencies and provides a better low end frequency response.

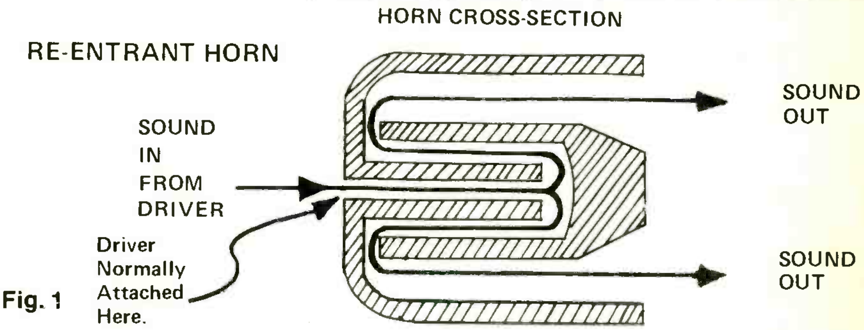


Fig. 1

The majority of these devices are small in size and they are used in fixed installations for paging, background music, etc., in areas that have fairly high ambient noise levels such as factories, playgrounds, etc. Larger versions are called folded horns and these are usually used in very large indoor/outdoor "live" concert applications.

A rear loaded horn is typically used in hi-fi speaker applications. The close proximity of the speaker to the corner of the room allows the speaker to use the walls of the room as an extension of the speaker horn thus enhancing low

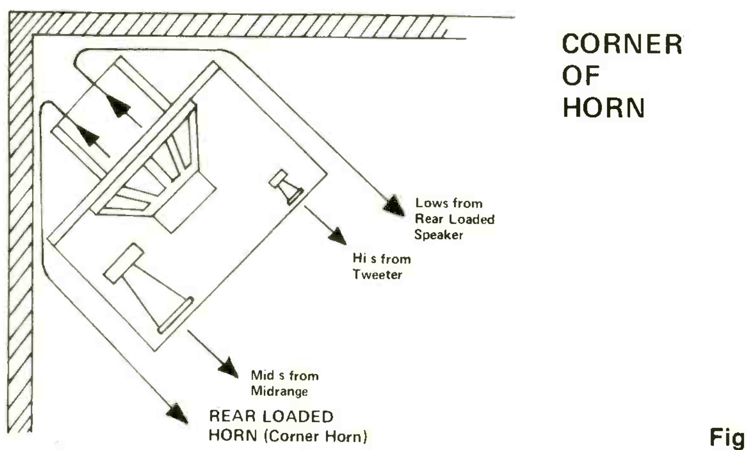


Fig. 2

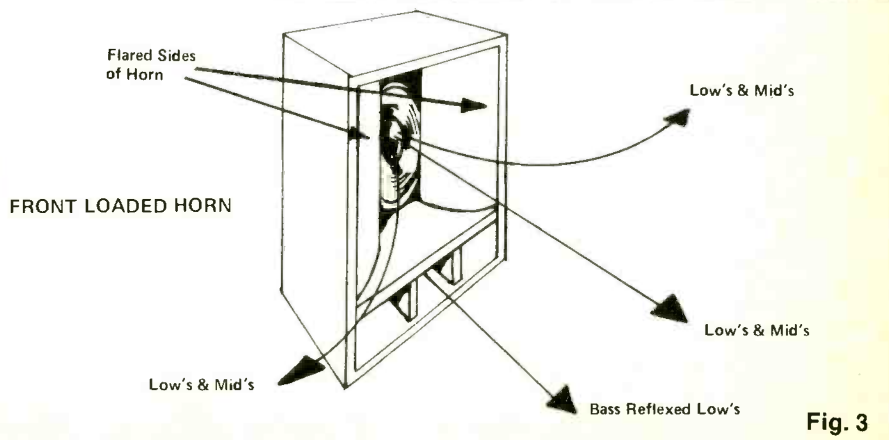


Fig. 3

end frequency response and efficiency (see Figure 2).

A front loaded horn such as can be found on today's market is typically too small to work properly. These things have to be somewhat portable. Usually the lowest frequencies need to be sustained via a tuning port (bass reflex). Even with a tuning port, the low end tends to droop as compared to a larger, well designed re-entrant (folded horn) type. (See Figure 3.) If the front loaded horn has an advantage it would be that the crossover point to the mid-range horn can be higher than the re-entrant or rear loaded horn because it has no extended path length for sound to travel as to the latter.

—Bob Herrold
Marketing Specialist—
Consumer Products
Electro-Voice, Inc.
Buchanan, Mi.

Crossover Info

What is a crossover network?

—Jack Peters
St. Louis, Mo.

Crossover networks are of two types: passive and active (electronic). Both crossovers perform the same function, but at different points in the audio system. Either type can be designed to split the audio frequency range into two or more sections: low pass, high pass or bandpass.

Passive crossovers are constructed to handle high level signals and divide the audio signal directly before the speakers.

A two-way electronic crossover will divide a low-level signal, after the pre-amp, into two distinct passbands, low and high. Active crossovers are employed in bi-amped systems, the woofer and high frequency transducer being driven by separate amplifiers. Active crossover systems are more expensive than passive networks, but the cost is more than made up for in lower distortion and higher efficiency.

—Bruce Poe
Heil Sound
Marissa, Il.

Cleaner Heads

Perhaps you can clarify something for me. Some audio salespeople have told me that the best thing to clean tape heads with is plain rubbing alcohol; others have said that it is too strong and advise buying one of the bottled solvents which range in price from $.50 to

$3.00. What's your opinion on this?

—Ken Roberts
Fort Lauderdale, Fl.

It has been my personal experience that the use of rubbing alcohol is not abrasive in any way when used to clean tape recorder heads. I've also recommended it to others who use it religiously in the cleaning of the head's pinch roller and capstan.

I would prefer, if one could obtain it, the use of denatured alcohol since I have achieved the best results with this. In any case, either one will do nicely with no adverse effects on the tape recorder.

—Claude E. Dunn
Tape Specialist
Technics Division
Panasonic Co.
Secaucus, N.J.

While it is true that rubbing alcohol will remove tape oxide residue very nicely from both heads—as well as all other metal tape-path parts—and rubber pinch rollers, continued use of alcohol will create a nasty problem for the heads, in that the esters in alcohol (of *any* type) react with the varnish on the surface of the head and dissolve it bit by bit. After a while, your expensive tape heads have no protection from the moisture in the air and will begin to oxidize themselves. To put it more simply, alcohol will eventually cause your heads to rust.

You do not, however, have to buy pre-packaged, retail brands of tape head cleaner. Almost all professionals use a much cheaper substitute, which does *not* dissolve the protective varnish and still cleans things up nicely. The magic elixir is common lighter fluid, of the type used in wick lighters. (Lighter fluid also does less harm to the rubber parts in the tape path than does alcohol.) A 12-oz. can which cost me 99 cents three years ago is still going strong after cleaning three decks at least twice a day.

—T. H. Richards
President
ILNY Records, Inc.
New York, N.Y.

Bias Settings

I have a Pioneer CT-F9191 cassette deck which has a (high) bias button. I have been using Fuji tapes lately and like the sound I'm getting, but my question is this: should I activate the bias control with this type of tape? It isn't indicated on the tape package whether it is high

or low bias and I can't really hear that much difference either way.

—Martin Gelatt
Peoria, Il.

[To best answer this question, Pioneer has provided us with the following chart. If this information doesn't completely solve your problem, we suggest you contact Pioneer or Fuji personally.]

BIAS & EQ SELECTOR BUTTONS

Bias and equalization selector buttons are provided for matching tape characteristics in order to derive full tape performance and produce low distortion recordings. Although these buttons can be set according to personal preference, Table 2 shows the recommended settings based on tape types.

Major Tape Brands & Button Settings — Table 2

BIAS & EQ SETTINGS	TAPE	
STD POSITION (BUTTONS IN UNDEPRESSED POSITION)	MEMOREX	C-60, C-90
	BASF	C-60LH, C-90H
	AGFA	C-60, C-90 SUPER C-60+6 SUPER C-90+6
	SCOTCH	C-60, C-90(DYNARANGE)
	MAXELL	LN C-60, C-90 UD C-60, C-90 UDXL C-60
	TDK	D C-60, D C-90 SD C-60, SD C-90 ED C-60, ED C-90
	FUJI	FM C-60, FL C-60, FX C-60 FM C-90, FL C-90, FX C-90
	SONY	C-60, C-90 C-60HF, C-90HF
CrO_2 POSITION (BUTTONS IN DEPRESSED POSITION)	MEMOREX	CHROMIUM DIOXIDE C-60 CHROMIUM DIOXIDE C-90
	BASF	CHROMDIOXID C-60 CHROMDIOXID C-90
	PHILIPS	CHROMIUM DIOXIDE C-60 CHROMIUM DIOXIDE C-90
	MAXELL	CHROME DIOXIDE C-60(CR) CHROME DIOXIDE C-90(CR)
	TDK	KR C-60, KR C-90 SA-C-60
	FUJI	FC C-60, FC C-90
	SONY	C-60CR, C-90CR
DURING RECORDING BIAS-STD EQ-CrO_2	SONY	DUAD C-60 C-90
DURING PLAY EQ-CrO_2	SCOTCH	CLASSIC C-60 CLASSIC C-90 In some cases, setting EQ to STD may be preferable on playback.

In addition to these, different button settings according to tape type may provide improved results.

NOTES:

1. *When playing commercially pre-recorded chrome tape, set EQ to CrO_2 (depressed) for 70µs high frequency response tape, and to STD (undepressed) for general type chrome tape.*
2. *If the chrome tape is provided with indexing holes, the CT-F9191 BIAS and EQ settings become automatically performed. In this case, it is not necessary to operate these buttons.*

—Pioneer, Inc.
Carlstadt, N.J.

Capacitance Requirements

I have a Dynaco PAT-4 preamplifier which has served me well for the past five years. However, I have always been less than satisfied with the sound from my turntable. My other equipment includes a Garrard Z-2000 B turntable, a Dynaco ST-120 amplifier and a Shure M91ED cartridge. Assuming the Dynaco PAT-4 preamplifier, the Dynaco ST-120 amp, phono and loudspeakers are all working properly, my question is this: Is the match of the preamp and the phono cartridge critical? A friend of mine has a circuit that somehow matches the magnetic phono cartridge and magnetic phono preamp capacitance. I need to know the magnetic preamp input capacitance and DC resistance of my equipment. The circuit would ideally result in a "flatter" response when using the Shure cartridge. Does this sound right to you? Any comments on this or any additional suggestions will be appreciated.

—Paul Tenhula
Bessemer, Pa.

The total capacitive load pl conventional magnetic phon does indeed have a significant e the overall sound when they are with state of the art high fidelity equ ment. This effect is particularly significant in the mid-range linearity and high frequency response of the phono playback system.

We recommend a simple procedure for matching the phono input of the PAT-4 to the optimum capacitance requirements of the cartridge with which it is being used. Subtract the total wiring harness capacitance of your turntable

...acitance recom-
...ige. In your case,
... pf as the total
...itance of the
... Shure recom-
... optimum capa-
...stereo cartridges.
... 10 pf = 340 pf.
Since the phono input capacitance of the PAT-4 is negligible, you can simply add the amount of the capacitance arrived at from the above equation. Purchase the nearest standard value (330 pf or 360 pf) disc capacitor, 100 V, 20% should be sufficient, and solder it in parallel with the phono input socket loop and shield lugs. Repeat this procedure for the other channel.

This approach will also work quite well in the Dynaco SCA-80Q and PAS-3X.

—Bill Cheadle
Technical Services
Dynaco, Inc.
Blackwood, N.J.

Help Improve My Ratio

I recently purchased an Ampex 351 with the old electronics (the tube type). Unfortunately, I do not have the equipment to measure this myself, so could you please tell me which of today's new tapes give a better signal-to-noise ratio. Since I always bulk my tape before recording, can I disconnect the erase head to get a better signal-to-noise ratio?

—Norman Thomas
Roslyn, N.Y.

The question asked can only be answered in two parts. Modern tapes differ from tape available when the Ampex 350 series was young in two important ways. First, their bias noise is lower, and second, the maximum level that can be recorded on them, for a given amount of distortion, has increased.

Since the signal-to-noise ratio is strictly speaking the ratio of signal at some distortion level to the noise, you can see that modern tapes are better at both ends. For example: a professional mastering tape like Ampex "Grandmaster" can be recorded 6 dB "hotter" than older tapes, with the same (or even less) distortion. The biased tape noise is 3-5 dB quieter also.

Therefore, to take the fullest advantage of the improvements, you should reduce your playback gain by 4-6 dB, and increase your record gain by a like amount. This keeps your VU meter registering correctly. In addition, when the above two adjustments have been made, the record calibrate will also need resetting.

Unfortunately the record equalization may also require some modification to "fit" the better high-frequency response of the newer tapes. Your local Ampex field service office should be able to advise on this.

The second question is simpler to answer. If the erase head is not magnetized, then there will be no benefit to be obtained by disconnecting it. In addition, disconnecting the erase head will disturb the bias circuitry. If the erase head, or any other head is magnetized, it should be degaussed using a normal head demagnetizer.

—Alastair Heaslett
Professional Audio Engineer
Ampex
Redwood City, Ca.

Make Mine Mono

With a tube stereo amp, can you safely parallel the two outputs (make them monaural) without decreasing stability? I've tried, but the results have always been blown tubes. Is there any trans-

former available that could safely do the trick?

—Kenneth Beukelaer
Lynbrook, New York

Many stereo tube amplifiers can be connected for increased power output (approximately two times the per channel rating) as monophonic amplifiers. To be sure it will work for yours, check with the amplifier manufacturer.

For mono operation the two channels should be paralleled. Be sure both amplifier inputs receive the same signal (a "Y" type cable connector could be utilized). Then connect the ground or common terminals together for the common side of the speaker drive connection. For an 8-ohm loudspeaker, connect the 16-ohm terminals together for the speaker "hot" drive connection (connect the 8-ohm terminals together for 4-ohm speakers, etc.). When using a 16-ohm loudspeaker in this fashion, the amplifier's maximum power output is not as great because of the impedance mismatch, but there is no loss of power when using a 4-ohm or 8-ohm loudspeaker as specified above.

—Wade D. Burns
Director of Engineering
Dynaco, Inc.
Blackwood, N.J.

Stop That Buzz!

In taping off someone else's audio board, how do I eliminate the buzz and interference from the dimming system of the lights in the room that I am recording in?

Why do they put carpeting inside speakers at concerts?

—Samantha Georges
Columbia, Mo.

The buzz that you refer to is RFI (Radio Frequency Interference) from the SCR-type light dimmers that are very commonly used today.

There are a couple of approaches to the problem. Firstly, there are many simple RFI filters available commercially. These are of several types. One type is placed in the wall socket into which the power cord of the equipment you are using is being plugged. The equipment is then plugged into the filter. This type of filter helps to bypass RFI which is entering through the AC power line. Another approach is to use a type of RFI bypass filter which is placed in the audio line between the board and the tape machine. This type of filter is really just a simple bypass capacitor between the audio line and ground. About .001

mfd will work fine. Be sure AC grounds are okay.

If this does not solve your problem, (and I wonder how the person who owns the board you refer to deals with this) you might examine the possibility of RF shielding at the dimmers themselves.

Of course, if you are using microphones in this type of environment, all the cable shields should be checked to make sure they are intact. The metal shells of the microphone cable connectors should be connected to pin 1 (ground) and the braided shield, so that the shells are grounded to the shields. Good Luck!

I expect that the carpeting to which you refer in your second question is the sound absorbent material commonly used to deaden resonances inside speaker cabinets.

Most speaker cabinets use fiberglass batting or carpet or some other acoustically absorbent material inside the cabinet to absorb unwanted frequencies and dampen high frequency resonances within the enclosure.

—Guruka Singh Khalsa
Engineer
Appalachia Sound Studios
Chillicothe, Oh.

Protect Your Edges

When using my cheap tape machine to rewind tapes, the tape gets rewound unevenly. I am concerned with any damage to the edges of the tape. Friends have advised me to use plastic reels instead of the metal ones because the plastic reels are narrower. This seems to help some, but not enough. What can I use to improve this situation?

—Jake Wingfield
Little Rock, Ark.

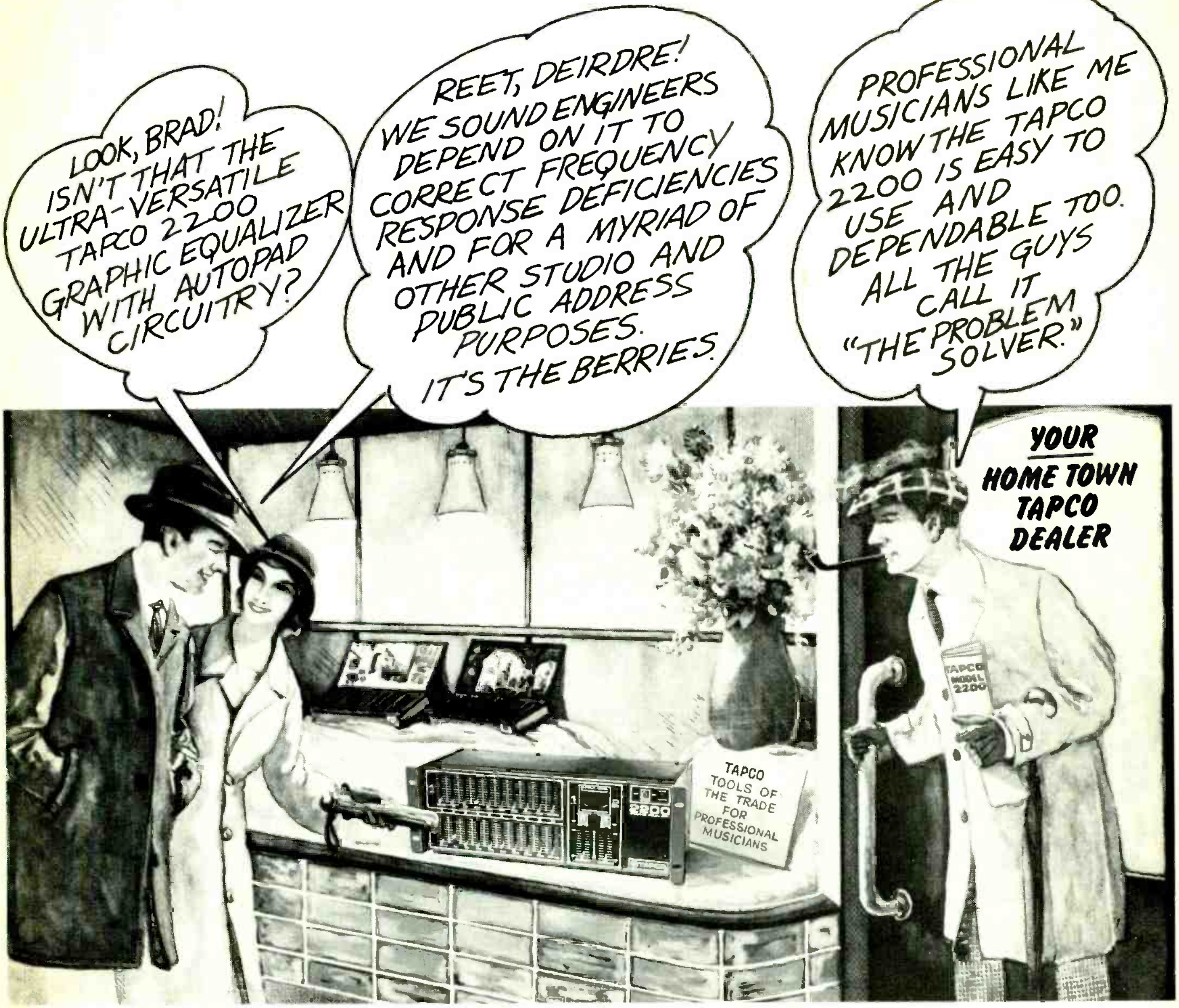
A good tape wind is not solely dependent on your tape machine. If the tape you are using is poorly slit, even the best of equipment may not produce an even wind. In addition, non-backcoated tapes do not generally wind as well as do backcoated.

If you are using a good quality tape, there are several mechanical components you can check on your machine to insure that it produces its best wind possible. All guides and lifters should be perpendicular, wobble free, and correctly adjusted for height. All bearings in the tape path should rotate freely and smoothly. Reel turntables and spindles should rotate without noticeable wobble or play.

Sufficient hold-back tension is also a critical component for a smooth tape pack. Older professional machines (before the advent of servo-controlled spooling motors) achieved a smooth tape pack by maintaining fairly high hold-back tensions in their wind modes. Consumer tape machines, however, generally produce significantly less spooling motor torque (if they have such motors at all). Thus, in order to wind tape as fast as possible, hold back tension is reduced to a minimum and much of the control of the tape pack is lost in the wind modes. This is the reason that tapes fast-wound on consumer machines are much more loosely packed than tapes which have been allowed to play through.

The best way to protect tapes from edge damage is to follow standard recording studio practice. Allow your tapes to wind through in the play mode, store them tails out, and rewind them only when they are next going to be used.

—Frederick C. Layn
Engineer
Studer America
Nashville, Tn.



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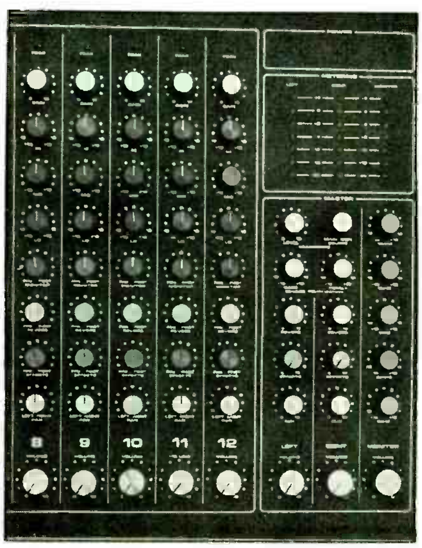
CIRCLE 30 ON READER SERVICE CARD

THE PRODUCT SCENE

By Norman Eisenberg

VERSATILE MIXING CONSOLE

Designed for both sound reinforcement and recording is the new Model 1202 stereo mixing console from Tangent Systems, Inc., of Phoenix, Arizona. Each input channel has balanced lo-Z and unbalanced hi-Z mic connections. Line-level signals for multitrack recording mixdown also may be used. Each channel also has three-band EQ with a ±15 dB range. The 1202 includes pre/post capability for its effects-send, reverb-send and monitor-send controls. Stereo-pan and channel-volume controls are provided. Five-band EQ for fighting feedback is offered on the monitor output; bass and treble controls provide reverb-return-EQ on the mains. The mic preamp has variable gain in conjunction with an LED peak indicator. Headphone volume and source-select controls permit monitoring of the mains or monitor channel through a built-in headphone amplifier. Each main has an auxiliary input; both mains and monitor outputs are balanced. The LED array provides level indications for left main, right main and monitor outputs.



CIRCLE 14 ON READER SERVICE CARD

NEW ITEMS FROM INTERFACE

Interface Electronics of Houston, Texas has announced two new items in its extensive line of studio products. The Series 316 includes 16-track recording consoles available in either 24- or 32-input mainframes with sixteen pushbutton-selected outputs plus four cue/echo sends which can be either pre- or post-slider. The complete console comes with masters, 4-inch lighted VU meters, monitor mixdown with solo on monitor only and talk/slate module. Input modules (types 316D and 316B) are similar to those known as types 108D and 108B except that they have the sixteen trackswitching buttons and a longer slider. In the 316B a parametric midfrequency equalizer is included. Also included are phantom power, phase reverse, panpot, six-step preamp gain set-switch with two input pad positions and the long-travel Duncan conductive plastic slider attenuator. Construction is all modular and plug-in; numerous options are available in both input and output sections, and mainframes may be ordered with less than a full complement of input modules (more modules may be added later). Specifications include overall response within ±1dB, 20 Hz to 20 kHz and ±12 dB of EQ at specified frequencies. Distortion (at 400 Hz) is given as under 0.1% up to almost the clipping level. Maximum level is about 10 volts RMS with bridging load; since zero level is 1-volt RMS, headroom is of the order of 20 dB. Noise is rated at −126 dBm.

Interface's other new item is the model 104/108B parametric input module which has a mid-frequency equalizer that is tuneable from 150 to 7500 Hz with adjustable Q, and a maximum of 15 dB boost or cut. The Q remains constant as frequency is tuned.

CIRCLE 5 ON READER SERVICE CARD

MEMOREX RELEASES QUANTUM TAPE

Described as an advanced ferric-oxide tape engineered to provide "outstanding characteristics," Quantum open-reel tape from Memorex is now being released for retail sale. Performance advantages claimed for Quantum include low harmonic distortion, very high sensitivity, excellent S/N ratio and high saturation. The enhanced dynamic range of Quantum, says Memorex, allows the recordist to drive the tape "much harder before experiencing distortion." Memorex's tests indicate that Quantum permits up to 4 dB more of signal, and its ferric-oxide formulation permits using the tape without special bias. Quantum tape will be available on 7-inch reels in 1800- and 2400-foot lengths, and on a 10½-inch aluminum reel in a 3600-foot length.

CIRCLE 9 ON READER SERVICE CARD

PEAK LIMITER-COMPRESSOR

Ashly Audio of Rochester, N.Y. is offering a new low-priced ($299 suggested list) peak limiter-compressor said to fit a variety of applications. Accurate, independent adjustment of all AGC characteristics is accomplished by a closed loop detector circuit that keeps the output ceiling accurate at high compression ratios, yet remains smooth down "to a gentle 2:1 ratio." The wide range of attack, release and ratio adjustment allows tailoring of the limiting action to suit any program source. Pumping and breathing effects are said to be eliminated by a program-dependent dual-release-time action that provides quick recovery from isolated transients while allowing slower release from sustained overdrive. An LED display shows gain reduction

and threshold. Suggested applications include loudspeaker protection, vocal compression, broadcast limiting, loudness enhancement and special effects for musical instruments.

CIRCLE 7 ON READER SERVICE CARD

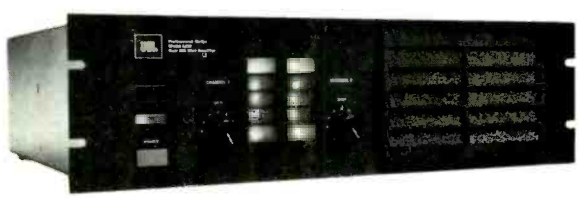
NEW SPECTRUM ANALYZER

From Tektronix comes word of its new model 5L4N spectrum analyzer. Covering the range from 0 to 100 kHz, the device features pushbutton selection of 50-ohm, 600-ohm or 1 megohm input impedance, with calibration appropriate to the selected Z. Dynamic range is 80 dB; IM is rated as better than 70-dB down from two full-screen signals. The input is single-ended for linear operation. Resolution bandwidth is 10 Hz to 3 kHz. Featured in the output is a tracking generator as a 600-ohm source of the analyzer's input frequency.

CIRCLE 2 ON READER SERVICE CARD

JBL ANNOUNCES AMPLIFIER

A dual-channel power amp, the model 6233, has been announced by JBL's professional division. The unit is rated for the following power outputs: 300 watts RMS per channel into 4 ohms; 200 watts RMS per channel into 8 ohms; 700 watts RMS bridged into 8 ohms; 400 watts RMS bridged into 16 ohms. S/N ratio is given as 100 dB, 20 Hz to 20 kHz; THD is less than 0.05% across the same range. Weighing 34½ pounds, the amplifier features an automatic two-speed fan; a chatter-free protection system for any non-standard load, including momentary short circuit; thermal protect with auto-reset; no turn-on transients; and rear-panel switch to select 100-120 or 200-240 VAC. An accessory model 5195 matching/bridging transformer is available for 15,000-ohm bridging or 600-ohm matching (one is required per channel). Con-

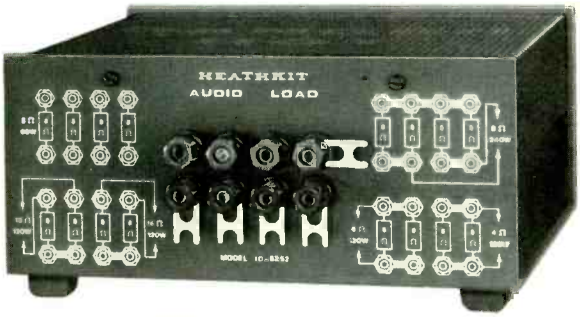


trols include power switch, level adjustment and the line voltage selector. Indicators include green for power on, red for protect on each channel and five sequential lights per channel for level. The amplifier fits standard rack mounts. Input connectors are XL-type, 3-pin female latching. According to JBL the amplifier's power supply is shielded and filtered sufficiently to permit stacking the unit with tuners or tape decks without the danger of magnetic, electrostatic or thermal interference.

CIRCLE 18 ON READER SERVICE CARD

AUDIO LOAD KIT

A new kit from Heath, the model ID-5252, provides audio loads of 2, 4, 8, 16 or 32 ohms as an aid in amplifier testing according to manufacturer's specifications and the IHF standard. A series of five-way binding posts allows the device to handle up to 240 watts mono, or four 60-watt inputs, into 8 ohms. In addition to the various resistor values, there are jacks for connecting a voltmeter, oscilloscope or other instruments. The unit includes four 3-foot no. 12-gauge leads with spade lugs. Mail-order price is $44.95.



CIRCLE 1 ON READER SERVICE CARD

NEW 100-WATT PER CHANNEL AMPLIFIER

Designed for professional studio monitoring and sound system applications is the new model 7100 stereo power amplifier from Modular Audio Products, division of Modular Devices, Inc., Bohemia, N.Y. The model 7100 is rated for 100 watts RMS output per channel into 4 ohms, or 75 watts RMS per channel into 8 ohms. Bridged mono output into 8 ohms is 200 watts RMS or 150 watts RMS into 16 ohms. A rear switch permits instant stereo/mono selection without internal wiring changes. Measuring only 3½ inches high, the amplifier has provision for either 19-inch rack-mounting or bench/shelf top mount. Featuring high-impedance bridging inputs with a dual IC operational preamp, the design incorporates two high-current, high-voltage hybrid op-amp modules in the direct-coupled output stages. Noise is rated at 110 dB below rated output, with maximum distortion of 0.1%. The amplifier may be ordered with internal optional 70.7-volt line-output transformers—particularly useful for P.A. systems and high-quality sound/speaker setups. When so equipped, the amp becomes model MAP 7100-1, and will provide an audio system with stereo power of 100 watts RMS per channel at 70.7 volts RMS into 50-ohm loads.

CIRCLE 6 ON READER SERVICE CARD

NEW TAPE ACCESSORIES

From Bib Hi-Fi Accessories Ltd. of England, comes news of two new kits for tape users. One is a tape-head maintenance kit which includes a cleaning tool with interchangeable head for easy access to tape heads regardless of angle of entry and claimed to be usable on reel-to-reel, cassette and cartridge decks. With the tool comes an anti-static fluid, inspection mirror, brush, cloth and pads all packed in a hinged plastic box.

The other device from Bib is a newly patented splicer for cassette editing and splicing. Diagonal and butt splices can be made with it. Included in this kit are tape cutters, tape piercer, splicing tape and an extractor-winder, plus instructions.

CIRCLE 20 ON READER SERVICE CARD

NAKAMICHI INTRODUCES HYBRID COMPONENT

Combining some of the features of its model 610 preamp-control unit with an advanced-design FM tuner, Nakamichi has announced a "hybrid-type" component, the model 630 FM tuner/preamplifier. Shaped and styled (wedge-like and with handles) to resemble the earlier cassette deck in this series, the model 630 includes a very high-sensitivity, low-noise phono preamp (moving-coil pickups can be used without a booster or "pre-preamp"); bass, treble and tonal contour adjustments; tape-deck monitoring and inter-dubbing facilities (two decks may be

monitored independently without affecting recordings in progress); front-panel switching for tape copying operations; built-in amplifier for headphones; Dolby noise-reduction facilities including B-type circuitry plus the 25-microsecond de-emphasis being used by many FM stations; FM tuning lights; and a large tuning dial-and-knob instead of the across-the-panel tuning dial. In addition to tape-feed and headphone monitoring outputs, the model 630 may be connected to normal power amplifiers for speaker-listening. Announced price is $600.

CIRCLE 12 ON READER SERVICE CARD

INOVONICS DESCRIBES PROFESSIONAL AUDIO PRODUCTS

A brochure from Inovonics, Inc., of Campbell, California describes products for use in professional audio applications. The model 220 Audio Level Optimizer incorporates the functions of a compressor, peak limiter and de-esser in one package. It allows both increase in average program level and positive protection from peaks. Three gating modes are available to eliminate "breathing" and "pumping" effects that occur during short program pauses.

Compatible with most professional tape transports, and able to accommodate a wide variety of original-equipment and replacement tape heads is

the model 375 Magnetic Recorder Electronics unit. The model 375 offers 3-speed EQ for any combination of NAB and IEC recording curves.

For playback, with 3-speed EQ for any combination of NAB and IEC recording curves at speeds of 3 to 30 ips, there's the model 376 Tape Reproduce Amplifier.

The Series 400 Tentrol is an accessory kit for maintaining constant tape holdback tension on most professional audio recorders and duplicator transports. Designed to reduce capstan slippage, eliminate pitch change, extend head life and improve high-frequency performance, the Series 400 handles tape widths of 1-inch or less. Tape speeds are in pairs from 3¾—7½ ips up to 60-120 ips.

Intended specifically for AM and FM broadcast use is Inovonics' model 230 Multiband Audio Processor. This device maximizes average carrier modulation while containing program peaks within prescribed limits. It features eight independent

bands with individual threshold and compression ratio adjustments. Gate expansion prevents "pumping." Both the 75- and 25-microsecond FM curves are provided. For stereo, two units may be interconnected.

Finally, there's the model 241 "Dynex"—a single-ended noise suppressor. This unit is a program-controlled filter/expander for suppressing residual background noise in audio reproduction systems.

CIRCLE 11 ON READER SERVICE CARD

AN UNUSUAL AND WORTHWHILE BOOK

Audio manufacturers over the years have ventured into areas other than equipment design and production—to wit, recordings, contests, show-biz and so on. Understandably, the area of publications is also a major "secondary" activity for manufacturers, but a new booklet just received from Lux Audio is truly unique and, as Lux puts it, may have set a new record for literature of its kind.

The 32-page illustrated publication is devoted to detailed test-reports of eighteen preamps and eighteen power amps representing in sum nineteen different manufacturers. The tests were conducted by a Japanese research laboratory at the request of Stereo Sound, Japan's leading audio magazine. The original report, in Japanese, took up thirty-two pages in Stereo Sound and included 162 charts, curves and tables of data.

With permission of the original publisher, Lux Audio has translated and reprinted the entire "megilla" and now is offering it FREE to audio-minded enthusiasts who request a copy by writing to Lux Audio of America, Ltd., 200 Aerial Way, Syosset, N.Y. 11791.

Brands named in the report include: Accuphase, C/M Labs, GAS, Harman-Kardon, Lux, Marantz, Mark Levinson, McIntosh, Onkyo/Integra, Otto/Sanyo, Pioneer, Quad, SAE, Sansui, Sony, Technics, Trio/Kenwood, Victor/JVC and Yamaha. A few models are identified only as "Brand X"—these we are told are units "whose performance fell below the standard set by the laboratory."

As you may suppose, the Lux units exhibited performance well above that standard and indeed they come in at or near the top of the pile—although they do not show up as "best" on all counts. Be that as it may, it is to Lux's everlasting credit that they saw fit to sponsor and finance this give-away (a company spokesman says that each copy of the report going into the mail costs the firm more than $2.50) which is so far above the usual promotion-type comparisons in which a given manufacturer "proves" that its products are "better" than everyone else's. Here we are presented with a raft of sober technical data on many well-known products, plus helpful explanations of the various tests and an attempt to relate measurements to listening performance. Even if you are not immediately in the market for a new amplifier or preamp, the booklet is worth getting and studying.

MUSICAL NEWSICALS

SOUND REINFORCEMENT. . .Electro-Voice, Inc. (600 Cecil Street, Buchanan, Mich. 49107) has introduced a new electret condenser microphone designed specifically for high quality sound reinforcement and recording applications. The new mic is designated the Model 1776 ($99.00), and is a "Single-D" cardioid design, which results in proximity effect, or boosting of bass frequencies when the mic is close up. Frequency response is 60 Hz to 18 kHz and transient response is said to be excellent.

Like other condenser microphones, all of which include a preamp or impedance converter circuit in the microphone itself, the 1776 has a high output level enabling it to drive almost any microphone input, high impedance or low. Unlike studio condenser mics, the 1776's condenser element is a permanently polarized electret which does not require a separate 48-volt phantom power supply for polarization; all that's necessary is a single, self-contained dry cell battery to power the preamp. The mic has a strong, machined die-cast casing finished in a non-reflective gray enamel, and a heavy wire screen over the head for ruggedness, and also features an on/off switch. The 1776 is furnished with mic stand adapter and fifteen feet of mic cable with an XLR-type connector at the microphone end, or it is available as the Model 1776P ($105.00) with stand adapter and a 25-foot cable with XLRs on both ends.

Interface Electronics (3810 Westheimer, Houston, Tx. 77027) showed their new series 316 mixing consoles at the Audio Engineering Society Convention in Los Angeles in early May. The 316 is similar to the company's familiar 104/108 series mixers except that the 316 has sixteen output buses instead of four or eight, and it uses 6-inch conductive plastic faders instead of 4-inch. Mixer mainframes are available for 16, 24 or 32 inputs and many user options in final configuration are also available. Interface also announced that all their mixers are being updated with improved IC op-amps with higher slew rates to improve transient response and to reduce transient intermodulation (TIM) distortion.

ACCESSORIES. . . There has been a lot of talk the last few years about "guitar synthesizers," and a number of very diverse electronic devices using that name have appeared on the market causing much confusion as to what a "guitar synthesizer" is or should be. All the units have roughly the same goal—to produce notes with synthesizer-type timbre using a guitar rather than a keyboard as the control device—but beyond that there are few similarities. Probably the closest thing to a true guitar-controlled synthesizer yet devised is the 360 Systems model, which caused quite a stir when it was shown at the New York convention of the Audio Engineering Society in the Fall of 1975. That system used a special guitar with a split pickup in order to get a separate output from each string, and derived control voltages proportional to pitch and amplitude envelope for each note played on each string. These control voltages in turn controlled six independent synthesizers each containing a VCO, VCF, VCA and envelope generator (ADSR). This system was incredibly versatile, allowing six guitar notes and six totally independent synthesized notes to be played simultaneously, but its sophistication and complexity included a price tag in excess of $12,000 for each of the half-dozen units produced and sold. Now, however, 360 Systems has introduced the Slavedriver Guitar/Synthesizer Interface at a much more affordable $795. The unit functions in much the same way as their original design except that it is not polyphonic and it does not include the actual synthesizer hardware. The Slavedriver comprises a special pickup to be mounted near the bridge of any solid-body six-string electric guitar and an interface unit designed to connect with an ARP, Moog, Oberheim or other make of synthesizer using 1 volt per octave control voltages and having the appropriate patch points. A pitch-to-voltage circuit derives a DC voltage proportional to the frequency or pitch of the highest note being played on the guitar, and precisely follows "bent" notes, trills or even the individual notes of a slowly-strummed chord, as well as the notes of a single-string melody. The "pitch" output of the Slavedriver normally controls

the VCO of the synthesizer so that the synthesizer will exactly track the pitch of the guitar. An octave select switch offsets the "pitch" output in 1 volt steps to transpose the synthesizer in 1 octave intervals over a five octave range, and a "transpose" footswitch further offsets the synthesizer by any interval preset by a control on the front panel allowing instant transition from unison to harmony synthesizer tuning. An envelope follower generates a control voltage which exactly follows the amplitude envelope of the note played; this "loudness" output normally drives the VCA of the synthesizer so that the synthesized note follows the guitar note in loudness and dynamics as well as in pitch. A "sustain" footswitch freezes the pitch and loudness voltages at their last values, creating "infinite sustain" of the synthesizer note. A third control output is the "trigger" output which carries a trigger pulse every time a new note is picked; this pulse is usually used to trigger the synthesizer's envelope generator (ADSR) which then controls the VCF for swept filtering of the synthesizer signal. The Slavedriver also provides a preamplified audio signal from the guitar which can be fed through the VCF or other synthesizer circuits for processed guitar notes in addition to or in place of the synthesized notes. 360 Systems also makes a version of the Slavedriver for Fender Precision or Jazz basses (also $795) and a pitch follower only for brass and reed instruments ($595). Further information on all 360 Systems products is available from the manufacturer at 2825 Hyans Street, Los Angeles, Ca. 90026.

"Thinc" stands for Technical Hardware, Inc. (P.O. Box 3609, Fullerton, Ca. 92634), and this company has developed a 256-note digital sequencer called the MMC-1 ($1195.00). In the "Program" mode, the musician programs a sequence of up to 256 notes using the

MMC-1's touch sensitive (no switch contacts) keyboard. In the "Play" modes, the sequencer reads out from its memory the information on tone, pitch and duration of notes and rests to exactly duplicate the sequence as originally programmed. By changing the clock rate, the sequence may be played back at any tempo, regardless of the original programming tempo, without altering any other characteristics of the sequence. A ½-inch liquid crystal display (LCD) continuously displays the current memory register (number 1 thru 256), the control function number (1 thru 7) and symbols to indicate several control states. The MMC-1 contains three dry-cell batteries to retain the contents of the memory even if the unit is unplugged. The MMC-1 has program-controlled outputs for pitch voltage, control voltage and gate voltage which are the only connections used for normal operation. If the optional DS-1 Decoder circuit board ($100.00) is wired into the synthesizer, the MMC-1 can also be used for programmable control of any seven functions that can be activated by an on/off switch.

An accessory item which is never fully appreciated until a favorite instrument or amplifier has been lost to damage by

ANVIL

ANVIL

ANVIL

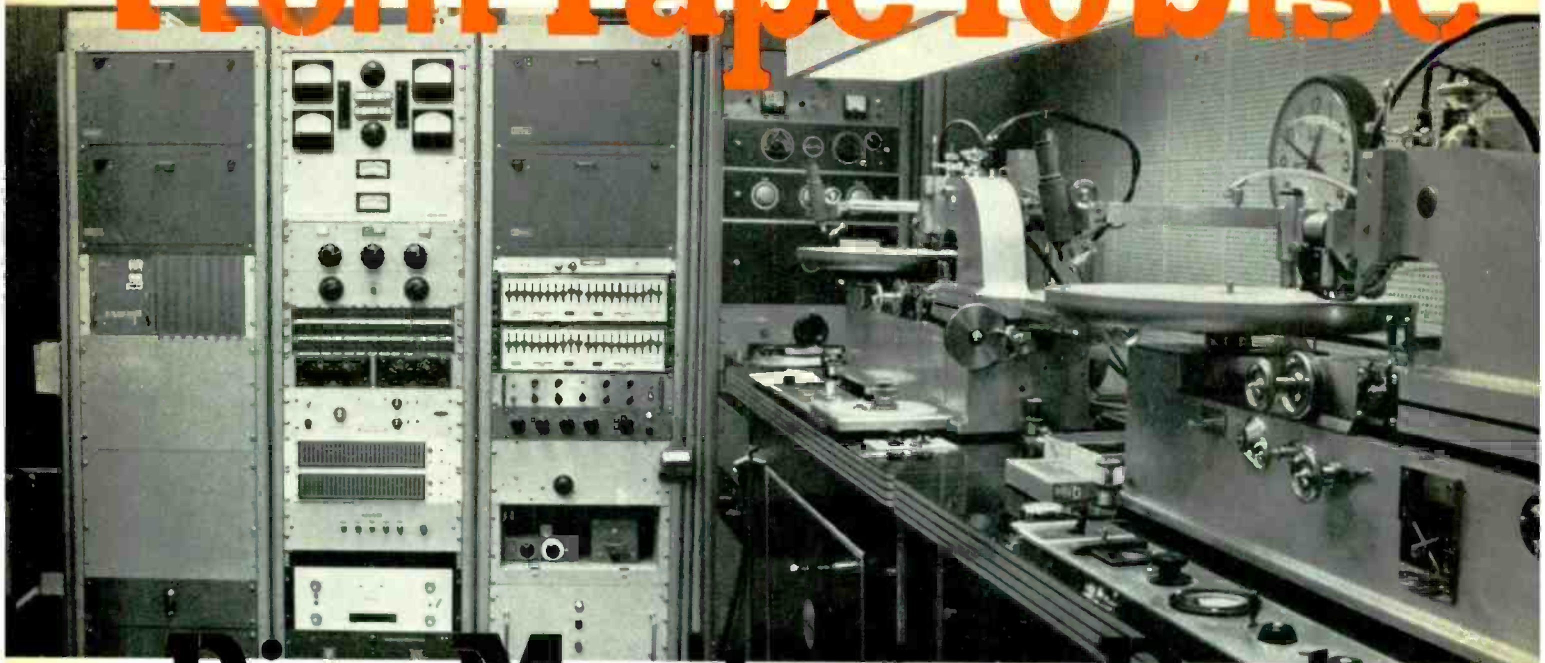
airline baggage handlers or clumsy roadies is a good road case. The brand name that has become almost synonymous with ATA-type cases (ATA stands for Air Transport Association, an airline trade group which has set standards for containers and packaging used in air shipping) is Anvil Cases (2501 North Rosemead Blvd., South El Monte, Ca. 91733). Anvil was the pioneer in fiberglas- or plastic-clad, foam-filled plywood cases to meet the ATA standards, and they are still an industry leader despite a profusion of rival manufacturers. Anvil has a very wide range of stock sizes for various applications, but they are also geared to accomodating more specialized needs. Among the latest models added to their stock lineup are customized cases for the Tapco line of audio mixers and sound reinforcement acces-

ANVIL

sories, and a new amp rack case. This latter is a case-within-a-case design which affords maximum protection while allowing easy access to the front and back of equipment mounted in the rack.

A small but potentially useful accessory item from Switchcraft, Inc. is the N3MS, a shorting plug for XLR type connectors. The N3MS has a male three-pin connector which mates with Switchcraft, Cannon or Amphenol three-pin female connectors, and features a captive 6-inch security chain to prevent loss when not in use. The normal application for this shorting plug is to terminate or "short-out" unused microphone inputs to minimize hum and noise pickup.

From Tape To Disc



Disc Mastering Part I

By Dave Moyssiadis

The studio was drab in appearance, with muted grey-brown and buff walls on which tan carpets were hung. Musicians and singers were there and they were all crowded around a megaphone-like device. The studio engineer pointed his finger into the air, and then lowered his arm deliberately and pointed at the assembly of musicians. They all began to play a song. When the song was over, they were completely silent until the engineer gave the all clear. After a few moments of discussion the take was pronounced good. The group had just made a record. Now in case you're wondering, the tape recorder hadn't been invented yet. The record had been made on a cake of wax. That "megaphone" had a diaphragm at the end of it and attached to the diaphragm was a needle. The needle had cut a groove into the cake of wax, which was driven by pure music power.

Well, we don't do it that way any more, and although we may have lost some of the spontaneity, some of the "feel," we have gained eons of versatility and productive convenience. Back then, if somebody goofed everyone went back to the beginning. Today, of course, we record on tape first and any mistakes can be edited out. In fact, some good parts can and are edited out. But we eventually still come back to that ubiquitous black disc we call a phonograph record—however circuitous the path may be.

Obscure But Not Gone

Since the introduction of the magnetic tape recorder in this country, disc cutting (or mastering) has faded into obscurity, completely overshadowed by the infinitely more versatile tape device. Disc cutting became so obscure, in fact, that even today, some record people under forty years of age haven't the foggiest idea of what happens to their tape after they pay the studio bill.

I thought you'd never ask! (Mind you, I said disc cutting had *faded* into obscurity, not disappeared. Virtually every record ever made had to be mastered on a disc-mastering lathe.) Well, first of all, some poor assistant has to sort out all the good takes from the out-takes and assemble them in the correct order with about four seconds of leader tape between each song on the album. The tape is then sent out to one of the independent mastering houses, *if* the studio doesn't have its own facilities—and most don't, or at least know enough not to get involved in matters which are more specialized than they reasonably can be expected to handle. (Disc cutting has become rather sophisticated in recent years, as we shall see.) When the tape gets to the mastering facility, the program material is transformed from the electro-magnetic medium (tape) into the mechanical medium (disc). From there it is shipped to a plating plant where the master record is electroplated and a metal stamper is derived from it, only the stamper is a negative —the grooves have become ridges. The stamper is then shipped to a pressing plant where it is placed in a giant waffle-iron-like device called a "record press." One side is affixed to the top plate, the other side to the bottom. Labels (hopefully corresponding to the proper sides) are inserted, a blob of plastic called a "biscuit" is placed in the press and a button is pressed. Then a whole bunch of things happen. The press closes under tons of hydraulic pressure, steam is circulated through labyrinths in the press to heat the vinyl so that it is certain to flow into every part of the groove and then cool water is circulated to firm up the plastic so it can be handled without coming out like an uncooked pizza pie. Next, a hole is punched somewhere near the center, the outside flash is trimmed to the familiar 11⅞-inch diameter and the finished record is stacked on a spindle. Later, the record will be packaged in a sleeve and jacket, shrink-wrapped and shipped to your friendly neighborhood record store.

The Mastering Studio

Interesting, but let's back up to the master record (which is what they tell me this article is supposed to be about). If you were able to make yourself invisible and gain entrance to a mastering studio, you would witness the following scenario:

The mastering engineer would remove the tape from the box, read the job order, place the tape on the tape recorder and rewind the tape while timing it with a high-speed timer. After adjusting the recorder to the tones supplied with the tape, he would listen to a short portion of the tape through the equipment chain, make some observations and adjustments and cue up the tape again. Then he would place a blank master on the recording lathe, make a few more adjustments, push the start button and sit back and relax. When the record ended and the recorder and lathe had stopped, he would look through a microscope positioned over the master, pronounce it good, remove it from the lathe, pack it into a box and ship it to the plating plant. Simple, right? Easy, right? Boy, wouldn't you like to get paid for a soft gig like that? If you answered yes to the first two questions, you'd be dead wrong. If you answered yes to the last question, you'd be right, but only if you were a certified nut, because you've missed about a hundred mental and mechanical processes, all of which went into a very complicated procedure.

Inside Disc Mastering

Here is what disc mastering is all about, and how it differs from recording a master tape. The first thing to clobber one's sense of security is that while tape is a very forgiving medium, disc is a cruel perfectionist, a dictator. One tiny mistake and your work is forever banished to the scrap pile. There is no erase button, no "sinister force" that can be used to conveniently remove an error or indiscretion. Also, there is no editing—you start at the beginning and go flawlessly to the end, or you don't go at all. Now, if all tapes were recorded perfectly and tailored to the disc medium, there would be no problem and what the invisible man saw would be as simple as its prima facie appearance. But alas, that is not the case. The whole problem in a coconut shell is that things that cannot be successfully transferred to disc can be and *are* recorded on tape. What is needed is a better knowledge of those things by technical studio people and by production people. This article will attempt to point out (and clear up) some of the problem areas.

It is actually unfair to say that disc recording equipment is not on a par with tape recording equipment, since state of the art disc recorders can handle almost anything a tape recorder can. What has to be understood is the discrepancy in the disc *playback* system, or record players, to use the vernacular. Disc playback technology, in its present form, has not and can not keep pace with disc recording technology. This is because of certain physical parameters within which the disc system must remain. For example, a disc is absolutely limited in how much information can be crammed onto its surface. With tape you can always splice on a few more feet, but on a disc you have only the space between 11½ inches and 4¾ inches, which is slightly over 86 square inches, and that's it—period. You can't take up an extra 1/16 of an inch—it just isn't there. If you try, you fall off the edge of the earth—so to speak—or at least off the edge of the record. So what happens if you need an extra minute? Well, you have to make a trade for less bass or less overall level. Another problem with the basic differences between the two mediums is that the tape system will handle almost any form of abuse short of putting line voltage on the input. But the disc system will not tolerate the slightest bit of hyperactiveness from the VU meters. Excessive unexpected peaks will make a mess of the master.

As mentioned earlier, you can put almost anything *on* a disc, the real problem lies in trying to get it *off* the disc. The reason for this problem is that the recording stylus is basically triangular in shape, while the playback stylus is round or oval in shape. So, we really are trying to put a round (or oval) peg in a triangular hole. The strange fact is that, oddly enough, we have been rather successful at it. Still, the basic discrepancy is there.

Before we take a closer look and try to understand some of the physical aspects of disc recording, let's look at the elements of the system.

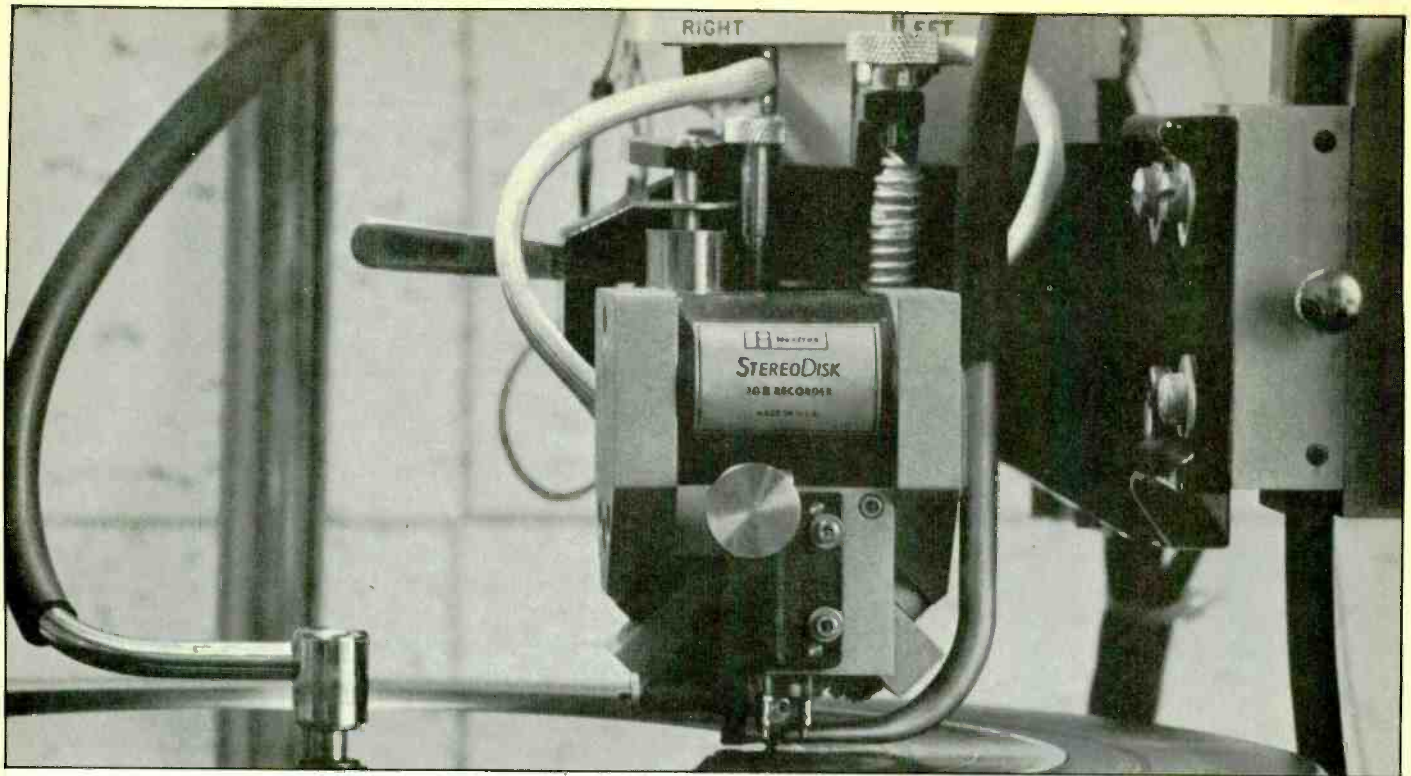
Software

The Recording Blank: This is a "perfectly" flat, stretched, aluminum disc of the highest quality, and it is coated with cellulose nitrate on both sides—even though only one side will be used. There are only four or five manufacturers in the world that make these things and three of them are in the U.S. As you may well have guessed, this lacquer [cellulose nitrate] coating must be of the highest quality and absolutely pure—totally free from even the smallest microscopic particle. It must be applied in a uniform manner under proper temperature and humidity conditions, and must dry at the proper rate—not too fast, not too slow. All of this procedure has a bearing on the cutting or mastering stage. These blanks come in several sizes and vary in quality (ranging from cruddy to almost good these days). Seven-inch diameter blanks are used only for 45 RPM reference dubs, ten-inch blanks are used for 45 RPM reference dubs and some seven-inch 45 RPM masters, twelve-inch blanks are for 45 RPM seven-inch masters and LP reference dubs, and fourteen-inch blanks are used for LP masters. Needless to say, only the highest-quality blanks are used for mastering, while the lesser quality blanks may be used for reference dubs. All masters are over-sized, that is, they are larger than the actual record that will come from them. This is for ease of handling.

The Recording Stylus: This is generally made from a synthetic ruby approximately one-half-inch long (a Japanese manufacturer has recently developed three types—diamond, sapphire and, of course, ruby) and cut with several facets. Some facets are visible to the naked eye, while other facets are so small they are measured in microns (millionths of an inch). Each facet has a purpose.

Hardware

The Lathe or Disc-Cutting (Recording) Machine: This is a highly complex, expensive and exacting machine, but, in truth, it is nothing more than a lathe that a machinist might use. It operates on the same principles as any other lathe. It is just adapted to the specialized job of cutting microscopic grooves on a flat disc. There are of



Westrex Stereo Disc-Cutting Head

course many parts to it, but basically there is the *base*, the *carriage* and the *cutter head*. Obviously, the base holds the whole thing together; the carriage rides on tracks in the base and supports the cutter head; and the cutter head is what holds the recording stylus and drives the stylus up and down and from side to side to carve out the groove in the record blank. By itself, the lathe can do nothing but cut quiet spirals on the blank.

The peripheral equipment has grown enormously complex in recent years to help the basic lathe do its job better and better. Simplest to understand is the signal reproducing and processing equipment, much of which is the same stuff you find laying around the average recording studio. This consists of a tape recorder (used only for playback), equalizers, limiters, cross-overs, filters, gain controls, meter monitoring equipment, amplifiers and speakers. All of which are used in similar application as their counterparts in the recording studio. More specialized is the preview and variable pitch/variable depth systems and their related and recently introduced computer systems.

The above is the equipment needed to make a master phonograph record. But as with anything else it is useless without the skilled guidance of a real live human being. Though computers appear to be taking over even in this specialized world of record production, with automated recording consoles and automated disc recording machines, in the disc mastering field, there will never be that take over that people in the outside world fear. The meddlesome human touch is essential in this field. And even more important, there isn't a computer wacky enough to put up with we neurotics. So, we are safe from control by computers. It is *we* who will control *them* and triumph over all! So there! So here we have a computer that for once is under our control instead of vice versa, but what do we do with it? First a bit about the parameters and specifications which must be adhered to in disc cutting. And mind you, most are iron clad *law*. These laws do not bend, they are only broken and punishment is swift and sure—total rejection. Let us for the sake of simplicity discuss only the LP record.

The Master Record

Just what is a master record anyway? Well, it's simply a V-shaped groove about a quarter-mile long cut into the surface of an acetate record blank. It starts at the outer diameter and slowly spirals in toward the center. *How* it spirals in is of prime concern when we cut a record. If you go to your record collection, pick out a record—any LP will do—and look closely at it, you will notice several things that you probably never really took note of before, although they were always there and you saw them every time you picked up the record. At the outer edge you will notice very coarse grooves, so coarse that you can actually count them. You will see about two or three of them, sometimes four or five. This is called the "lead-in." Then you will note that where the music begins, the grooves get very close together. Here the grooves are so close that as many as 150 to 400 of them can fit in the space of about an inch. At the end of the first song you will see what appears to be a small space or gap. This is not really a gap, if you look really close, perhaps with the aid of a big magnifying glass, you will note that there is one turn of a groove that spans this small gap. This part of the groove is called the "band" or "spiral." There may be several of these depending on how many songs there are on the record. At the end of the last song you will note that the groove moves very quickly toward the label area and stops. This part is called the "lead-out," and the part where it stops is simply a circle that no longer spirals in, and it is called the "lock groove."

All of these things occur at specified diameters. The lead-in leads in from a diameter of twelve inches (which is not on the record—for some reason, a 12-inch record is only $11\frac{7}{8}$) inches to a diameter of $11\frac{1}{2}$ inches where the music begins. The music *must* end no closer to the center than $4\frac{3}{4}$ inches or the record will be rejected by the guys at quality control. Even if it gets past them chances are that (take a deep breath here, you'll need it) an automatic changer will reject the record before it is over, which will cause the guy who bought the record to run back to the store demanding his money back, which will cause the record store to run back to the distributor demanding its money back, which will cause the distributor to run back to the record company demanding its money back, which will cause the record company to dump 400,000 pressings on the mastering engineer's door step so they can be eaten for lunch. This tends to be most embarrassing.

Finally, the lock groove must be at exactly $4\frac{3}{16}$ inches from the center of the record. You will note that all these figures are in diameters rather than radii. Just divide by two to get the actual distance from the center. For example, the lock groove is really $2\frac{3}{32}$ inches from the center.

The rate at which the groove moves inward toward the center is called the "pitch," the same kind of pitch as would describe the pitch of a screw or bolt. Pitch is given in "lines per inch," and that is exactly what it means. "100 lpi" means that for every inch there are 100 lines or grooves. Normal record pitches range roughly from 70 lpi to about 300-400 lpi. Lead-in and banding pitches are somewhat coarser being from 8-32 lpi, and lead-out pitch is very coarse, 2-4 lpi.

When cutting a record, it is the total time per side that determines the recording pitch. Now, you may have noticed that among different records some were louder than others. Take note that the louder ones are shorter. This is because the longer the record the finer the pitch has to be, and this means that the grooves have to be closer together. All of which leaves less room to wiggle and less wiggle equals less level or loudness. Less level often results in a very loud sound from the record producer. There is a theory in disc cutting called the *Inversed Producer Law* which states that:

The loudness of the producer's scream is inversely proportional to the square of the loudness of the record. A theory born from hearing producers screaming into the phone "cut it HOTTER! HOTTER!! *HOTTER!*" Which of course is impossible. Producers are extremely proficient at requesting the impossible. All that aside, there are other considerations that help make matters worse. One is that bass and low frequency information make very big wiggles even at low levels. A problem which occurs because the vast majority of power in music lies in the lower portion of the frequency spectrum, and even the RIAA curve, which tries to even out the proportions, does not fully solve the problem.

Another problem is that in order to get any stereo effect, there must be a difference between the two groove walls. A typical stereo signal is really two separate and independent signals on the left and right walls. Any common information (the instruments that are in the center) will affect both walls the same as a mono signal. So you have to picture the mono signal showing both walls moving in unison while two separate signals are superimposed on the left and right walls respectively. The trick here is to not let the up and down wiggling get out of hand, because if there is too much up and down wiggling, the needle will literally wiggle right off the record. The groove disappears, and this is one of the things that makes a record skip. Another cause of a record skip is if the side-to-side wiggle is so severe that it wiggles right into the next groove and breaks down the walls of the grooves. This is called an "over-cut." One other reason for a record to skip is if the program material contains a component which causes such extreme lateral acceleration that the groove actually makes a right angle so sharp that the playback stylus cannot follow it. The stylus will continue in a straight line instead of tracing the groove and jump right out of the groove and over into the next one. Those are the three things that make a record skip, and the things that disc mastering engineers look for under the microscope. By the way, these wiggles that we have been talking about are called "excursions" (not to be confused with incursions), so to sound professional we call side-to-side wiggles "lateral excursions" and up-and-down wiggles "vertical excursions."

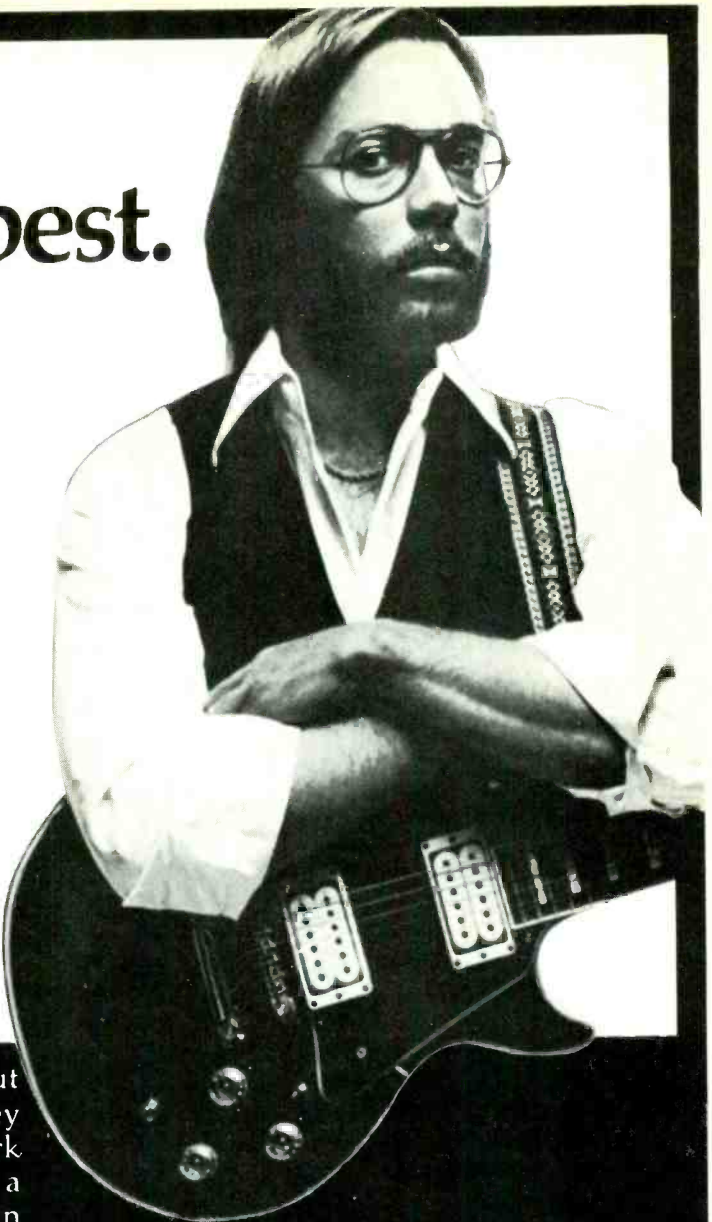
In order to keep somewhere between the sky and the deep shiny aluminum, normal groove depths range from 1½ to 5 mils [1 mil equals .001 of a an inch]. You don't want to get any shallower than 1½ to 2 mils deep, since playback stylii are either 1 mil or 0.7 mil and obviously a groove shallower than that would not be sufficient to hold the stylus in the groove. On the other hand, 5 mils is about as deep as you would want to go since the lacquer is only about 8-10 mils deep. If the guy who was spraying the lacquer on the disc in the factory was in a hurry that day, the lacquer might be even thinner. So if there is any great vertical excursion the groove might go as deep as 7-8 mils, and at that depth it may be possible to cut right down to the aluminum —ruining the master and the recording stylus. Cutting too deep also makes it difficult to plate in the plating process.

One more problem to consider is high frequencies. Although these do not take up much room (very little excursion) even at high levels, they still cause their share of problems. Did you ever hear a record with a cymbal crash that sounded more like a garbage can cover being dragged down the sidewalk? Or a female vocalist with sibilancies which seemed to smear? Well, that is called "tracing distortion." It occurs when the rounded playback stylus tip cannot fit into the notch-shaped groove that the recording stylus has cut. Since it cannot trace the groove properly, it merely chatters through that section crashing from side to side instead of gliding. Tracing error is not to be confused with tracking error which is another concept and another problem.

In the next issue we will be talking about such things as tracing and tracking error, diameter losses, vertical and lateral over-cuts, groove echo and what the disc mastering engineer must do to avoid getting those potential headaches. We will also discuss what recording studio personnel and producers can do to prevent forcing the disc guy to use his arsenal of vicious anti-problem weapons on your tapes.

[Note: Because we will be illustrating many of the concepts introduced here in Part I in "From Tape to Disc—Disc Mastering, Part II" (scheduled to appear in the September issue), the reader should make an attempt to save this August issue as a source of reference. —Ed.]

One of the best.



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recording engineer

Shelly Yakus

Engineer Shelly Yakus is considered to be one of the premier recording engineers in the business today. He has made his mark by engineering sessions for a long list of prominent recording artists including Alice Cooper, John Lennon, the Raspberries, Johnny Winter, Peter, Paul and Mary, Blue Oyster Cult, Henry Gross and Chick Corea.

Shelly's recording interests began in Boston, Ma., where his father owned and operated a studio called "Ace" Recording. There he was introduced to wire recorders, mono, stereo, three-track and finally four-track recorders.

Several years later, after much soul searching and many trips between Boston and New York, Shelly came to New York (and the Y.M.C.A.) to work for A&R recording studios. He stayed at A&R for three years before moving over to his present domain, the Record Plant, in 1970.

MR: What are the qualifications needed to become an engineer?

SY: The first thing I can think of is *patience*. Patience and ears. You've got to be able to distinguish the differences among the many sounds you hear. Also, something that's very important is personality. If a guy is a phenomenal engineer but has a lousy personality, very few clients are going to want to work with him. On the other hand, if a guy has average engineering skills but a great personality, he's going to be very much in demand. It's very important to be able to put your clients at ease, because it enables them to have confidence in you.

MR: Coming from Boston, how did you end up in New York City?

SY: I was managing a group in Boston at the time, and I had some of their material. I brought it to New York to try and sell it to the record companies. I was about seventeen or eighteen and I went around to the different studios. In one of the studios I met a union representative, who I didn't know, but we became friendly and he said, "Come on, I'll take you around to all the studios." In those days, they had a union rep because it was against union rules to overdub—otherwise you had to pay the band again. He took me around and I met a guy by the name of Brooks Arthur who was working at the original Mirror Sound. I had never seen anything like what he had. For instance, he had a tape loop. The tape loop was coming off the machine, around a microphone stand in a corner of the room and back again. It was a continuous loop of thunder.

They were doing a record called "Give Us Your Blessings" on Red Bird Records. This rep introduced me to Brooks and we got talking and he said, "Look, I'm busy now and it's hard for me to talk because the producer is coming in." I asked, "Who is the producer?" He says, "Phil Spector." I didn't know who Phil Spector was in those days. I asked when the record was going to come out and he said it would be out in ten days. In ten days it *was* in the record shops and I couldn't believe it. I had heard them mixing it and all that stuff.

MR: That got you thinking a bit?

SY: Yeah, I started seeing a new way of techniques.

Getting back to Brooks Arthur, he was really nice to me. I went back to Boston, and about six months later I had to hand carry something to New York for my father so I figured I would stop in and see Brooks. But he had quit Mirror Sound and gone to work at A&R Recording. I went over to A&R looking for him but he wasn't in. However, I met a guy by the name of Roy Cicala, who is now the owner of Record Plant, and we talked and he showed me around the whole studio.

MR: You told him about your father's studio?

SY: Yeah, and at that time I said I didn't know if I really wanted to be an engineer, but I thought while I'm here I might as well apply for a job and see what happens. I started getting into it, I was disillusioned by my father's place and everything, so I went up to the office to apply for a job. They told me to come back. I said, "I'm going back to Boston today, can't somebody just talk to me for a couple of minutes?" So Don Frey, who is one of the owners of A&R Studios, interviewed me and then I went back to Boston. I had gotten a job in a TV station where they taught me how to be a cameraman—to put slides in the projectors. It was a summer job.

A&R Studios called me a couple of weeks later and asked me if I was still interested. I was down in one week. I lived at the YMCA for a bit. So I started working at A&R.

MR: Who did you study engineering under?

SY: All the guys. Phil Ramone, Roy Cicala, Tony May, Dave Sanders; they were the best. Each guy, for his own type of thing, is phenomenal.

MR: What was it like at A&R?

SY: I became an engineer after the first year; they had told me it would take at least five years. Mainly it was Roy who taught me. He taught me really without saying anything. He used to do his own projects late at night, but on paying dates we'd start at twelve in the afternoon, work till six and then from seven to twelve. Then he would start his own stuff and I would stay with him. I asked him how

is what...

Chick Corea, Alice Cooper, John Lennon & Peter, Paul & Mary have in common.

By Bob Bank

Photo by Bob Bank

come he never got tired, and he said that after a while, you get used to it. But it took me months and months to get used to the hours. I started to get into it and formulated a schedule; we were up day and night. Sleeping three, four hours a night. Roy pushed me to be an engineer; I was afraid because I didn't think I was good enough.

MR: Do you remember your first engineering session at A&R?

SY: Yes, it was for a group called Orient Express and was produced by Bob Shab.

MR: What were your actual duties at that session?

SY: Engineer. That was one of the ones that Roy gave me. Then I worked with the Amboy Dukes on which I was the assistant engineer. In those days, they would mix a stereo and a mono mix at the same time. But they would mix them both without listening to them. In other words, they'd set it up by tones on the board. They'd get the stereo balance, then push a button and listen to the mono balance—using slight adjustments to make up for what happened when you went from stereo to mono. They set up "Journey to the Center of Your Mind," which has a big guitar piece, in that manner.

Well, when Roy panned it on the mono, because he hadn't set up the mono exactly right, the level went up and down and it would disappear. It got louder and softer as opposed to going left to right. The stereo was fine, but when it came time to use the mono we couldn't use it. So Roy called me up and asked me to listen to it and I listened to it the same night he had finished it. I took it and actually remixed the middle of the song from the mono, copied it onto the tape and rolled the level up and down to correct what had happened in the mix.

MR: You worked with the Band on their *Music From Big Pink*, didn't you?

SY: Yes.

MR: How many tracks were used?

SY: Four. Two is the vocals, three is the basic drums, four is the organ and one is the strings. That's it. Then you could go four to four, combine things and then open up tracks. But in those days it was either four track or eight track.

MR: Do you remember the microphones that were used then?

SY: U-87s, U-86s, Shure 57s and E-VRE-15s. Some Sonys—that was basically it.

MR: What about today?

SY: Same stuff, because it works. As far as the microphones that we use with the rock groups, it's hard to say because I get a vocal sound by putting up five or six microphones. It's like tailoring a suit for somebody. I ask the vocalist to sing into each one, whichever one sounds good with his/her voice is the one we'll use. It's as simple as that. You could use the same microphones for hard rock that you use for classical. Certain instruments sound better with certain mics. The basic concept that I have learned is that if you have a vocal that is basically real "warm" sounding, and you use an 87, which is basically a warm sounding microphone, the midrange will be a little dull sounding. It's got plenty of top and bottom, but it's dull sounding in the middle. I call that warm. If you use a microphone that is the opposite of your vocal—a "hot" mic on a warm vocal—you usually come out with a pretty clear vocal sound.

MR: What microphones produce that "hot" sound?

SY: One is the Neumann U-86. There are some mics that are characteristically big, "booming" sounding microphones. They have a full-frequency spectrum. Use one of those on an acoustic guitar and you're in trouble. There is so much sound coming out of that guitar, so much bottom, that if you use the wrong microphone it just picks up everything. Then you have to roll it all off with an equalizer and you end up with something that sounds terrible. I'm learning all the time. A lot of things I found by accident, because it's easy to get locked into a certain thing.

I used to mic an acoustic guitar right in front—that is where most of the sound comes out—and then I had to roll off the bottom and get it clear. Now I find if I put the microphone higher than the guitar, facing down, that it misses all that extra bottom and gets a clear sound from the strings.

MR: What are some of your mixing techniques?

SY: Well, as far as techniques go, I never really thought about it, except for two important things. One is that I try to encourage the producer and the artist to be part of their own mixes. If they don't want to be, that's fine. But I find that the best mixes come out when the producer or the artist—it's usually the producer—takes half the board or the faders that he knows the best. After all, they wrote the songs, they know it better than any engineer will ever know it. Who am I to say I'm going to mix the thing and you're not going to touch the faders? How can I say that it's going to be my concept of the mix? I find that they usually have a good feel for guitars, horns and strings, and that I usually have a good feel for the rhythm— bass, drums and the vocals. I try and create a framework of a good bass sound, good drum sound and a clear vocal, and I try to make them fit their stuff into that framework. Then I know I can help them. I can control the limits.

It's a wonderful combination because I hear the rhythm differently than they would hear it, and they hear the other stuff differently than I would hear it. So together we get it, and it works. The best mixes I've done have been in combination with a producer.

MR: How do you feel about computer mixing?

SY: On most levels, I don't like it.

MR: Why?

SY: In the beginning, my theory was one of creating a mix by working at it until the mix "peaked." In other words, you keep working at it until the mix grows and gets better and better until you get a terrific mix. Usually the mix after that falls apart, so it's a good indication that you got *the* mix on the one before. If you need inserts or something to make better sections, you do it.

It seems like computerized mixing took the personal aspect out of mixing. It was then shown to me that you could do computer mixing and get yourself to the point of "getting" a mix and then updating it, if you wanted without losing that mix.

Our problem here [at the Record Plant] is that we use a lot of outboard equipment—DDLs, outside limiters and equalizers which the computer doesn't set. People were saying "well, the great thing about the computer is that two weeks later you can come back and re-mix your song without going through everything you went through before." Then you get into the thing where a guy says, "I love the mix, I listen to it at home. I love the mix, but I wish it had a little more vocal." Then he says to me he wishes we had computer mixing, because then I could do the same mix again and get a little more vocal. Well, that is true, except to get that mix that he loved we had to use tape delay, echo, EQs and all the stuff that isn't attached to the computer. If one of those things is the slightest bit different, especially the echo sound, the slightest bit, if there is dirt on the machine, then the whole mix changes. Also, I've heard where high frequency things trip the computer and make it less accurate.

MR: Do you use an individual approach for each artist/group or one usual approach for all artists in the studio?

SY: I had to establish a basic approach in the studio that I knew would work fast. I always wanted to do it differently for each person, but because some people come in and don't know what they want, and because there are other factors, like time and studio musicians, I had to come up with something I knew would work. So I have a basic foundation. The artist and I will discuss what he wants to do, and he'll play me records and sounds that he likes. Later on, we'll discuss it again, and from that foundation, I go to what he wants. Each artist is different. If an artist is unsure I just go back to what I know works and we build on it from there. Once he gets his security with the studio, me and his own project, then he'll be able to say, "Hey, let's try this and then we'll go from there."

MR: What problems do you have when you mix an album?

SY: Any time I have trouble mixing a song, I ask the producer if the song is in a minor key. He almost always says yes. Minor keys are very difficult to get clarity on. If you have a song in a minor key, I'm not saying it can't sound good, just that it's difficult to work with. Certain keys and registers are also difficult. A big problem is getting the individual band members to make a decision on a final mix before the excitement wears off. Also, sometimes people get so involved with all the outboard equipment available—flangers, phasers, DDLs—that they lose the "feel" they wanted and had originally.

MR: What is the most difficult part of the recording session as far as putting things down on tracks?

SY: Getting down the drum sound is important. It's the backbone of the record. Without a clear drum sound it's almost impossible to have a clear mix.

The hardest part of making an album for any artist is getting the artist to say "I really love my album and I want to listen to it at home." Most people can't listen to their album at home.

MR: One of the experiments that you did was miking a small speaker for the Raspberries song "Overnight Sensation—Hit Record." How did that work out?

SY: Producer Jimmy Ienner wanted the music to sound like it was coming out of a radio. First I thought of using a filter, but I decided it would *sound* filtered. Next I thought, "How about using a real radio? I have a real radio that sounds great." So we took the mix, put it through this radio and we put a really good quality microphone through it. It was a U-87. It sounded like you were hearing it over the radio. What I did was to play the mix through the radio while I had the regular mix playing in the control room—the radio was in the studio—and at the correct point I faded down the track and brought up the radio. Then, I brought the track back in. I segued the song back in over the radio and we edited it into the real song.

MR: Should an engineer be used again and again with the same artist for that "marriage" to work?

SY: I believe that if you have a hit with an engineer you should stick with him. Unless you are so uncomfortable that you can't create and you can't work. I mean, you shouldn't break a combination; there are magical combinations and there are "marriages." Bob Crewe and Bob Gaudio were magic together. When they split up, they didn't have any hits for many years. There is a certain thing that happens with the engineer and what he does with the sound. It's a fact as far as I'm concerned that while a poor sound may not stop a record from being a hit, it would certainly sell more records if it did sound good. But I find that after two, three or four albums with the same engineer, sometimes people feel too comfortable and they say "well, it's time for a change." Or they'll say "let's go to a different studio."

MR: Now, concerning the feelings and rapport between the engineer and producer, how much should the engineer say or not say during a production?

SY: That is governed by the artist and the producer. You get a feeling in the room of how much you can say and how much you can't say. I don't ever say anything unless I'm asked.

MR: They'll ask you what you think?

SY: They'll say to me, look we're having trouble coming up with a part for this. But I don't make suggestions about parts unless they ask me. If they ask me if I hear an instrument, I might say, "Well I've been thinking about it and I hear a couple of things." It might be a harp, harmonica or something like that. Sometimes my ideas work and then again sometimes they don't. I can accept that. I am there to help make them make their record sound as terrific as it can sound.

MR: What role does a producer play in the studio?

SY: It's different on different sessions. Some artists really need an actively creative producer. Other artists simply need someone to look to for a final decision—someone who knows the studio and the artist.

Milt Okun, who was Peter Paul & Mary's producer, was perfect for them because he always got the sound that the group was looking for without changing their music around. Producer Bob Ezrin, who produces Alice Cooper, is different. Because Bob has a theatrical background, he was, with the help of Alice and the band, able to inject dynamism, theatrics and excitement into those tracks. He was constantly changing things, coming up with new concepts.

Both Okun and Ezrin are very good at what they do, but they go about it in completely different ways. In the music business, it's all by feel—what is necessary at the time.

MR: Have you produced a group?

SY: Yes, I produced a single with Chick Corea and I produced an album with Johnny Winter. I stay away from it until it's the right kind of a project. I mean, I do get a lot of offers, but it's so time consuming. It's the kind of thing that to be good in it, you have to really believe in it. My main concern is being a recording engineer and trying to help other people make their records.

MR: What has been the most interesting session you have ever done?

SY: There are a lot of them, but a unique one was Alice Cooper doing *Billion Dollar Babies*. We did it in their house, with a remote truck parked in the driveway. We used the truck as a control room and used TV monitors in the truck to see the band. To have it come out really good was a challenge. Also, doing John Lennon's *Walls and Bridges* album.

MR: What about the Lennon session sticks out in your mind?

SY: Well, he's a special person. I don't mean a special person in terms of publicity, but simply because he's special. He's a witty guy, he's a sensitive guy, he's wonderful and he wouldn't hurt a fly.

MR: Who was the easiest artist to work with?

SY: Chick Corea. He was the easiest because he knows what he wants, and he just needs someone to get it down.

MR: Is it important for a group to come in and know what they want?

SY: It costs them a lot less money when they know what they want as far as how things should sound. Although it's really not so much the sound as it is the concept. In other words, maybe it's just that Chick's music lends itself to a concept. He knows "well this part belongs here, and this has to play with this and it has to make a certain sound." A lot of rock and roll is built from nothing. It's bass and drums and then they start adding to it. That's the slowest process, adding a piece at a time. It's very time consuming and it's very difficult sometimes to make things fit. If they don't have a concept to start with, it's a lot of work and a lot of intense listening. That gets expensive for them, whereas Chick is so rehearsed that he has all his parts down. He just sits down and we take one day, and we get all the sounds in that one day. We don't record it till the end of the day. We just put something down so we can say we put something on tape. Then we hear it back and it gives me a day to think about what I might want to change. Then the next two days we do all the tracks. Frequently, he accepts the first or second take.

MR: What are your feelings on female engineers? Are there enough of them?

SY: No, too few.

MR: Why is that?

SY: Well, we find that it's very hard for them to deal with the men. What does a woman do at 3 a.m., when she's alone with the producer and he starts coming on to her?

MR: Is there a female engineer currently at The Record Plant?

SY: No.

MR: Would you like to see it happen soon?

SY: I would love to see a female engineer here. They hear differently than a man. They mix differently than a man does. Women mix with less highs, a warmer sound—more like a "womanly" sound. The right woman for the Record Plant hasn't yet come along.

MR: How would you hire a woman for the Record Plant?

SY: First of all, you have to get past the girls that are the groupies. You have to find a girl who really wants the job. They are hard to find because there are many more men that want the job; there are only a handful of women that want engineering jobs.

MR: How do you train your engineers when you first hire them?

SY: Their personality is just as important as their training is. The person who I would hire has to be relaxed, not a pushy kind of a person. The person has to be like a good mix. Pushy in certain ways so they can get ahead, but who genuinely wants to be an engineer.

MR: What type of advice would you give to the person who has dreams of becoming a recording engineer.

SY: Don't.

MR: Seriously Shelly.

SY: My advice to them is to realize that you really don't have to know anything to start. The pay is terrible in the beginning, and you have no social life. If you are living with your family it makes it easier only because the money is so bad. I mean, it's only like $2.50 an hour to start, so if you are married it's almost an impossibility. You have to really want it, and you have got to be out of your mind. Those are the main qualifications. I could never have done this if I had been married when I got into the business. It really takes a lot out of a person.

MR: Someone who is working in the tape department at Record Plant won't be there forever. Right?

SY: He'll be out in six months and he'll be on to something else—*if* he's got what it takes. We spend lots of time with people and we train them. I've seen kids I've hired start to become engineers and it's a thrill for me because some of them were terrible in the beginning. Just like me, I was terrible. I was the worst, but I guess I had something in me that allowed me to learn why I was bad and to eventually get better. Some people stay bad. I was bad, but Roy Cicala believed in me. A lot of people want to work here, but if there are no openings, I tell them to get a job in any studio; you'll learn.

MR: They're all [the studios] basically the same?

SY: Yes, except in the way they are run. That's not all the same.

Our amp had a few rough critics to please. Moog. Gibson. Les Paul. Ronnie Montrose. Our oscilloscopes. And you.

Nobody knows everything there is to know about amplifiers. The Lab Series engineers were ready to admit it... and do something about it. We went to several outstanding audio experts. And they helped us put more good ideas into the Lab Series than any one company could ever think of.

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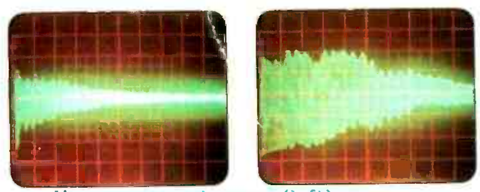
As for musical electronics, there's no better place to go than Moog.® So that's where we went. And we studied with the people who have an ingenious reputation for putting electronics under the musician's control.

Then we put the Lab Series amplifiers through some heavy studio sessions. And set out to please some heavy artists, like Les Paul and Ronnie Montrose.

Next, we took it to the streets to talk to our most important critics—the people we really built the Lab Series for—working musicians.

And for two years, we kept asking for criticism, until all we got were compliments.

Advancing an amplifier takes advanced technology—something that today's amplifiers haven't really enjoyed. For example, the Lab Series engineers applied new advancements in high voltage transistor technology that let us overcome the problems of low voltage transistors. And that enabled us to create the exact sound our musician advisors were after. But technology has a lot more to offer. After all, if Moog can engineer an orchestra of sounds into a small, 20 lb. keyboard, we felt an amplifier should offer more control over its voicing. Our new circuits overcame basic amp shortcomings. For example, if you like the overtones you get when you overdrive your amp, chances are you like your sound at high volume levels. But if the room is small, you're either too loud, or too lackluster. The Lab Series Compressor lets you sound great at any volume level, so you can turn on intense overdriving distortion without overpowering the room. You can even equalize your sound to help overcome poor room acoustics.



Uncompressed sound (left) vs. Compressed sound. Note difference in loudness and time relationships.

What's more, we engineered several outstanding sounds into the Lab Series. The Frequency and Midrange controls let you tune into different frequency bands and tone colors the same way you tune in stations on your radio. It also lets you overdrive sections of a chord, so you can create new chord colors that really expand your rhythm horizons.

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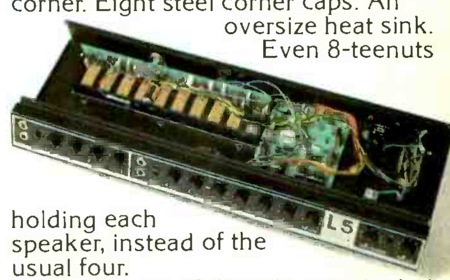

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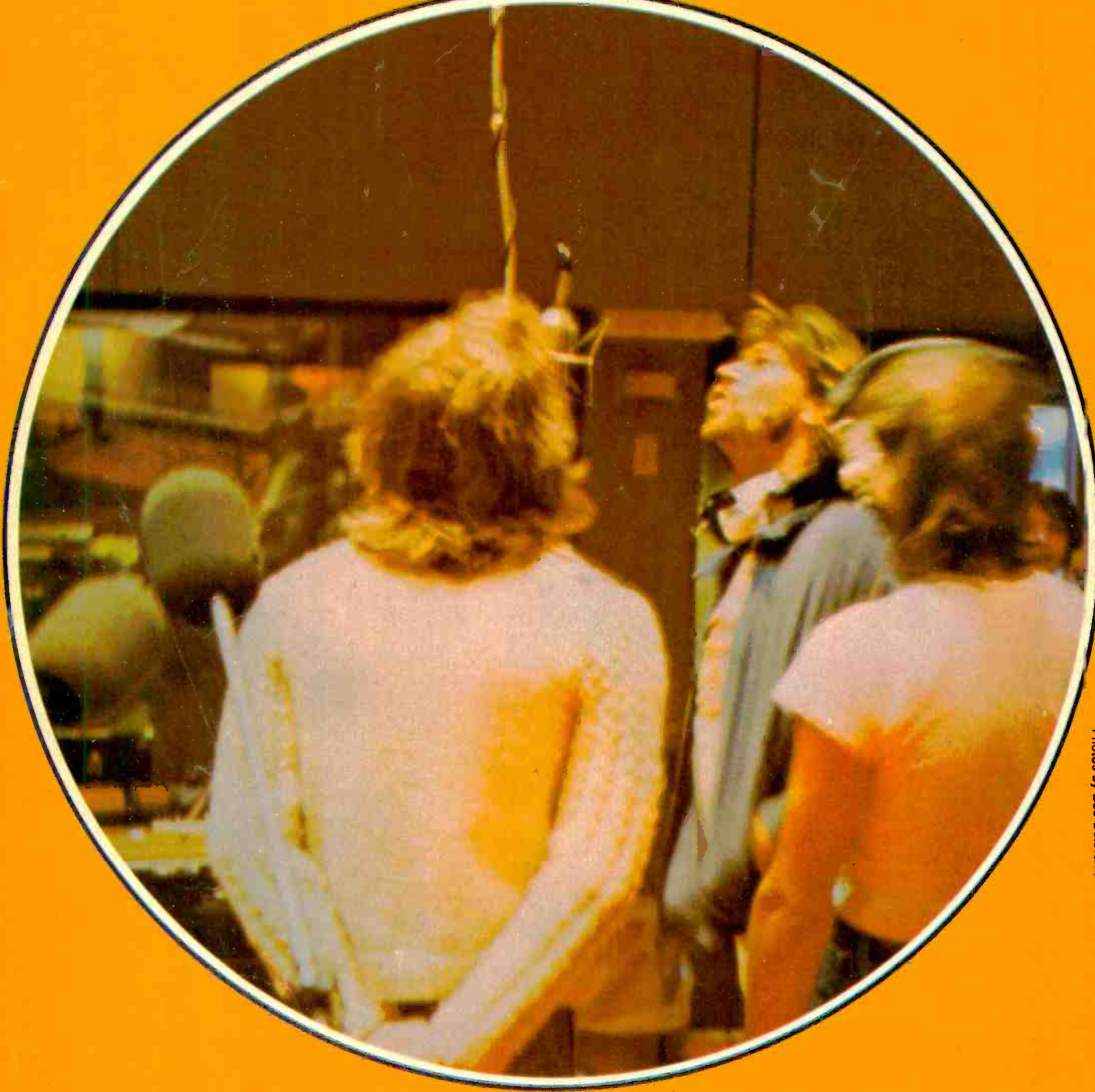
CIRCLE 52 ON READER SERVICE CARD

A Session with the Bee Gees

By John Linde



Photos by Bob Delstonie

Recently, the author of this piece, John Linde, spent time with the Bee Gees while they were at Criteria recording studios in Miami, Florida. The Bee Gees were involved in a project which entailed recording four songs—"How Deep Is Your Love," "Night Fever," "Stayin' Alive" and "If I Can't Have You"—for an upcoming ABC-TV special starring John Travolta of "Welcome Back Kotter" fame.

For me, it's a high to be touched by the kind of music the Bee Gees originate and produce, but it's an even greater high to be touched personally by them.

It doesn't take long to realize the Bee Gees and company are a highly professional but yet informal musical family. Ideas and opinions are everywhere. Three hours would go into the six-note vocal intro on "How Deep is Your Love." When it was finished the sound was the reward. When all twenty-four tracks came up, I got a rush.

On the Other Side

The production team of Karl Richardson and Albhy Galuten—corporately named KARLBHY Productions—is very impressive. Karl has been at Criteria studios for eight years. He helped wire it together, and has engineered there for a number of top producers. Karl is highly technical, he's the pilot. The nice thing about taking off with Karl is that you always know your going to land. Karl plays the board like an accomplished musician, faders, EQ, echo—his performance is outstanding.

Albhy Galuten is the ear *extremely musical.* He can read music, write it, sharp it, flat it and hum "Nights On Broadway" *backwards.*

The thing that makes Karl and Albhy so appealing is their lack of ego. Like the Bee Gees, they are *real*. That's the magic that seems to dominate, that down to earth concern and love for what they are doing. The Bee Gees, Criteria, Karl and Albhy came together thru guitarist Eric Clapton. Eric had recorded "I Shot the Sheriff" at Criteria and was so pleased he mentioned it to Barry Gibbs. At that time the Bee Gees needed to be stimulated, and Miami and Criteria fulfilled that need. "It blew my mind," stated Barry, "the creative energy started flowing, and when that happens you realize there's no bottom to the barrel. All you have to do is get your head right, and reach down into the barrel; you'll keep coming up with good ideas." *After a two year drought, ideas were raining on the Bee Gees.*

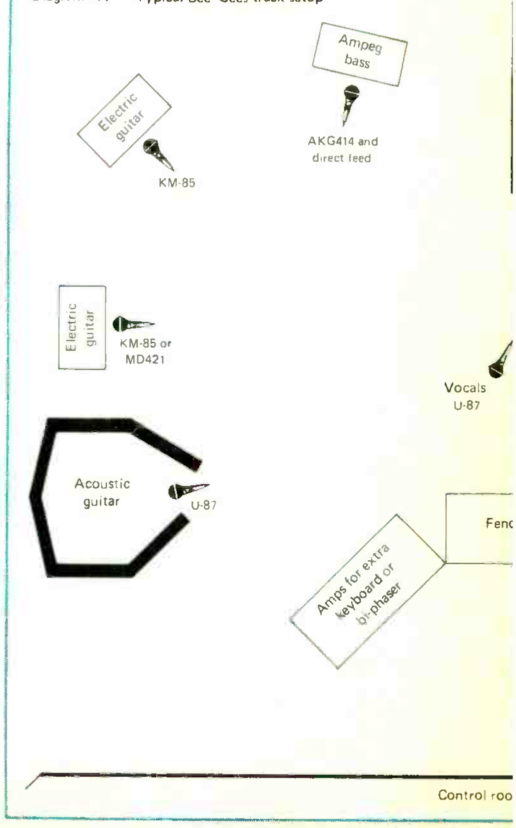
Writing and Rehearsing

The Brothers Gibbs put a lot of time and thought into melodies; writing is an art only a fool would try to explain. Robin, Barry and Maurice are master craftsmen. Nevertheless, even master craftsmen must rehearse, and its during rehearsels that all the writing, vocal talent and instrumentation are put together.

Rehearsals are guidelines, they are never detailed, concrete blueprints. Creative juices will be flowing and there will be change, constant change. Ideas will become playbacks; the ideas that work will get printed, the ones that don't get exiled into the idea graveyard by means of the erase head. Try the idea, if it doesn't work try another and other.

This method of recording/rehearsing is the luxury of being successful (it's expensive). Since most record manufacturers work within a budget, a lot of producers must sacrifice creativity to please the clock and the accounting

Diagram "A" – Typical Bee-Gees track setup



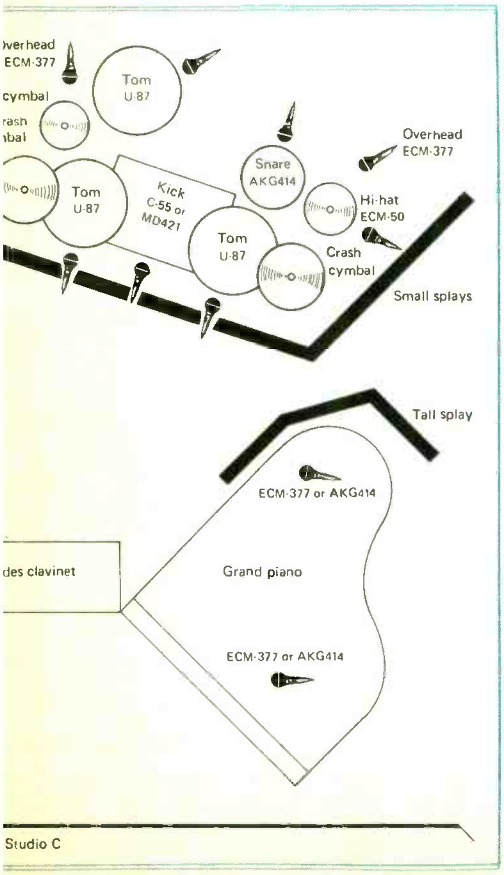
department, and that very often becomes musical assassination. I mention this because a lesson can be learned from the Bee Gees. Not that every artist who obtains a recording contract should be given an open checkbook, it's just that no one can regulate the mind to successfully create. (There is no ON/OFF switch.) If one enters the studio and the creative juices flow, well, then do it. If the juice doesn't flow, get the hell out.

But above all, be honest. Although the clock was never mentioned by the Bee Gees, it was never abused either. Every tick brought an idea.

Drum Tracks

Since the drum tracks must be a song's foundation, the Bee Gees record them in a manner which insures them that the tracks will be solid. First they record the drums with accompanying instruments, continuing until the drums are "right." Then, all the tracks except for the drums are erased.

Drums and drummers are very physical—hard to record and affected by everything. Bee Gees drummer Dennis Byron will set up the drums his way (the way he is most comfortable), and then Karl Richardson will do his thing. Acoustic baffles are used to surround the drum set, eight mics will be positioned— a C55 or 421 is used on the bass drum, three U-87s will be placed over the toms, one ECM 50 on the hi- hat, a 414 on the snare drum and two ECM 377s for overheads. The Bee Gees do not have a standard or predetermined drum sound, it varies with the particular musical composition. It may call for the bass drum head to be padded, unpadded or tightened to where it nearly breaks. This particular session the drums had very little padding, but there are times when the bass drum is filled with pillows. Again, it's the business of sound, and electronics can do just so much (especially with drums). The action a drummer uses has a lot to do with the sound the drums create. If you know what you want to hear your half way home. The Bee Gees, Karl and Albhy always seem to know what they want to hear. The other half of the journey is the performance, when sound and performance come together it's on to the bass.



Adding Tracks

Maurice Gibbs plays a Rickenbacker and a Fender bass, usually thru an Ampeg bass amp. He is given time to work with the drum track, going over the parts he's rehearsed and trying new ones. Everyone tunes into the production. They criticize, comment, applaud and omit (a sort of process of elimination). Ideas and musical compliments pass around the control room —it's no time to be defensive. Maurice plays the part he feels, accenting the *groove*, not himself; this isn't solo time. That's the sign of a true professional and Maurice is just that. The bass must compliment the drums and never take away or resist the mood. Maurice stands in the control room when he records, his line cord travels out the door, then some twenty feet to the amp. He likes being where all the ideas are tested.

Patience is the secret ingredient that's incorporated into all Bee Gee productions. No one keeps score, it doesn't matter how many times you criticize or applaud, just be honest and say what you feel—not what you think someone wants to hear.

Barry Gibbs and Allen Kendall do the guitar work—electric and acoustic —for the Bee Gees; today it would be Allen's turn.

The process of recording repeats

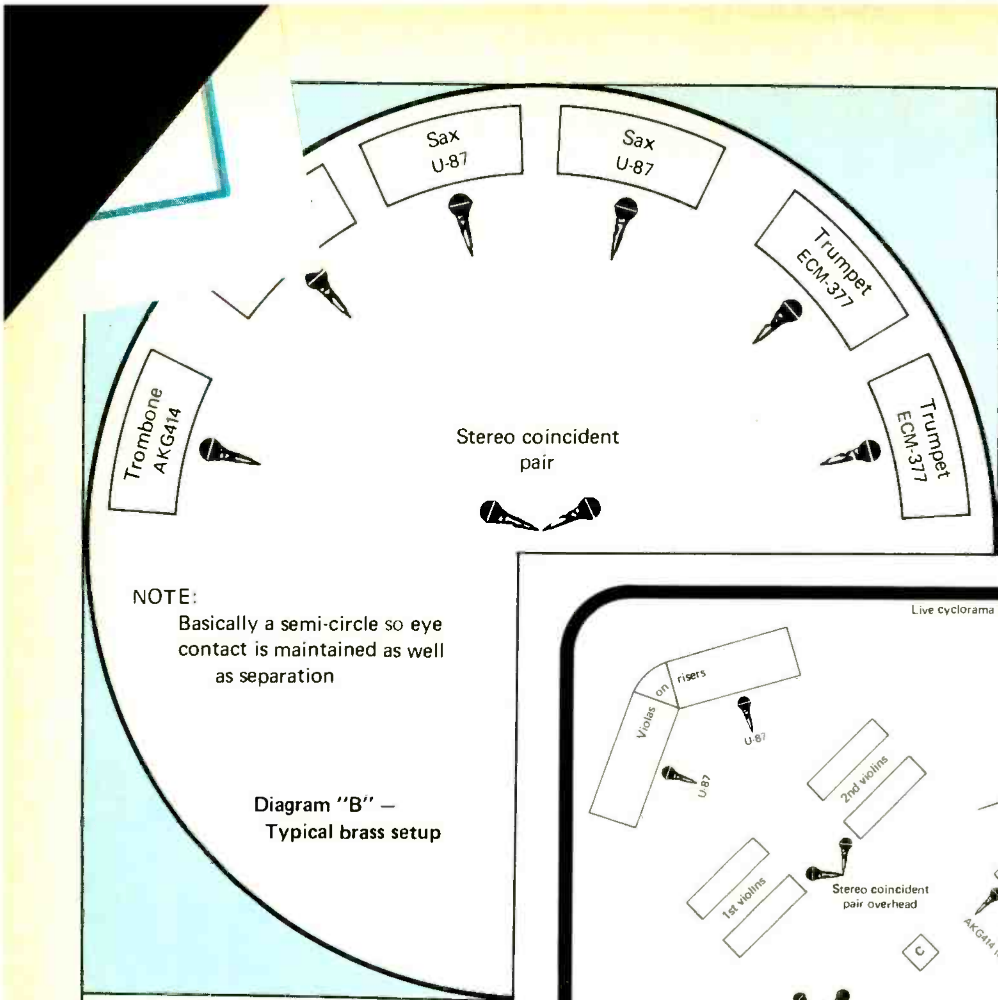


Diagram "B" — Typical brass setup

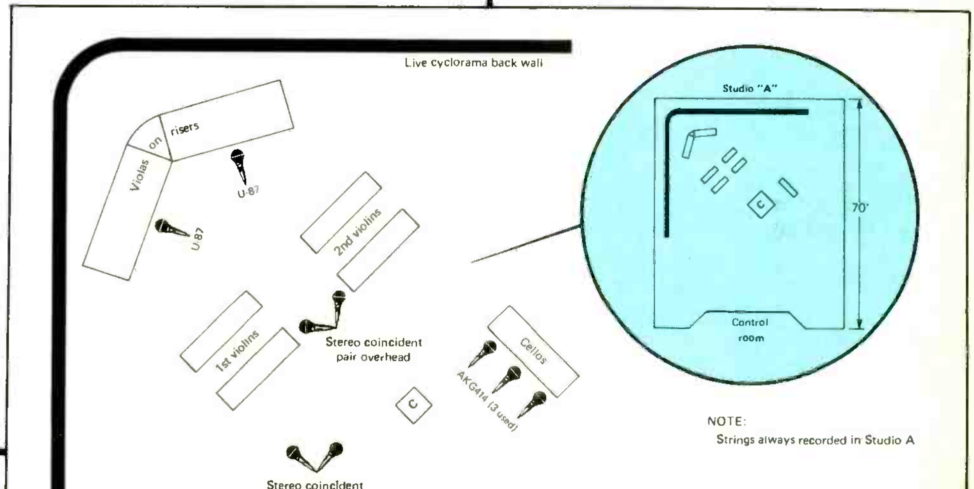
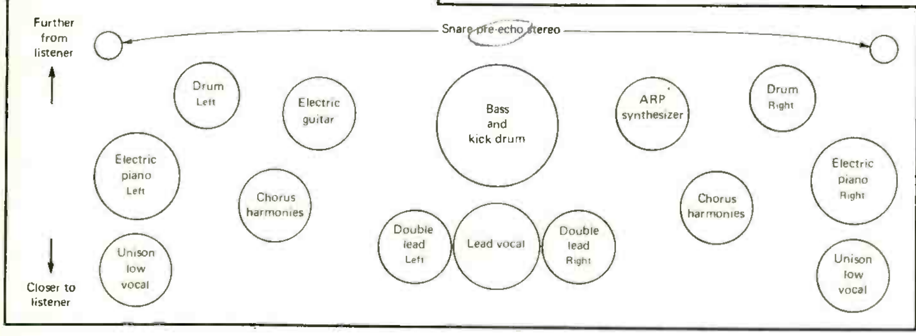


Diagram "C" — Typical strings setup

Diagram "D" — Basic panning on "HOW DEEP IS YOUR LOVE" "stereo setup"



itself, comment after comment, idea after idea—it's instant replay of the past few hours. Allen will play what he feels, listen to suggestions and brew all the control room ad libs. The end result will always be what sounds best.

The Technical Art

It's time to clear your head. As everyone sits in the outer waiting room, the conversation turns technical. All tunes on this session are recorded 30 ips, 24-track with Dolby at a + 3 level on Ampex 456 tape. All tape machines used at Criteria are MCIs. Karl states, "We usually use three echo chambers or more for each mix, consisting of one or two 'live' chambers, plus a stereo EMT and a stereo AKG. Some form of echo delay before the chamber—7½, 15½, DDL or whatever—is usually mixed in with the vocal echo. Strings and brass are usually 'live' chamber echoed. The Bee Gees EQ and echo onto the master tape if a particular sound is wanted, or to eliminate unwanted sounds. Normally, post EQ in the mix has been found to be the safer method because radical changes are sometimes made sonically if the recording has changed substantially due to new overdubs, or if, since the record is basically finished, we hear it differently."

This technical chatter ends when I'm told some not so secret secrets by Maurice Gibbs about guitar tuning. If you're interested in how the Bee Gees get that fantastic rhythm sound, try this—three acoustic guitars, guitar #1 tune conventionally, guitar #2 open-cord tuning, guitar #3 tune the first and second strings conventionally, and the third thru sixth one octave higher.

Vocal Magic

Recording studios are like sodium pentothal. Sure electronics, gimmicks, echo and all that good stuff can help improve a bad sound. They can build courage, but if the weak have any degree of intelligence they will learn from every recording experience. I mention this because lets face it, the studio puts talent to the test more so then any other medium of exposure. Don't ever, even for a moment, think that it is easy. I don't care how much electricity is being pumped into this studio, the reason I'm sitting here with my mouth open is because I've been *reached.* The Bee Gees are that perfect; they have the talent, the uniqueness, the personality. A producer's dream, and if you have spent any amount of time in a recording studio you know what I mean. It's difficult to write about recording techniques when Robin, Barry and Maurice enter the studio to work their vocal magic. The Bee Gees are professional; there is no such thing as a bad take, the good ones just get better. It's as complicated as that. The vocal overdubbing experience leaves me realizing that the Bee Gees deserve the spotlight, the recognition, the fame and anything else that goes along with the ability to make people forget everything else and just enjoy!

ANNOUNCING: THE ROLAND GUITAR SYNTHESIZER

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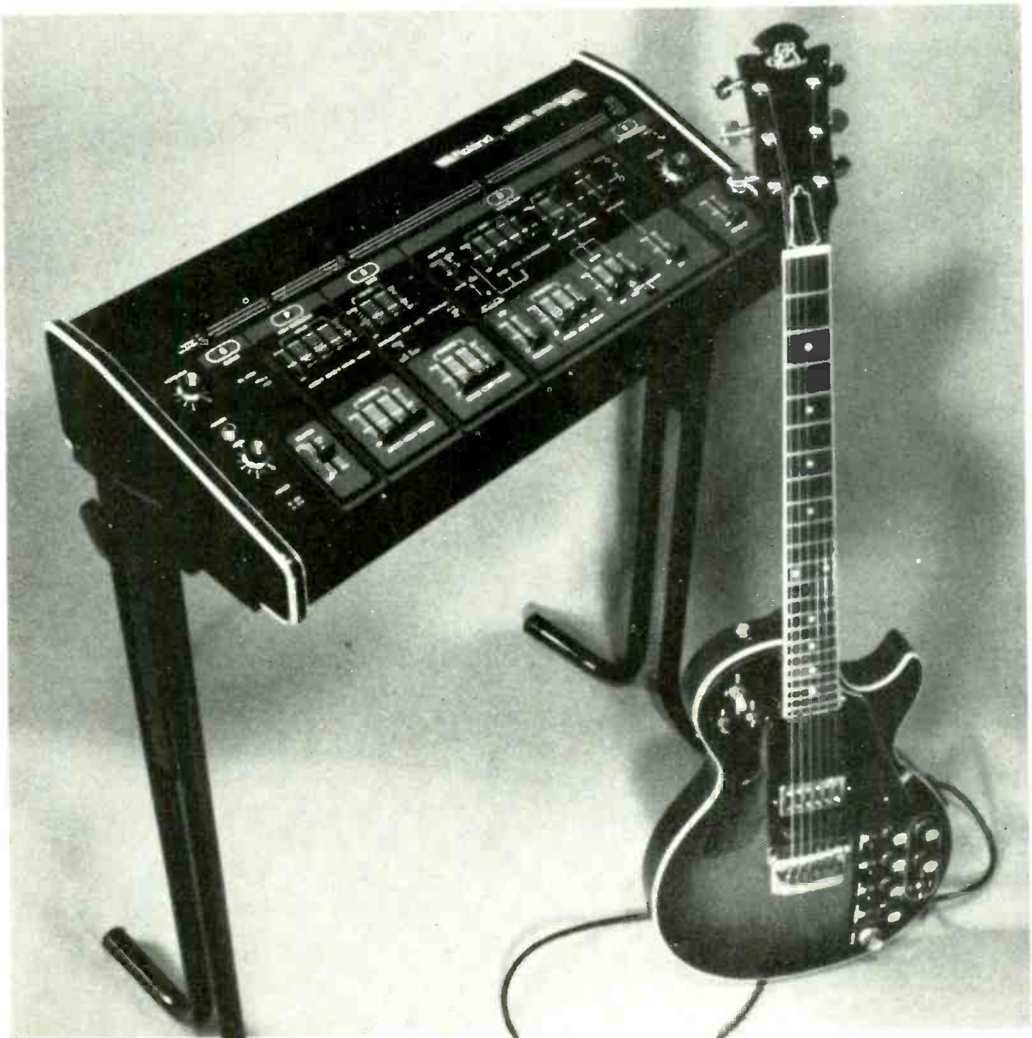

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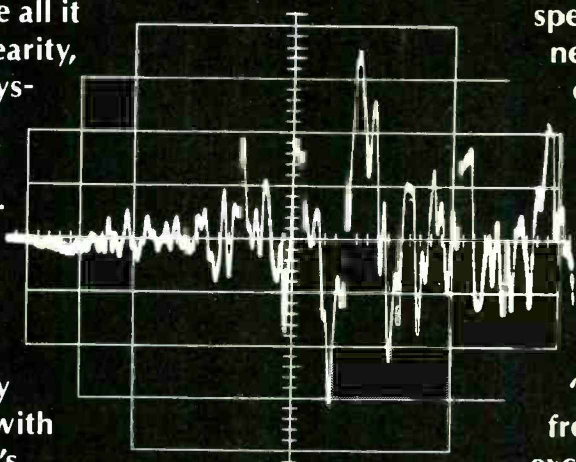
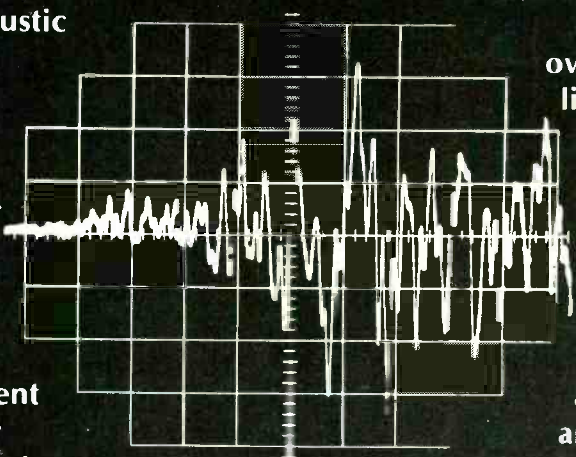
CIRCLE 92 ON READER SERVICE CARD

Technics knows there's more to Linear Phase than staggered speakers.

If staggered speakers were all it took to achieve phase linearity, other staggered speaker systems would sound like ours. But Technics knows it takes more. Much more. Like a phase-controlled crossover network that takes into account the phase characteristics of each driver. Like extremely wide-range drivers, each with a frequency response that's as flat as it is wide. And finally, aligning the acoustic center of each for the optimum acoustic position.

But just as important, Technics knows that to achieve phase linearity as well as a wide and flat frequency response is also to achieve the ultimate in high fidelity: waveform fidelity. With it the output waveform of any component or speaker will be a mirror image of the waveform put into it. And that sounds better than good. It sounds live.


Live Piano Waveform.


Piano Waveform Reproduced by SB-7000A.

And if seeing is believing, look at the waveforms. On top is the oscilloscope reading (the fingerprint) of a live piano waveform. The other, the piano as reproduced by Technics Linear Phase SB-7000A. That's waveform fidelity you can see, as well as hear.

How did we do it? By designing a crossover network that would provide an overall linear phase characteristic for the entire speaker system, while simultaneously compensating for the different acoustic pressures of the individual drivers.

When we finished we ended up with a unique phase-controlled crossover network consisting of 6 dB and 18 dB/octave cut-off slopes. It not only eliminates "audible dip" at the crossover frequencies, but also assures excellent localization of the original sound source within the acoustic field.

But as important as the crossover network is in achieving linear phase, so are the individual driver units. That's why we designed and manufactured the speaker drivers with the flattest amplitude, widest frequency response and lowest distortion possible. A goal we achieved only after exhaustive amplitude and phase studies in anechoic chambers.

Our final step was aligning the acoustic center of each driver in precisely the same vertical plane. But it took more than anechoic chambers. Technics had to develop a new time-delay system using BBD (Bucket Brigade Device).

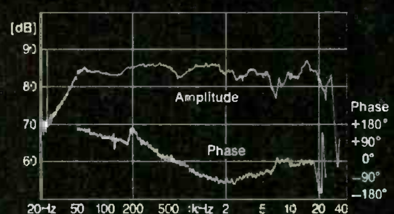
Only then could we locate the optimum acoustic position for each driver. In addition, each unit is positioned vertically for the best horizontal dispersion and then spaced as

Much more.

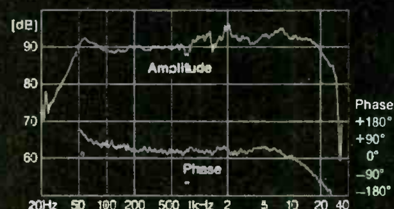
closely as possible for the best vertical dispersion of all audio frequencies. What's more, after alignment each unit is fine-tuned to assure precise linearity.

The result, with the SB-7000A for example, is an overall phase response, linear between 0° ±45° between 100Hz and 15kHz. A figure that's virtually flat and definitely unsurpassed by any other multi-range speaker system.

As the graphs prove, even staggered speaker systems with seemingly "linear phase" characteristics show moderate to



Staggered Speaker Configuration.



Technics Linear Phase SB-7000A.

severe phase shifts at different frequencies. But as you can see, the Technics SB-7000A has an unprecedented flat amplitude/frequency response and linear phase response.

But we don't expect you to buy any speaker system based on how good it sounds on paper. Audition the world's most linear phase speaker systems: the Technics SB-7000A, SB-6000A, SB-5000A. You'll find out just how much more there is to Technics Linear Phase than staggered speakers.

Technics
by Panasonic

CIRCLE 26 ON READER SERVICE CARD

Ambient Sound

BY LEN FELDMAN

Bi-Amping, Tri-Amping and Such

I can remember way back to the time when I read my first book about high-fidelity sound. While I can't remember who wrote the book, I do recall that in the chapter devoted to loudspeaker systems, the need for separate woofers, tweeters and mid-range drivers in high-quality speaker systems was explained by offering an analogy to human singing voices. The writer pointed out that just as no one human voice can reproduce the entire range of vocal music, so too, is it necessary to employ different transducers in a speaker system, each of which "specializes" in a given range of frequencies and "does its best" when reproducing only that limited range.

It wasn't until years later that I realized that the analogy doesn't quite hold up completely. After all, when you talk about singers handling different vocal passages in different ranges, you are talking about totally separate "sound producers." These sound producers, while occupying the same stage, nevertheless create totally independent sounds which do not interact with each other and do not in any way affect sound reproduced by their singing companions.

When considering multi-driver loudspeakers, on the other hand, those loudspeaker drivers, in most cases, are interconnected to each other by a common element —the so-called *crossover network*. Crossover networks in multi-driver systems may be anything from a simple capacitor, connected in series with the tweeter of an inexpensive two-way loudspeaker system, to elaborate inductance-capacitance-resistance networks designed as legitimate low-pass, high-pass, or band-pass multipole filters. In any case, it should be recognized that regardless of the crossover network's design, a certain amount of electrical interaction between drivers takes place, since no realizable filter can be designed to completely cut off one driver at a single frequency while the next driver suddenly turns on at precisely the same frequency. Most passive crossover networks are designed to have cut-off (or turn-on) slopes of either 6 dB/octave, 12 dB per octave or, in some cases, 18 DB per octave. If one examines only the frequency response characteristics of a typical crossover network, one can easily come up with a pair of curves which, at their intersection point (the crossover point), are 3 dB "down" from the flat response which they present in their pass-band region. Theoretically, then, each driver is putting out 3 dB less power and the sum of the energy from the two drivers should equal that of either driver operating within its pass-band, so that uniform energy propogation takes place at the cross-over point. Unfortunately, several other factors enter the picture.

A 6 dB/octave crossover network which offers good response and phase characteristics in the region of crossover, suffers because its slope is so gradual that the two drivers associated with it must be capable of having good response for a full four octaves beyond the so-called crossover point—thereby defeating the idea of having each driver perform only within its optimum frequency range. 12 dB/octave crossover networks, the type most commonly found in popular multi-driver systems, present out-of-phase conditions at the crossover frequency and this out-of-phase situation has, in recent years, been recognized as being audible, especially where rapid transients in music material are concerned. 18 dB per octave passive crossover networks in loudspeakers do offer a flat response characteristic in the region of the crossover frequency, but transient response is, again, poor because of phase distortion over a wide frequency range to either side of the crossover frequency.

In addition to the phase and transient response problems cited, passive crossover networks offer other disadvantages. For one thing, in high-powered systems, they do dissipate a significant percentage of the power available to the drivers. Having to be designed to match the low output and input impedances of the amplifier and drivers which they interconnect, well-designed crossover networks must use heavy coils which are wound with large gauge copper wire. Capacitors which will have low dissipation, as well as these coils, can often be quite expensive, and component compromises are often made even in very expensive loudspeaker systems.

All of which leads us to the subject of this discussion—bi-amping and tri-amping. For those not familiar with this rather old approach to multi-driver speaker operation, we should explain that in a bi-amped or tri-amped audio system, each driver of the system is powered by a separate power amplifier. Thus, if you are going to set up a three-way loudspeaker system (woofer, mid-range and tweeter), three separate channels of audio amplifier power will be required. In addition, an electronic component known as an electronic crossover network must be connected between the program source (or the preamplifier) and the individual power amplifiers used to drive the individual speakers. The electronic crossover network fulfills the same function formerly performed by the passive crossover components located inside the speaker enclosure. It is nothing more than a series of low-pass, band-pass and high-pass filters. But there is a big difference. Because the electronic crossover network fits into the system at a low-voltage point in the signal chain and is called upon to dissipate virtually no *power*, actual components used can be quite small, and precision values of capacitance and inductance are more readily realized. In fact, some modern electronic crossover networks utilize no real inductors at all, but resort to active filter circuits which can duplicate the characteristics of discrete L-C networks without using actual coils.

In addition to providing true isolation between the drivers of the multi-way speaker system, the use of an electronic crossover network in a bi-amped or tri-amped system offers additional benefits. The interaction between drivers caused by the presence of the common-connected passive crossover network is eliminated , as are such audible effects as intermodulation distortion caused by such interaction. Furthermore, while the cost of the additional power amplifiers seems at first prohibitive, it should be pointed out that, in most cases, the power output capability of the individual amplifiers used in such multi-amp systems need not be the same for the woofer amp, mid-range amp and tweeter amps. As you are probably aware, energy distribution in music is not linear over the entire audible range. Extreme bass reproduction requires far more amplifier power than does mid-range reproduction and the energy content of "highs" in music is still lower. Thus, typically, while the woofers of a tri-amped system might require 100 watts of power for adequate reproduction, the mid-range amplifier might be rated at, say, only 40 or 50 watts, while the amplifier that drives the tweeter need have only 10 or 20 watts of power output capability. As a matter of fact, speaker system manufacturers have been aware of this fact for years and hardly ever install—in commercially available systems—tweeters with the same power handling capability as their woofers. (A fact which accounts for a lot of tweeter burnouts if your amplifier suddenly starts to oscillate at supersonic frequencies, or if you use it for the reproduction of electronic music, where the natural rules of frequency versus energy distribution no longer apply.)

Most available high-quality electronic crossover networks offer a wide choice of frequency crossover points (to suit the needs of the drivers that might be used in the speaker system) and more than a few even offer a choice of slopes at the crossover points, including the popular 6, 12 and 18 dB per octave values. This flexibility in crossover networks lends itself to a great deal of experimenting when you set up your first bi-amped or tri-amped system—without having to wind new coils and choose new low-loss capacitors every time you want to try a different crossover combination.

De-bunking an Old Tri-Amp Myth

One of the other arguments put forth in favor of bi-amped or tri-amped systems does *not*, unfortunately, turn out to be true—though the logic of the simple math involved once trapped me into writing it up as if it were valid. The argument runs something like this:

Suppose you were trying to reproduce a musical tone which consisted of a low, bass tone at 100 Hz and a mid-range tone of 3 kHz. Next, suppose that for accurate reproduction, the 100 Hz tone demands a power output of 100 watts while the 3 kHz tone requires a power output of 50 watts. In a bi-amped system, you would therefore need a 100-watt amp for the woofer and a 50-watt amp for the mid-range driver. The false argument proposes that the *voltage* equivalent at the speaker terminals (assuming 8 ohm loads) for the 100 watt component of the signal is 28.3 V RMS, while the voltage equivalent of the 50 watt 3 kHz signal is 20.0 volts (V = PxZ, where V is voltage, P is power and Z is impedance). The argument goes on to state that the voltage that would be required at the output of a *single* amplifier used to drive a speaker system containing a passive crossover network would therefore be the sum of the two voltages, or 48.3 volts. But the power output across 8 ohms that would be developed by 48.3 volts works out to be 291.6 watts! Thus, the argument goes, going to bi-amping means needing *less* total power (even though two separate amps are used) than is needed for single amplifier operation for the same level of sound reproduction. There's only one flaw in this argument, and that is that when we deal with complex music signals containing a multiplicity of frequencies, you cannot simple *add* RMS values of voltage together to arrive at a new RMS equivalent voltage. If you think about it for a moment, you will realize that *instantaneously* the two voltages may be additive while at other times they will subtract from each other. So much for the power you thought you might save by going to bi-amping or tri-amping. But, even discarding that long-held myth, separate drive for your woofer, mid-range and tweeter does offer great advantages, not the least of which are more accurate control of crossover-points, lack of electrical interaction between drivers and, most important of all, audibly cleaner and more faithful sound reproduction —all other things being equal.

MODERN RECORDING **LAB REPORT**

NORMAN EISENBERG AND LEN FELDMAN

DELTA-GRAPH MODEL EQ-10 GRAPHIC EQUALIZER

General Description: The EQ-10 from Delta-Graph Electronics of Seattle, Washington is a graphic equalizer system in kit form. The basic unit is the EQ-10M module, a single-channel, ten-band equalizer. Additional modules may be added as required. The power supply is another module, PS-4, which can power up to four EQ-10 modules. Various combinations are available (see prices below), as well as a cabinet and a tie-plate for rack-mounting.

The simplest format would be a single EQ-10M and its PS-4 power supply which, when wired by the kit-builder, are linked electrically by three wires. The equalizer module is enclosed in metal, although the leads from the power supply, plus the in and out signal cables are exposed on a terminal strip at its rear. The power supply itself sits exposed on a small chassis.

The equalizer module contains only the ten sliders which are marked from −15 through 0 to +15. Nominal center frequencies are 31.2, 62.5, 125, 250, 500, 1000, 2000, 4000, 8000 and 16000 Hz. The "O" marking, or flat position, for each slider has a detent, or "click-stop."

For a stereo equalizer, two EQ-10M modules and one PS-4 power supply would be required; for a four-channel equalizer, four EQ-10M modules and one power supply are required; for an 8-channel equalizer, eight 10M modules plus two power supplies would be required. The available cabinet will handle a two-channel version.

Test Results: While relatively Spartan in appearance, and obviously directed to the inveterate (and knowledgeable) do-it-yourselfer, the EQ-10 proved to offer excellent performance of a caliber that lends itself to professional as well as to home-audiophile applications. Not counting the time required to wire and assemble the device, its cost would be well below that of equivalent performing equipment in a completely-assembled form.

In MR's tests, the EQ-10 met or exceeded its published specifications. Response, with all sliders detented, was confirmed as within 0.5 dB from 20 Hz

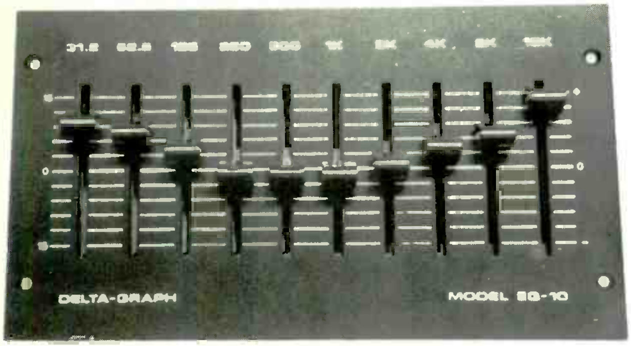


Fig. 1: Delta-Graph EQ-10: Completed EQ-10-M module.

to 20 kHz. Hum and noise were 90 dB below a 2-volt output level; harmonic distortion remained below the rated 0.05%, and IM remained lower than the rated 0.0075%. To supplement instrument measurements, MR made its customary series of sweep tests using our spectrum analyzer. Fig. 4 represents the response for three different conditions of the ten controls. The upper trace is system response from 20 Hz to 20 kHz with all sliders in maximum boost position. The lower trace is overall response with all sliders set to maximum cut. The middle trace, representing a sine wave, is actually a response curve produced by placing alternate octave controls to maximum and minimum (e.g., 31.25 Hz at maximum boost; 62.5 Hz at maximum cut, etc.). The results are impressive in that they show the lack of interaction of adjacent controls, a characteris-

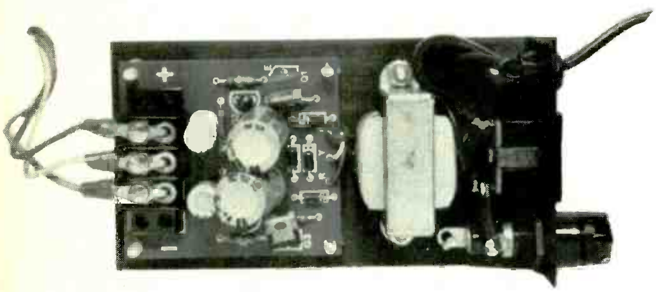


Fig. 2: Delta-Graph EQ-10: Separate PS-4 power supply can handle up to four equalizer modules.

tic not always managed by less sophisticated equalizers in which opposite settings of adjacent controls tend to neutralize or cancel each other out. What this indicates, simply, is that the EQ-10 can provide a fine degree of control to permit tailoring overall response to specific needs of program material and/or acoustic environment.

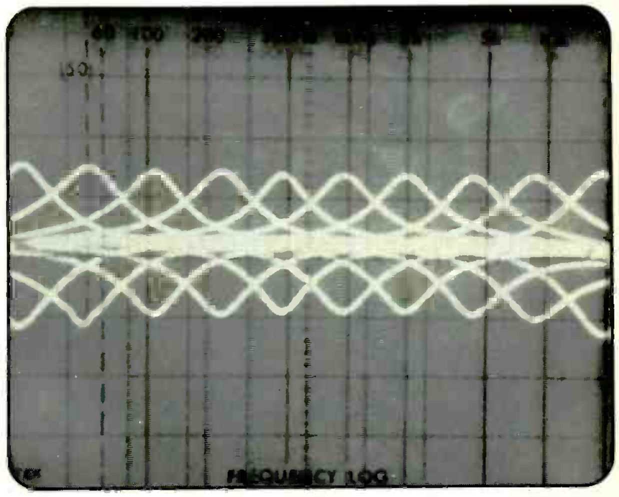


Fig. 3: Delta-Graph EQ-10: Range of boost and cut for each of the ten octave-band controls on a Delta-Graph equalizer channel.

General Info: EQ-10M module for one channel—dimensions: front-panel mounting flange, 5¼ high by 9½ inches wide (standard EIA and RETMA "half-

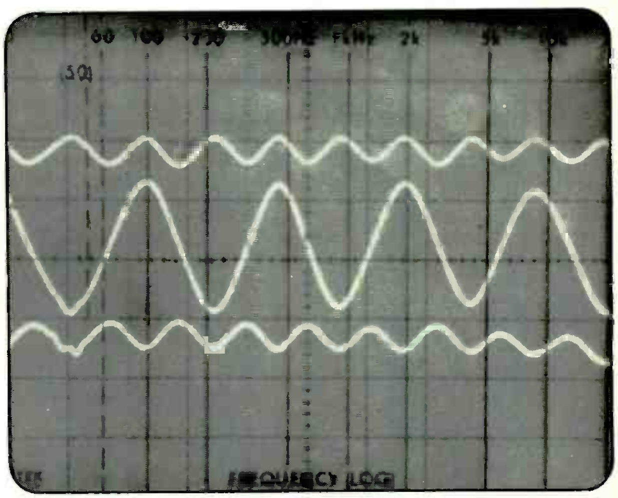


Fig. 4: Delta-Graph EQ-10: Upper trace shows response with all controls set at maximum boost; lower trace is response with controls set at maximum cut; middle trace is response with alternate controls at maximum cut and boost.

rack"); depth behind mounting surface, 2 inches; price, $65. PS-4 power supply module—3 by 6½ by 2 inches; price, $25. EQ-10SP (two EQ-10M modules plus one PS-4 power supply), $150. EQ-10QP (four EQ-10M modules plus one PS-4 power supply), $275. EQ108-2P (eight EQ-10M modules plus two PS-4 power supplies), $540. EQ-10WC cabinet (two-channel), $30. RA-2 (tie plate for rack mounting), $6.50.

Individual Comment by N.E.: The instructions and general presentation of this product indicate it is aimed primarily at the seasoned technician and not intended for the first-time or novice kit-builder. And it is certainly not a unit for the "out-in-front-only" control man. Not only does it present a genuine electrical hazard (which the manufacturer is careful to call attention to in the manual), but many of its possible applications require a do-it-yourself skill and inclination that may be beyond the scope of many potential users regardless of how deeply involved they are with sound and related equipment. While the finished unit does indeed perform as claimed, it might be well for the potential buyer to look through the manual beforehand to see what he or she may be getting involved with before the device can be put to work.

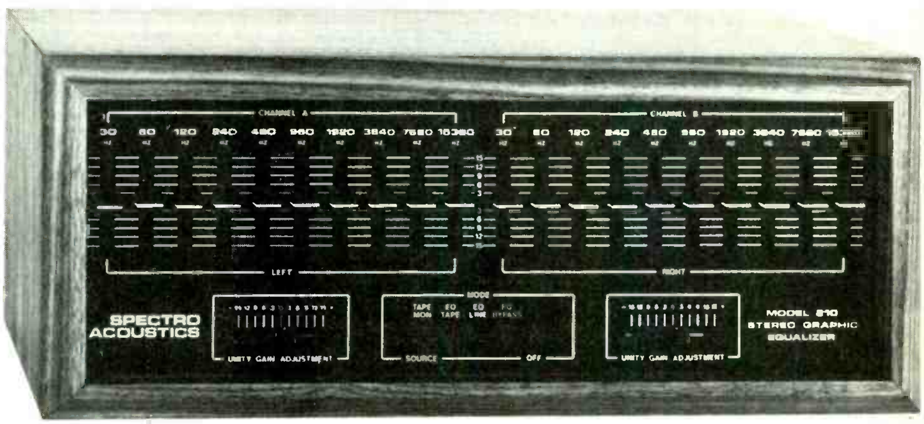
Individual Comment by L.F.: As our test results and vital statistics show, the EQ-10 does an excellent job of equalizing, and at a relatively low price vis-a-vis factory-assembled models. But I would urge the manufacturer to come up with a more professionally written and produced assembly and owner's manual. The small type (actually reduced from typewritten manuscript) is difficult to read, and the format of the assembly instructions presumes a fair amount of electronic wiring and assembly knowledge on the part of the buyer. If one of Delta-Graph's objectives is to enable non-professionals to own an equalizer that is as good as those used by professionals, the company ought to be more aware of the realistic capabilities of such possible buyers.

DELTA-GRAPH EQ-10 GRAPHIC EQUALIZER: Vital Statistics

PERFORMANCE CHARACTERISTIC	LAB MEASUREMENT
Range, per band	±12 dB, controls operated individually, all others at flat. ±15 dB, controls operated in concert.
Frequency response, sliders on flat	±0.25 dB, 20 Hz to 20 kHz.
Band center frequencies	As indicated on panel, all within 5%
THD, controls flat	0.01% at 1 kHz; 0.015% at 20 kHz; 0.03% at 20 Hz.
THD, any control setting	0.01% at 1 kHz; 0.024% at 20 kHz; 0.1% at 20 Hz.
IM distortion, controls flat	0.003%
IM distortion, any control setting	0.006%
Hum and noise (below 1-volt rms input, 10 kHz bandwidth, 600-ohm source, any control setting	90 dB down.
Rated output	2 volts rms (used for MR measurement)
Maximum output	12 volts rms
Output impedance	300 ohms, resistive, direct-coupled. 600 ohms balanced, short-circuit proof.
Input impedance	68 k ohms single-ended; 600 ohms to 100 k ohms balanced.
Input balance	Balanced line common-mode rejection greater than 40 dB at 60 Hz untrimmed; trimmable to greater than 80 dB at 60 Hz.
Output balance	Phase difference between outputs 180° amplitude match within 0.2 dB.
Power consumption	+15 VDC at 40 mA nominal per module; ±18 VDC at 50 mA maximum. (Higher voltage supplies usable with external current-limiting resistors added.)

CIRCLE 10 ON READER SERVICE CARD

SPECTRO ACOUSTICS MODEL 210 STEREO GRAPHIC EQUALIZER



General Description: The model 210 from Spectro Acoustics Inc. of Richland, Washington is a stereo (two-channel) graphic equalizer offering ten octave-bands per channel of equalization. Nominal center frequencies are 30, 60, 120, 240, 480, 960, 1920, 3840, 7680 and 15,360 Hz. The usual up-and-down slider arrangement is used, with the vertical ranges marked from −15 through 0 to +15. This control array occupies most of the front panel. Below the sliders are unity gain adjustments (horizontal sliders) for each channel, similarly marked from -15 through 0 to +15. Centered between these controls is a group of five pushbuttons: tape monitor/source; EQ tape; EQ line; EQ bypass; and the unit's AC power switch. The front panel also has a pilot LED that shows power on.

Connections at the rear—phono pin-jacks—provide for signal hookups, with stereo pairs for main outputs, tape inputs, tape outputs and main inputs. The rear also contains the unit's AC line cord plus an unswitched AC convenience outlet rated for up to 500 watts maximum. The equalizer's line cord is fitted with a two-prong polarized plug which will fit certain wall sockets, or heavy-duty extension sockets, but will

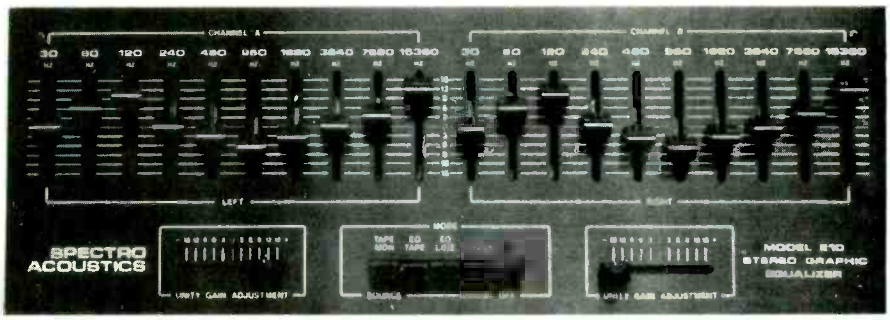


Fig. 1: Spectro Acoustics 210: Front panel view.

not fit many others including most of the convenience outlets customarily supplied on hi-fi components.

The model 210 is fairly light in weight and comes in two housings, one of which is suitable for rack-mounting (model 210R). The owner's manual is very good, explicitly written, adequately illustrated and contains much valuable information on EQ generally and on the uses of the device specifically. The model

Fig. 2: Spectro Acoustics 210: Rear panel view.

210's preferred hookup is in the tape-monitor loop of a sound system. However, alternate interfacing methods may be used, and are described in the owner's manual. In addition, the company welcomes inquiries for advice or help on additional applications. Briefly, the pushbuttons may be used for the following functions: the tape monitor button handles the standard tape-monitor circuit for monitoring the actual recorded signal as it is being recorded. It also is used for playback of a recorded tape. The EQ tape button inserts the equalizer into the taping path to enable the user to record pre-equalized tapes. The EQ line button puts the equalizer into the playback (or monitor) line without affecting tape recordings made while in this mode. The EQ bypass button bypasses the EQ circuitry and also the unity-gain adjustments.

Test Results: Except for minor aspects, the model 210 met all of its published specifications, and exceeded some of the more important ones. The range per band was measured by MR as ±13 dB as compared to the ±15 dB claimed. Band center frequencies were found to be within 8% (worst case situation). Frequency response with all controls centered (in the EQ line or tape modes) was confirmed as ±0.25 dB from 20 Hz to 20 kHz.

THD was found to be much lower than spec'd, and S/N was better by at least 10 dB than the 90 dB below 1-volt output claimed.

Connecting the device into a system presented no problems, and putting it through its paces by following instructions supplied was a smooth process that produced the anticipated sonic effects.

Fig. 1 shows the equalizer's front panel, while Fig. 2 shows the rear panel. In Fig. 3, MR plotted sequentially (by means of 'scope photos taken from our spectrum analyzer's storage 'scope face) the complete range of each octave control. Finally, to illustrate the precision with which the model 210 can create a given response curve, we photographed (Fig. 4) the actual response curve obtained when the ten controls were adjusted as shown in Fig. 1.

General Info: Supplied in black metal case with white markings. Dimensions: 17 inches wide; 6 inches high; 6½ inches deep. (Rack-mount version is model 210R.) Weight is 12 lbs. Advertised retail price is $275. Optional cabinet, $30.

Joint Comment by N.E. and L.F.: The model 210's wide range of available gain adjustment is not commonly found on most equalizers. While in most instances (and in our listening tests) one probably will not find the need to move these overall-gain sliders too far off their center positions, it is quite possible that an occasional program source may have widely differing gain from other sources—and then these gain controls would make for a convenient means of adjusting overall gain to preferred levels, while at the same time, making certain that the equalizer itself is neither being overloaded nor being fed a signal that will not utilize its best S/N capabilities.

Clearly, the model 210 was designed with the tape recordist in mind. In addition to providing a substitute "tape in/tape out" circuit for the one you may have

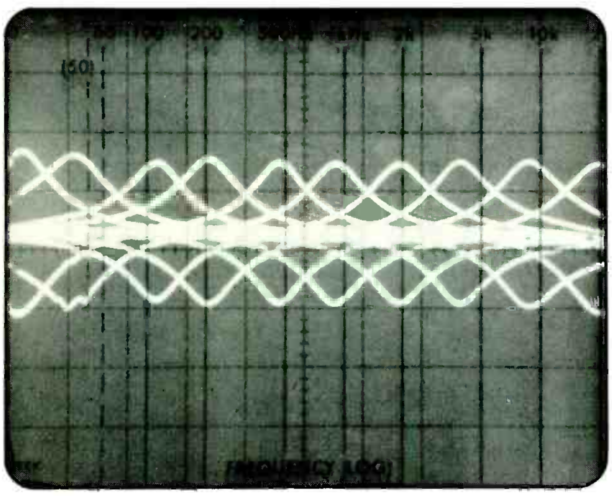


Fig. 3: Spectro Acoustics 210: Boost and cut ranges of each of the octave controls.

preempted when interconnecting the 210, the device also provides for both pre- and post-record equalization. Circuitry was judged to be up-to-date in that inductorless filters, employing an op-amp with feedback, are used. Sometimes called "gyrator" circuits, these "synthesized inductors" do away with the possibility of hum pickup and the need for unusual sheltering. They also help reduce manufacturing costs vis-a-vis the use of conventionally-wound coils.

Listening tests confirmed that the model 210 introduces no audible coloration when controls are all set flat, and the EQ is switched in. By the same token, in-

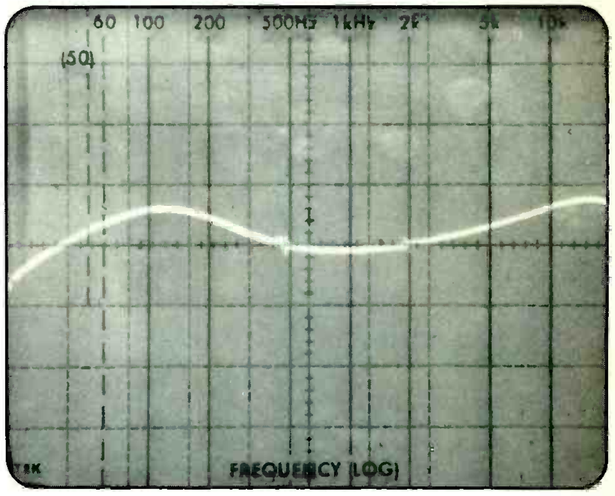


Fig. 4: Spectro Acoustics 210: Response obtained with octave controls set as in figure 1.

teraction between adjacent controls is minimal, which means you can pinpoint the exact response curve desired without having to go back-and-forth several times in trial-and-error fashion while readjusting nearby controls each time you shift other controls of the ten-band group.

Our one criticism of this otherwise excellent unit has to do with the lack of a detent, or "click-stop" position on the individual octave sliders. Since the span of each control is not all that long, one has to get real close and stare directly at the front panel in order to return a slider to its flat position while avoiding the visual-parallax error.

SPECTRO ACOUSTICS MODEL 210 STEREO GRAPHIC EQUALIZER: Vital Statistics

PERFORMANCE CHARACTERISTIC	LAB MEASUREMENT
Range, per band	±13 dB.
Filter Q	2.4 (960-Hz band measured)
Frequency response, EQ in, sliders on flat	±0.25 dB, 20 Hz to 20 kHz.
Band center frequencies	As indicated on panel, all within 8%, worst case
Gain adjustment per channel	±15 dB re: unity
THD, controls flat	0.0027% at 1 kHz; 0.0025% at 20 Hz; 0.04% at 20 kHz; all for 2 volts (rated) output.
Hum and noise	−100 dB re: 1 volt out, controls flat; −95 dB re: 1 volt out, controls at maximum cut; −106 dB re: 1 volt output, controls at boost.
Output voltage, clipping	11.5 V; 3.5 V at 20 kHz
Input impedance	50 k ohms
Output impedance	600 ohms
Power consumption	13 watts

ADVENT 201A CASSETTE RECORDER

General Description: Advent's Model 201A is an updated version of the model 201, which itself was a lineal descendant of the model 200, the first cassette recorder designed for handling chromium-dioxide tape and with built-in Dolby-B noise reduction. Vis-a-vis the 201, the new version looks very much the same, but certain improvements have been added.

The 201A includes a headphone output with ample drive for just about any headphones except electrostatics. The record/play head is made of a new, Sendust alloy for improved audio response. Meter equalization circuitry has been changed to provide a better idea of recording levels (especially at high frequencies). And the line output level has been increased to 1 volt (from a little better than 0.5 volt in the older model). Output level for headphone listening, and for line output to an external amplifier (or receiver), is controlled by a single knob.

A "top-loader" design (that may be installed horizontally or vertically), the 201A has a black dress plate with white lettering fitted into a walnut case. The single VU meter is at the upper left. The meter may be switched to indicate channel A, or channel B, or the higher of A or B. It shows recording level, playback level, and is also used to calibrate the Dolby system. Occupying the center of the dress plate is the drop-in cassette well. Flanking it are a tape-index counter and reset button, and the unit's AC power switch. Just below are switches for stereo/mono (input only); Dolby noise-reduction off/on; tape selector (CrO_2 or "regular"); the meter switch; the eject button. Below this row are input level controls (separate for each channel) and a master recording level control; the transport high-speed lever (it may be moved either way for fast forward or rewind); the recording button; a pause control; and the two transport keys for play and stop.

Along the left side of the deck and recessed in a cutout in the wooden side-panel are the line in and out jacks, the headphone jack, the output level control, a special jack that supplies 18 volts DC (for powering an accessory Advent microphone preamplifier) and a small pushbutton to activate a record-calibration tone.

The underside of the 201A contains additional adjustments that are not intended for use by the owner; these are factory set and are supplied as a service contingency. They include playback calibration for each channel; oscillator calibration for each channel; meter sensitivity adjustment. However, recording calibration adjustments (on the rear of the recorder) may be handled as per instructions in the owner's manual.

The 201A has no microphone input; to record "live" requires the use of the Advent accessory mic preamp, model MPR-1 (not tested).

A list of tapes by brand and type, with recommended tape switch settings, is appended to the owner's manual. Also included is a timing chart relating tape-counter numbers to elapsed time and remaining time for C-60, C-90 and C-120 size cassettes. The 201A comes with signal cables, a sturdy plastic transparent cover and a demonstration tape containing five short excerpts from Advent's own commercially recorded cassette library.

Test Results: Advent has upped its own specs somewhat for the 201A, as compared to the older 201, and happy to report, the tested model met them all with some room to spare. Response with normal tape

Advent 201A: Front panel view.

ran within ± 2 dB from 30 Hz to 14 kHz; with CrO_2 tape it improved to within ± 2 dB from 28 Hz to 15.5 kHz. Wow and flutter, measured at 0.12% (DIN) went better than the claimed amount of 0.15%. The most telling improvement, however, in audio performance of the new 201A was in its markedly better S/N figures.

The best S/N obtained was an impressive 63 dB, using CrO_2 tape and with the Dolby-B switched on.

Our tests were conducted using Advent's own CrO_2 tape (for measurements of performance with chrome), and TDK's Audua (for measurements of "regular" or "normal" tape). Unaccountably, the distortion level using the CrO_2 sample was lower than with the TDK ferric variety. This, in spite of the fact that we were careful to set the switch as forewarned in Advent's addendum ("Reg" for record, "CrO_2" for playback). Obviously, the bias was incorrectly set for this tape. Figure 1 shows a spectrum analysis of distortion as observed from a 1 kHz played back signal recorded at 0 dB, using the CrO_2 tape. In Fig. 2, the same observations were made, after recording a 1 kHz tone at 0 dB level, using the TDK sample, and second and third harmonics are both considerably higher. We *know* it's not the TDK tape, for we have used that brand with a host of other machines and gotten much lower THD figures. Obviously, if you are going to use that tape, keep away from the 0 dB record level, which turns out to be the nominal 3% distortion point (as read on a regular distortion analyzer) as the machine is presently calibrated.

Frequency response of the complete record/play cycle, using TDK Audua recorded at a − 20 dB level is

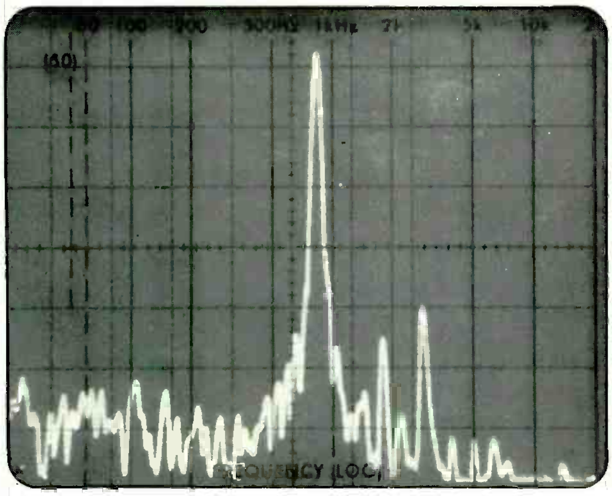


Fig. 1: Advent 201A: Using Advent CrO_2 tape at 0 dB recording level, 2nd and 3rd harmonics are 52 dB and 44 dB below 1 kHz playback fundamental.

plotted in Fig. 3, and results equalled claims for "standard tape" at the high end, and exceeded claims somewhat at the low frequency extremes. In the case of CrO_2, results are plotted in Fig. 4, and this tape did a bit better than claimed at the high end (the − 2 dB point occurred at 15.5 kHz), and as claimed at the low end. There was also more "headroom" for the CrO_2 tape, as the machine is presently calibrated and, if we had measured signal-to-noise from a reference of +2.5 dB instead of from 0 dB (as Advent specifies it), results would have been better by the same 2.5 dB. Most manufacturers do measure S/N from a 3% THD record level reference, so, to be fair, think of the S/N for the 201A as being 58.5 dB, "A" weighted, without Dolby and a superb 65.5 dB, "A" weighted, with Dolby noise reduction circuitry activated.

Mechanically, the 201A ran perfectly well, although the action of some of the switches seemed "stiff," and the fast-wind lever must be held in position to operate.

General Info: Supplied in walnut case. Dimensions: 13⅞ inches wide; 9⅝ inches deep; 5⅛ inches high (with cover in place). Weight: 18 lbs. Suggested retail price: $399.95.

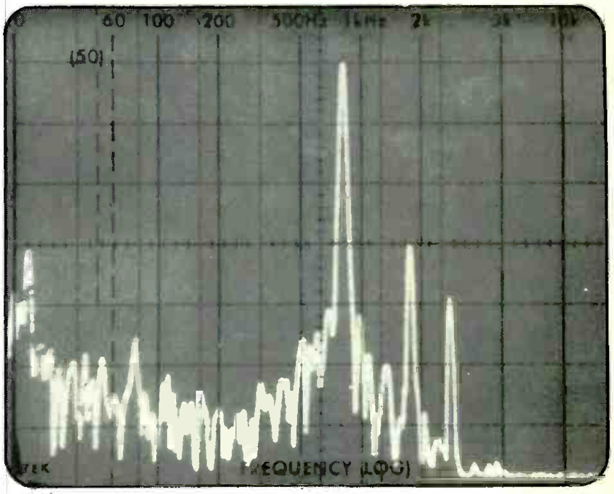


Fig. 2: Advent 201A: Using TDK "Audua" tape at 0 dB recording level, 2nd and 3rd harmonics are 30 dB and 39 dB below 1 kHz playback fundamental.

Individual Comment by L.F.: When I first learned that the Advent Corporation had updated their long-lived Model 201, I felt certain that the latest version would take care of some of the objections I had voiced with regard to the earlier machine. Those objections related to mechanical operation and external features, and not to electrical performance. The old 201, in its day, was one of the first stereo cassette decks to demonstrate what could really be accomplished with the then "lowly" 1-⅞ ips cassette tape format. Indeed, until recently, it remained an excellent performing machine for its price, compared with what was available. But there were always those objections (at least on my part). The unit offered only *one* meter. (You had to pick the channel you wanted to monitor, or you could set a switch which then allowed the meter to read the "loudest" of the two channels being recorded.) The machine lacked a microphone (let alone two microphones) input, and one had to buy an "accessory" (available from Advent) to do any "live" recording. The old 201 even lacked a headphone amplifier and jack. Worst of all, in my opinion, the old machine had about the "clunkiest" transport controls going. You nearly had to break your thumb getting the machine into the pause mode, and, as far as the fast-wind modes were concerned, you had to hold the high speed lever with your fingers while doing fast forward or fast rewind operations.

Well, the new Advent 201A has solved a few of these problems, added some internal improvements that make its performance even better than it was, but, in the main, it has retained many of the mechanical things to which I objected in the first instance. There is still only one meter. I know that Advent has spent a great deal of time and effort justifying this arrangement, but I'm sorry—I just don't buy it. Anyone doing more than casual "dubbing" needs to know what is going on in each channel all the time. The "clunky" transport controls haven't changed at all, and, while I can't fault their actual operation, I find them awkward to use and unjustified in a deck at this price. The fast-forward-rewind lever *still* has to be held by hand when doing its job, and the pause control is still difficult to actuate.

The saving grace, of course, is the unit's performance. As you can see from our table of vital statistics the performance of the Advent 201A cannot be faulted. But, one has to care little about convenience and ease of operation to overlook the somewhat archaic design of this deck.

I recently came across a nice accessory from Teac. It's a 4-meter bridge which can be hooked into just

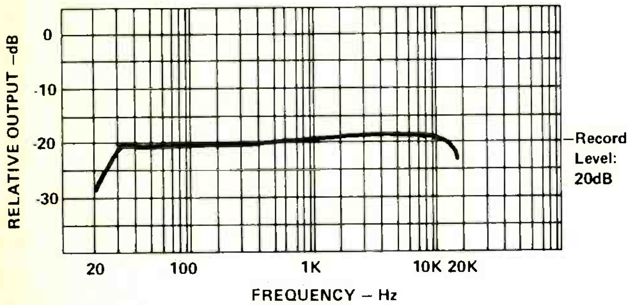


Fig. 3: Advent 201A, record/reduction response, using TDK "Audua" cassette tape.

about any tape equipment. With such an accessory added, and with the separate microphone preamp available from Advent, you could put together a pretty fair machine. But then, you'd probably end up spending a lot more, have a bunch of loose accessories crowding your table or shelf, and still have to wrestle with that pause control and hang on to the rewind lever for what seems like an eternity. No, there are some things I won't overlook, even when they're in the interest of good basic performance.

Individual Comment by N.E.: There are, it seems to me, two ways of thinking about the Advent 201A cassette recorder. One is to characterize it as a unit in which audio performance has been deliberately "gone after" and genuinely improved, while appearance and mechanical "silkiness" and many ancillary features (aspects that are plainly in evidence on competing models) have all been held to a minimum. If so, this seems to be in keeping with the general Advent

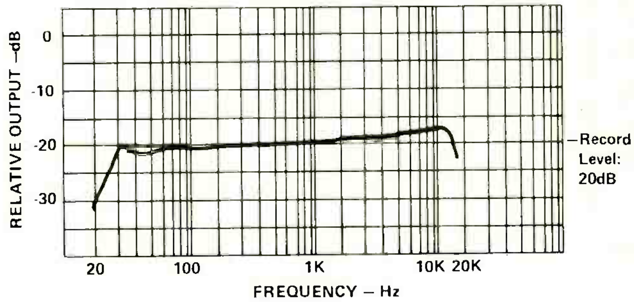


Fig. 4: Advent 201A, record/playback response, using Advent CrO_2 tape.

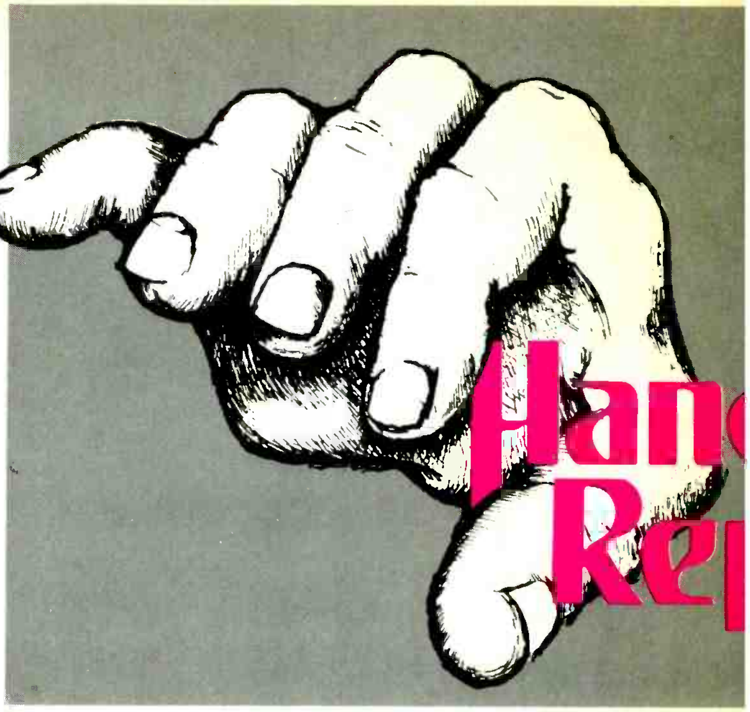
"product philosophy" which is manifest in other items from this company such as their speakers and receivers. The other view that one might take of the 201A is simply that it stands as a very good "consumer grade" cassette machine, but one that probably will be cussed-out at times by the tape-activist because of its single meter, its lack of a mic input, its awkwardly located output level control and the general stiffness of many of its switches.

Whatever view one takes, however, the fact of its price cannot be ignored. The cassette recorders that rival the Advent in terms of audio performance and that also include all the other niceties generally come higher than the $400 price of the Advent.

ADVENT 201A CASSETTE RECORDER: Vital Statistics

PERFORMANCE CHARACTERISTIC	LAB MEASUREMENT
Frequency response, normal tape	± 2 dB, 30 Hz to 14 kHz
CrO_2 tape	± 2 dB, 28 Hz to 15.5 kHz
S/N ratio, normal tape, Dolby off	52 dB
S/N ratio, normal tape, Dolby on	58 dB
S/N ratio, CrO_2 tape, Dolby off	56 dB
S/N ratio, CrO_2 Dolby on	63 dB
THD at 0 VU, normal tape/CrO_2	3%; 1.8% (see text)
Record level for 3% THD, normal tape	0 dB
Record level for 3% THD, CrO_2 tape	+ 2.5 dB
Line input sensitivity	36 mV
Line output level	0.9 V
Headphone output level (8 ohms)	8 mW
Wow and flutter	0.12% DIN; 0.07% WRMS
Fast-wind time (C-60)	45 seconds
Bias frequency	110 kHz
Power consumption	20 watts

By Jim Ford and Brian Roth



Yamaha P-2200 Power Amplifier

General Description: A new entry into the high-power amplifier market has been made by Yamaha International. Their Model P-2200, rated at 200 watts per channel into 8 ohms (350 watts into 4 ohms) is specifically designed for high wattage applications (sound reinforcement and recording studio monitoring purposes).

It has an impressive appearance. The black front panel is dominated by an oversized power switch, two detented and calibrated input volume controls and two large illuminated meters. The meters are quite interesting in that they are calibrated over a 55 dB range, and they feature a very fast attack time and a slow release. This makes the meters much more useful than regular VU meters; the P-2200 meters are capable of responding to instantaneous program peaks found in real music. Completing the front panel controls is a tiny power indicator LED and an overheat indicator LED. Two massive handles facilitate installation and removal of this large unit.

Extruded heat sink fins bristle from the sides of the amplifier. The top and bottom cover plates are perforated to allow air flow through the entire unit.

The necessary connectors for audio and power are found on the rear panel. Two pairs of "5-way" binding posts are supplied for loudspeaker hook-up. Each channel has a choice of four input connectors: a male 3-pin "cannon" jack, a female 3-pin "cannon" jack and a pair of ¼-inch phone jacks. The jacks are all parallel wired to allow "looping" of many different amplifiers together in a large sound system. Since the P-2200's inputs are unbalanced, a small slide switch for each channel determines how the pins of the "cannon" jacks are connected into the amplifier circuitry.

The power cord is thick since the P-2200 can draw up to 1000 watts of AC current under heavy drive conditions. The cord is permanently attached to the amp. Two AC power fuse holders are also found on the back. A feature very seldom found (and very often needed) in other large power amplifiers is a pair of AC power "convenience" outlets. Intended for low power demand and labeled "fan use only," these outlets can also power things like a trouble light, electronic crossover, even a soldering iron! Now you know why they're called convenience outlets. A pair of plastic feet allows the amplifier to be set with the front panel facing up without danger of damaging any of the rear panel connectors.

A peek at the innards of the amp revealed that the majority of the circuit components were arranged on two printed circuit boards, and that each board is attached to one of the heatsink assemblies. In the central portion of the interior is found a large round can that encloses a toroidal-wound power transformer. A toroid has the advantage of generally being smaller in size and weight as compared to the laminated type transformers most commonly used. A pair of large 2200 mfd electrolytic capacitors tells of stoutness of the power supply design.

The preliminary manual we looked through is quite thorough, not just on this one particular amplifier but on sound equipment in general! The service manual also appears complete enough for a good technician. We did get a laugh from some strange English mutilations of the English language (the service manual is printed in Japan) like "8 ohm lords," and bizarre statements like "improving reality."

The appearance and published specifications of the Yamaha P-2200 indicate that it should be a good performer, so, off to a job.

POWER OUTPUT

Load impedance	1 channel driven	Both channels driven
4 ohms	400 watts RMS	325 watts RMS
8 ohms	248 watts RMS	215 watts RMS
16 ohms	144 watts RMS	131 watts RMS

TABLE 1: Continuous sine wave power output at clipping, 1000 Hz.

TEST FREQUENCIES

Power Output	20 Hz	1000 Hz	20,000 Hz
200 watts	.02%	.006%	.01%
50 watts	.01%	.006%	.01%
12.5 watts	.04%*	.01%	.04%†
3 watts	.05%*	.02%	.04%†

*mostly hum †mostly noise

TABLE 2: Total harmonic distortion at various power levels and frequencies, 8 ohm loads, both channels driven.

Power Output	Intermodulation Distortion
200 watts	.016%
20 watts	.008% (mainly noise)
2 watts	.007% (noise)

TABLE 3: Intermodulation distortion (60 Hz and 7000 Hz mixed 4:1) 8 ohm loads, both channels driven.

Condition	Output noise - 20 Hz - 20 kHz unweighted
Vol. controls max.	500 microvolts or .03 microwatts (-98 dB)
Vol. controls min.	90 microvolts or .001 microwatts (-113 dB)

TABLE 4: Output noise (level in dB below maximum output of 200 watts shown in parenthesis).

Ch. A driven, Ch. B volume control max. with no input

Frequency	Crosstalk Level
20 Hz	82 dB below 200 watts
1 kHz	84 dB below 200 watts
20 kHz	73 dB below 200 watts

Ch. B driven, Ch. A volume control max. with no input

Frequency	Crosstalk Level
20 Hz	77 dB below 200 watts
1 kHz	84 dB below 200 watts
20 kHz	55 dB below 200 watts

TABLE 5: Crosstalk (worst case)

Field Test: We began our evaluation of the amplifier in our listening room; JBL 4315s and 4343s were used for monitoring. The Yamaha sounded fine, and compared well with several other high powered amps. We did notice that when driven to high levels at or near clipping—with full range program material—the P-2200 didn't sound as smooth as a competing amplifier with less sine-wave output. At slightly lower levels, the P-2200 lost this "edgy" sound.

The front panel meters were a constant source of comment since their behavior is somewhat different from regular meters. Everyone that saw their action agreed that the instantaneous indications were quite useful for monitoring the output signal of the amplifier under dynamic conditions.

The next evaluation of the unit came when the P-2200 was substituted for a "household name" power amp in a medium-sized bi-amped P.A. system. The Yamaha was powering four 8-ohm bass enclosures (two per channel), and an electronic crossover limited the input signal to the amplifier to 800 Hz and lower. In this application the P-2200 seemed capable of supplying an enormous amount of clean power. Since "A vs. B" comparison of power amplifiers is difficult in an actual "live" situation, only subjective impressions of apparent sound quality were possible. However, it appeared that the Yamaha had no problem equaling the performance of the amplifier previously used.

The amplifier became only slightly warm under any condition, even with the 4-ohm per channel bass speaker load. We did make a point to ensure free air flow around the amplifier by aiming a small "Boxer" fan over the top surface of the amplifier, but heat was so minimal we could have operated without the fan. There are probably some speaker loads that could overheat the Yamaha, but simple cooling arrangements should suffice even under extreme conditions.

The power indicator (and the similar sized overheat light) were much too small to be visible even over a short distance. Admittedly, the meter lamps will show when power is applied, but the overheat indicator must be visible in order to be of use. Upon learning from Yamaha that the overheat light will activate *before* actual shut- down of the amplifier, we feel that much stronger about the miniscule size of this potentially useful indicator.

The detented volume controls were very handy for returning to previous settings, and the 2 dB increments between steps was adequate.

In conclusion, the Yamaha P-2200 performed quite creditably even under fairly tough loading conditions. It did what it was supposed to do to the audio signal (amplify it) and little more. That is exactly what a quality power amp is supposed to do.

Lab Test: Amplifiers of this power class are always interesting to test, although sometimes our load resistors aren't particularly overjoyed. After a number of checks, we concluded that the Yamaha is a respectable performer.

Power output at the onset of clipping at 1 kHz under different loading conditions is given in table #1. Note that we chose a power line voltage of 115 volts AC ±1% for our measurements. It is a fact that nearly all solid state power amplifiers vary in maximum output wattage when the AC line voltage is varied. Consequently, we've listed the line voltage at which our lab tests have been made, and that is 115 volts rather than the commonly used 120 volts. Over the years (and after measuring hundreds of different line voltages in the country) we have established that 115 volts is a better national average than 120 volts. Of course the amplifiers will produce a little less output, but we have found that conservatively rated, high quality amps are capable of meeting "spec" at this lower voltage. We will conduct all of our basic tests at this line voltage, even if it disagrees with the manufacturer's test method.

While on the subject of line voltage, we used a variable transformer to run the line up and down over a wide range to check performance, and the only noticeable variation was maximum output power. This indicates a stable circuit design in the P-2200.

Harmonic and intermodulation distortion were quite low, and often the measurements were clouded by residual noise which itself was quite low (see tables 2, 3 and 4). Harmonic distortion products, when actually present, were generally lower ordered. Distortion measurements at 4 ohms were not significantly higher than at 8 ohms except for a somewhat higher 20 Hz value at high power output levels. The small amounts of both harmonic and intermodulation distortion at low power outputs indicate that only a minimum of crossover distortion is present. This means the amp should sound good at all loudness levels.

The clipping performance of the P-2200 was examined on the oscilloscope. We have observed that amplifiers that clip cleanly and evenly tend to sound more "graceful" when overdriven than amplifiers which clip unsymmetrically or in other odd fashions.

At low and mid frequencies, the clipping characteristics were quite good. However, at high frequencies the clipped waveform was quite unsymmetrical and ragged. This might indicate why the amp sounded as it did when driven hard with full range program material on our studio monitors.

The amplifier had little problem dealing with capacitive load. With 2 microfarads of capacitance in parallel with 8 ohms resistance some "ringing" of a square wave signal was noted.

Output noise levels were in the sub-fractional wattage region. Noise products were typically AC line components.

Crosstalk between channels was low when the inputs were terminated with low impedances. Interestingly, there was more crosstalk between ch. 2 into ch. 1 than the opposite situation; this would occur only with no input connected to channel 1, and channel 2 driven to 200 continuous watts with an 8 ohm load. Typical crosstalk will be *much* lower than the worst case measurements shown in table 5.

The amplifier was capable of a slew rate of greater than 40 volts per microsecond at 100 watts output. (Slew rate demonstrates the ability of the amplifier to deliver clean high frequency performance at high output levels.)

The input attenuators were checked and were found to be within .6 dB of the attenuation amount printed on the front panel. Channel to channel tracking was within .3 dB at all settings.

The meter calibrations were checked, and both meters read about 10% on the high side, plenty close for "live" usage.

Now for some really hot tests. First we drove both channels of the P-2200 to the verge of clipping at 1 kHz with an 8-ohm load. We went to lunch and upon returning found the amplifier barely warm. Power was then reduced to 1/3 output (as per FTC measurement method) and the unit began to warm up, but it still was far from being hot. Full power at 4 ohms was easily handled, but 1/3 output really heated the amplifier up. In fact, without any warning from the tiny "overheat" light the amp shut down after about forty-five minutes of operation. A check with Yamaha revealed that the actual thermal shut-down sensor was located at the power transformer, and not on the heatsinks where the indicator light sensors are located. The power transformer was indeed extremely hot, much hotter than the heatsinks.

After waiting a length of time for the amplifier to cool, we rigged a "Boxer" fan to keep air circulating around the P-2200, and ran the 4-ohm tests again. Although the amp still heated up, it was much less than without the fan.

Our conclusions are that under most conditions the P-2200 should not run too hot, even without a fan. Certain extreme conditions may require good air circulation. As a matter of principle, we always use forced air in our P.A. systems' amp racks, even if our tests don't show any need for it. This way, nothing overheats even if used outdoors under the summer sun.

We do question Yamaha's thermal sensor arrangements since the indicator and actual shutdown mechanisms are located at different locations, and could cause possible confusion. In fact, as we interpret the situation, the heat sinks can rise to astronomical temperatures (without shutting down the amp) as long as the power transformer stays comparatively cool.

In terms of actual results, both field and lab, the Yamaha P-2200 is one of the best units we have used.

Conclusion: This amplifier really pleased us. It included a number of useful features not found on competing units. It seemed to have plenty of soup even under heavy loads, and for the most part operated with low amounts of heat.

The engineering of the amp, both electronic and human, were excellent. True, we had a few criticisms, but you should hear what we sometimes say about the "household name" brands.

Yamaha has stepped into a tough field that has been dominated by American manufacturers for years, and it appears that they've done their homework.

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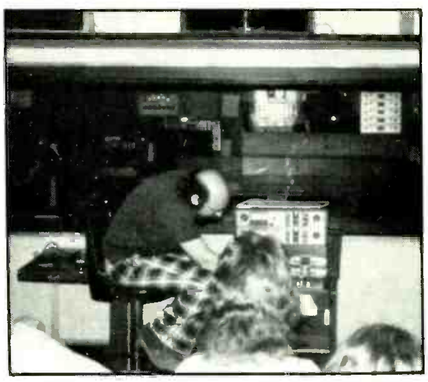

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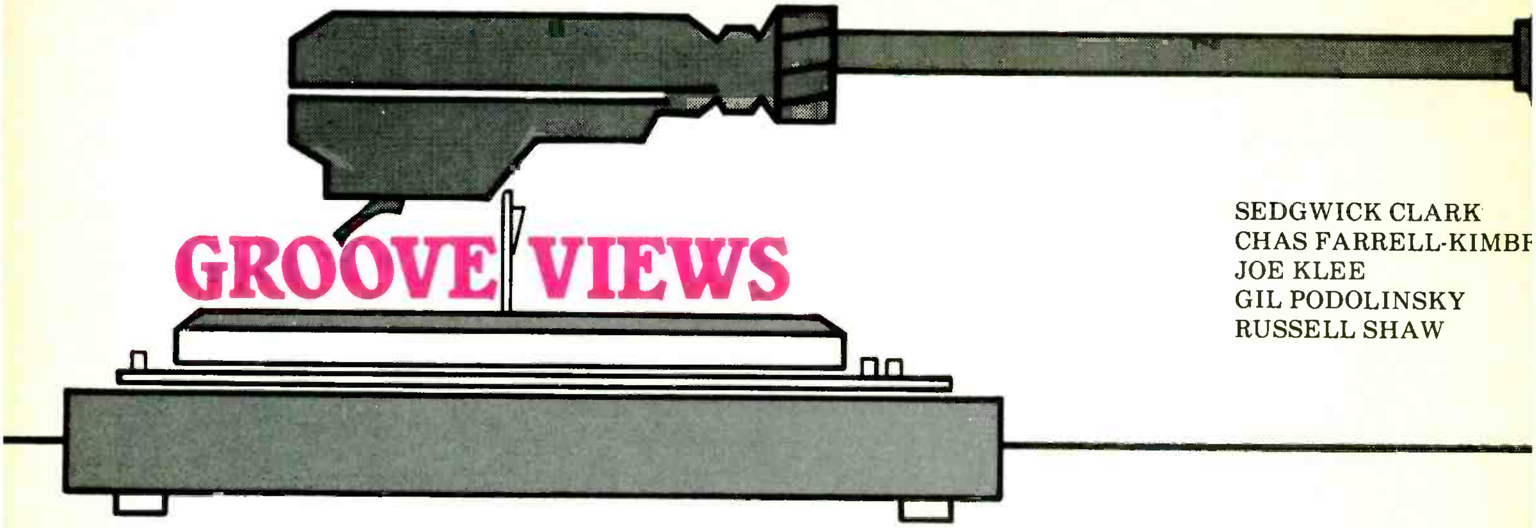

GROOVE VIEWS

SEDGWICK CLARK
CHAS FARRELL-KIMBF
JOE KLEE
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POPULAR

GENTLE GIANT LIVE: *Playing the Fool.* [Gentle Giant, producer; Paul Northfield, engineer; recorded during European tour by the Maison Rouge Mobile with sound system built and supplied by Recording Studio Design Cheshunt Herts; live mix by David Zammit; remixed at Advision Ltd., London, England.] Capitol SKBB 11592.

Performance: **Energetic**
Recording: **It's all here**

In interviewing Gentle Giant last summer, I was surprised to learn that what prompted this "live" double album was the popularity and sound reproduction of the three or four "live" bootlegs which exist of the band. Their frankness in that regard would lead me to believe that little or no re-recording was done, contrary to the norm of most "live" albums. Although there are differences between the studio and the "live" albums other than applause tracks, they are neither distracting nor detracting. Kerry Minnear, for example, doesn't carry as many keyboards on tour as he uses in the studio, so there is some substituting. Another is that Gary Green's guitar has more distortion "live" than in the studio. Very small points, but they make for a very interesting "live" recording.

The baroque introduction to "On Reflection," consisting of recorders, violin, viola and vibes is beautifully captured in performance. This is no small feat when you note the change in volume, tone, etc. when going from driving electronic instruments to soft acoustic chamber music. The material is a good cross-section from their eight previous albums, including one not available in this country. The lead vocal and harmony tracks abound in presence and take center stage in the mix. The sharp clarity of the vibes when used in conjunction with the other various instruments adds further dimensional beauty to the sound. Hopefully, by following Frampton's lead with a double "live" greatest hits LP they will attain the same success, for they have always been Giant in talent, but received only Gentle recognition. G.P.

MOE BANDY: *I'm Sorry For You My Friend.* [Ray Baker, producer; Lou Bradley, engineer; recorded at Columbia Recording Studio "B", Nashville, Tenn.] CBS KC 34443.

Performance: **Straightforward**
Recording: **Spotless**

Twice a year, Moe Bandy albums roll off the assembly line. Yet mass production need not always be a harbinger of mediocrity, only of familiarity and sameness.

Some associate the aforementioned

GENTLE GIANT: Giant in talent

values with security. Indeed, save the inclusion of a Texas swing number two issues back, Moe's releases have always followed an identical blueprint—a dominance of down-and-out drinking songs with an occasional love ballad thrown in.

Moe's disgust with women seems to take the approach of catharsis through self-pity and ceaseless residence at the local watering hole. The music matches the message perfectly. Wave after wave of mournful steel guitar, sobbing fiddles and Stelazined guitar passages help the man overcome his stated grief. In real life a happily married man, Moe does nevertheless carry off this twangy unrequitedness quite convincingly.

Under the tutelage of producer Ray Baker, Bandy-backing musicians are frequently provided with ten second lead-ins, and bridges to wail their woes. On this album, the impressive roster of Nashville session cats take turns wringing the crying towel; Johnny Gimble's opening fiddle break on "Does Fort Worth Ever Cross Your Mind" and country keyboard ace Pig Robbins' bluesy tinkle on "She's An Angel" both prefix Moe's laments beautifully.

One has no quarrel with the flawless mix on *I'm Sorry For You My Friend.* Bassist Bob Moore's mother might wonder why his floorings were a slight bit underbrewed; similarly, fans of trapper Kenny Malone might notice his prominent presence during the body of "She's Everybody's Woman." Yet when considered in the overall perspective of twenty-five crisp, ungimmicked minutes, such nitpicking should be quickly forgotten. Let those fiddles ooze with crystal clarity. If all country music were made this perfectly, the genre would not be so afflicted with credibility problems. R.S.

SEA LEVEL: *Sea Level.* [Stewart Levine, producer; Sam Whiteside, engineer; recorded at Capricorn Sound Studios, Macon, Ga.] Capricorn CP 0178.

Performance: **Worthy of the lineage**
Recording: **High quality**

Allman Brothers fans were essentially divided into three camps: those who liked the hard driving boogie-blues of early ABB work; afficianados of the chiming country sound exemplified best by the guitar work of Richard Betts; and devotees of the extended, exploratory, jazz-like jams which were at the core of the ensemble's creative output.

While band members with country and blues leanings work on their own solo projects, the core of the improvisatory aspect has banded together to form Sea Level. A truly innovative quartet who manages to overlay long, complex musical discussions with a heavily percussive backbeat, Sea Level is worth serious audit and a considerable amount of critical respect.

Much of the musical direction of the new group is contributed by Chuck Leavell, a superbly resourceful pianist whose work with the ABB was often overlooked in favor of less valid contributions by more colorful personalities. Leavell's conceptions, on a variety of keyboards, manage to avoid redundancy and pave new ground.

With the sympathetic fills of Jai Johanny Johannson, drums, Lamar Williams, bass, and Jimmy Nalls, guitar, the many colorations produced affirm Sea Level's wide perspective. Taken in total, the music seems to strike a middle ground 'twixt the fusion music of Chick Corea and the

A Coltrane Surprise as Granz Expands

By Nat Hentoff

Having ended his long sabbatical from the record business with the introduction of the Pablo label, international impressario Norman Granz swiftly proceeded to produce or unearth a sizeable catalogue of mostly studio sessions which were primarily mainstream (Count Basie, Roy Eldredge) but also extended to Dizzy Gillespie (some of his best recordings ever). Now, Granz has decided on a sister label, Pablo Live—devoted, of course, to jazz outside the studio. It, too, will be distributed here by RCA Records.

What is surprising about his first release is the presence of John Coltrane in *Afro-Blue Impressions*, a two-LP set never previously issued. Granz has been volubly on record as disdaining avant-garde jazz, but Coltrane, it turns out, was an exception because Granz had heard him in Johnny Hodge's small combo years before and so was convinced that Trane had "roots." We are fortunate that Granz did become a convert because these performances—recorded in 1963 in West Berlin and Stockholm on a Granz-directed European tour—are magnificent middle-period Trane.

Full-bodied, wide-ranging, deeply authoritative, Trane's tenor and soprano are backed by his optimum rhythm section—pianist McCoy Tyner, bassist Jimmy Garrison, and drummer Elvin Jones. As often as you've heard "My Favorite Things," there are unexpected dimensions here—as with "Naima," "Cousin Mary," and a majestic "Spiritual." Both for Coltrane devotees and for those who may have found the idea of Coltrane somewhat forbidding in the past, this is a quintessential set. The recording is judiciously balanced, capturing, but not being overwhelmed by, the excitement of the occasion so that the music, rather than the crowd, thoroughly prevails.

The second Pablo Live is another two-LP set, *Milt Jackson at the Kosei Nenkin*, 1976 Tokyo concert by the long-time swinging core of the Modern Jazz Quartet. It is impossible for Milt not to give an intriguing performance for he may well be the most consistently fresh improviser in jazz. What makes this set his most satisfying in years, however, is the quality of planning, not usually Milt's strong suit. The repertory, for instance, is uncommonly imaginative—superior pieces by Sonny Rollins, Dizzy Gillespie, and Jackson himself, as well as such vintage invitations to grooving as "Organ Grinder Swing" and "Bye Bye Blackbird."

Then there is the personnel. One of Jackson's complaints about the Modern Jazz Quartet was that its sound was too attenuated for his robust temperament. Well, there is no such problem here with Teddy Edwards (hard-swinging, warm-at-the-center tenor), bassist Ray Brown, pianist Cedar Walton, and drummer Billy Higgins. I first heard Billy with Ornette Coleman in the late 1950's and he sounds just as crisp, wittily resourceful, and enlivening as he did then. In fact, this is a paradigm of a firm, flowing rhythm section—as it also is of a masterful, mellow combo.

Fortunately, the music being so inherently alive, the engineering has preserved the full spirit-feel. I have seldom heard a concert recording sound so naturally "live" and in just about perfect balance. A pity there is no engineering credit.

MILT JACKSON: Fresh and intriguing

JOHN COLTRANE: *Afro-Blue Impressions.* [Norman Granz, producer; no engineering or other recording credits.] Pablo Live 2620 101.

MILT JACKSON: *At the Kosei Nenkin.* [Milt Jackson, Ray Brown, producers; no other recording credits.] Pablo Live 2620 103.

more traditional rock structures. Fortunately, the technical aspects of this disc enhance the viability of the music played.

The mix, brewed by veteran jazz producer Stew Levine (Crusaders, Hugh Masakela) and Capricorn engineer Sam Whiteside, completely avoids the pitfalls possible whenever an effort like this is undertaken. Bearing in

SEA LEVEL: Worthy of critical respect.

mind that Williams' bass attack is by design one of the most powerful in the medium, and that expressions of force are often indulged in by the rhythm section—the studio team has done a remarkable job in not allowing vibrant rhythmic underpinning to overwhelm and mechanize the recording.

This effort will be carefully scrutinized by a curious public, anxious to see what various members of the defunct ABB can accomplish on their own. In this instance, they won't be disappointed by Sea Level. R.S.

RABBIT: *Boys Will Be Boys.* [Patric Van Blerk, Trevor Rabin, Julian Laxton, producers; Julian Laxton, engineer; recorded at RPM Studios, Johannesburg, South Africa; mastered at RPM, remastered at Sterling Studios, N.Y.] Capricorn CP 0175.

Performance: **Slick**
Engineering: **Ditto**

This album has the potential to

RABBIT: A hop away from the top.

develop great staying power for this relatively new band. The material is well produced and the album flows with an ease that would suggest that these guys are old hands at recording albums. I'm beginning to feel that there is much to be said for musicians producing themselves with the help of an engineer. Trevor Rabin co-produced as well as carrying lead vocals, guitar, synthesizers, keyboards, and writing the bulk of the material.

The material is tight with excellent vocals and clever use of effects. Of special interest is the phasing on the vocals of "Lifeline" and the opening of a very lively and strong cut, "Hard Ride." The strings in "Charlie" are noticeably well-recorded. The only real sore spot is a pointless cover version of Ian Anderson's "Locomotive Breath." The arrangement isn't much different from Anderson's. Fortunately, production saw fit to place the cut at the end of side one making it easy to skip. The rest of the album, however, should not be skipped by anyone who appreciates a well-planned program of progressive rock. This group of rabbits seems to *know* what's up. C.F.-K.

JEFF BECK WITH THE JAN HAMMER GROUP: *Live.* [Jan Hammer, Tom Werman, producers; recorded live and remixed by Dennis Weinreich and Jan Hammer at Scorpio Sound Studios, London, England.] EPIC PE 34433.

Performance: **"Live," with editing**
Recording: **Very exact**

There is special mention on the album cover of the fact that "the stereo spectrum of this album duplicates the stage set-up with guitar positioned center right, keyboards center left, violin right and drums and bass center." True to their word, that is the way this "live" recording is presented. The separations are the most exact I've ever heard on a "live" recording, with each instrument presented honestly. With guitar and keyboard being the featured solo instruments, one would drop out when the other was featured, allowing for complete dominance of the center of the mix. In contrast, for example, the violin was relegated to the right of the mix throughout. Although the instruments were consistently cleanly recorded, there were two other areas that are weak by comparison. The first is the mixing of the ambient audience mics. The contrast is so drastic between the crisp instrument sound and the very muddy, cavernous audience response that it detracts from the excitement of the performance. The second is the inaudible vocal on "Earth (Still Our Only Home)" which improves slightly on "Full Moon Boogie." Neither the

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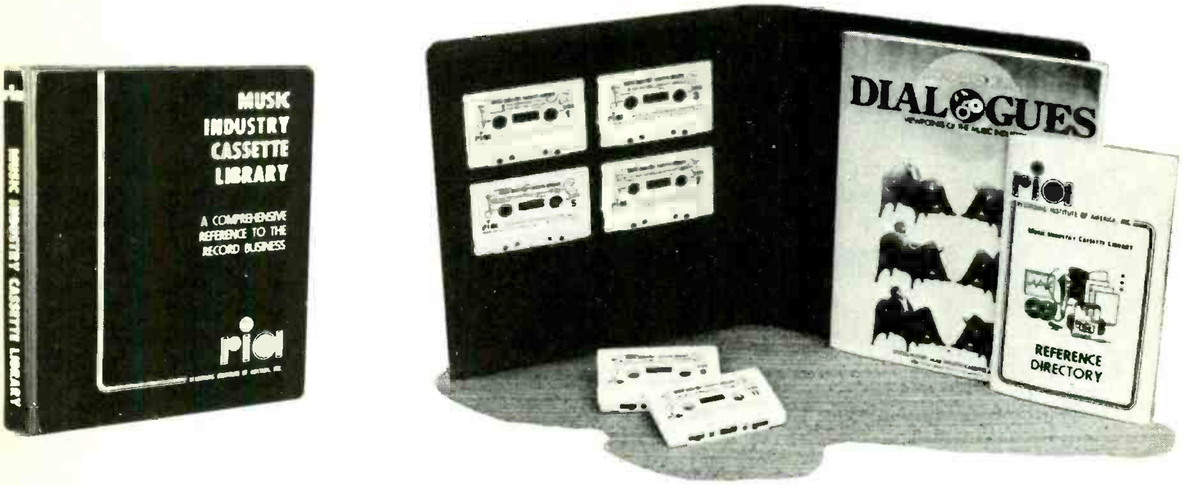

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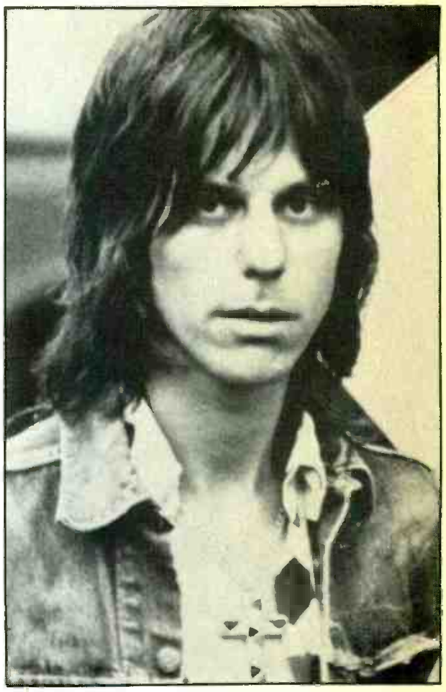
vocalists involved nor the lyrics are a highlight of the act, but with such care given to the instrumental aspects of this recording, one can only view these two faults as an oversight or lackadaisical attitude towards what may be considered the nonessentials. Finally, after



JEFF BECK: A lackadaisical attitude for the nonessentials

"seeing" this record in performance, it is safe to say, with reservation, that this recording represents the more successful aspects of the performance. There were several songs where the rhythm section of Saunders/Smith proved to be too much for Beck, running him over with various changes in rhythm. This was brought home to anyone paying attention as the famous Beck "Glare Of Annoyance" emerged often, underlining the reality that the absence of singing does not a jazz musician make. G.P.

VERNON BURCH: *When I Get Back Home.* [Vernon Burch, Tony Sobel, Joe Blocker, producers; Larry Miles, Tyrone Williams, Bob Mockler, engineers; recorded at Crystal Sound and Sun West, Los Angeles, Ca.] Columbia PC 34701.

Performance: **The Soundeffects doing Stevie Wonder**
Recording: **Accomplice in crime**

It won't be long until Stevie Wonder becomes passe thanks to everyone from tavern groups to George Benson copying

VERNON BURCH: Borrowing too much from "brother Stevie"

his material and vocal phrasing. The latest and most blatant is Vernon Burch. The material sounds like he composed the album after listening to a "Stevie Wonder Rebound Weekend." Not only does he credit his "brother, Stevie The Wonder Boy," but even the press release wants you to visualize him as a cross between Wonder and "the sex apppeal of Al Green." The sentence concludes with the sentiment that, "he still manages to be his own man."

Everything has been stolen from Stevie Wonder albums. The Motown percussion sound of the clipped hi-hat and tambourine with deep floor tom back beat is there; the Wonder use of horns to accentuate the pick up, with female vocal sweeteners on the chorus is there as well. The vocal is centered with slight echo and flanked by drums, rhythm, backing vocals—all on an even keel. I *will* say that the Rhodes is undistorted, though it does sound compressed as a result. Burch is also featured as a guitarist, but that's another hoax. Also, the occasional single horn lines have that echo effect that sound like they were recorded in the Holland Tunnel. If I were Stevie, I'd disown brother Vernon. G.P.

JAZZ

AL DIMEOLA: *Elegant Gypsy*. [Al Di-Meola, producer; Dave Palmer, Robert Ludwig, engineers; recorded at Electric Lady Studios, New York.] Columbia PC 34461.

Performance: **Captivating**
Recording: **Virtuous**

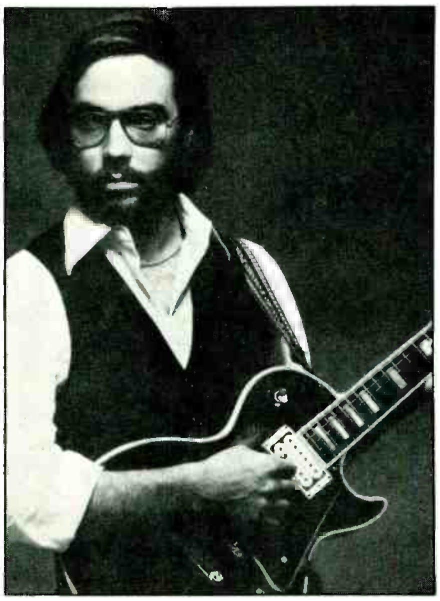
Formerly one of the creative forces in Chick Corea's Return To Forever, Al DiMeola has, in the space of little over a year, issued two solo albums of breathtaking imagination and design. Most new artists take at least that long to get off the ground. DiMeola, however, was seemingly born aloft.

The 23-year old whiz is a player with mature concepts of subtlety, rhythm, change and flow. Although possessed of a limitless technical vocabulary, he doesn't allow himself to become self-indulgent. The perfect team player, he coexists well with other personnel.

On *Elegant Gypsy*, an amalgam of all-star talent surrounds Al at every turn. "Flight Over Rio" is especially noteworthy via the rapid pacing of DiMeola, the characteristically sci-fi lead synthesizer lines of Jan Hammer, and the hammering bass of Anthony Jackson. A better than average percussion section, consisting of conga-bound Mingo Lewis and star session trapper Steve Gadd, underlays the proceedings with some frenetic voicings of its own.

While some would name the always-changing "Elegant (and that it is) Gypsy Suite" as the tour de force of the platter, preference here goes to the two acoustic tracks, "Lady Of Rome, Sister of Brazil," and especially "Mediterranean Sundance." As you might have already guessed from the titles, a Latin theme pervades this work, and on "Sundance" DiMeola is captured in an enrapturing duet with legendary flamenco guitarist Paco De Lucia.

The De Lucia collaborations underscore the brilliant engineering and production work that is consistent throughout the album. De Lucia is heard through the left speaker, and Al the right. There is such a distinction and separation that one can even take a cheap stereo unit and still perceive the total isolation. The same can be said for Al's own overdubbing on "Lady Of Rome," yet when each cut is centered, a symbiotic richness shines.



AL DIMEOLA: A lesson in taste

Those so-called "jazz rock" musicians who are either dealing in funk cliches or rambling Dorian-mode indulgencies should listen to DiMeola for some lessons in taste. R.S.

THE WORLD'S GREATEST JAZZ BAND: *The World 's Greatest Jazz Band on Tour.* [Barker Hickox, producer; engineer not listed; recorded at the Atlantic Club, Stockholm, Sweden, Oct. 1975.] World Jazz WJLP S-8.

Performance: **In the tradition of. . .**
Recording: **Honestly "live"**

THE WORLD'S GREATEST JAZZ BAND: Music's still the same

At the time of the inception of The World's Greatest Jazz Band, there were a lot of comments about the lack of modesty in the title. At that time, with Bud Freeman on tenor sax, Bob Wilber on clarinet and soprano sax and Lou McGarrity on trombone, a strong case could be made for the title. Since then, Lou McGarrity has passed away, Bud Freeman has moved to Ireland and Bob Wilber's got his own band, Soprano Summit. Yet, whatever they call themselves, this band is in the tradition of the Ben Pollack band which became Bob Crosby's Bobcats, then the Lawson /Haggert Jazz Band and finally evolved into the World's Greatest Jazz Band. Through all the name changes the music's been pretty much the same. The current players include some I find very enjoyable, such as trombonist George Masso and drummer Bobby Rosengarden and some, like clarinetist Peanuts Hucko and Al Klink on tenor sax, whom I find singularly inappropriate for this band. It's not for any lack of ability, but both Hucko and Klink are more swing-era based players than traditionalists and, therefore, they don't sit as well with a repertoire that includes "The Shiek of Araby," "Mandy Make Up Your Mind" and "The Saints." Easily the star of the record is Maxine Sullivan, whose vocals on "Wrap Your Troubles in Dreams" and "Just One Of Those Things" represent some of her best singing to date. Also commendable is Yank Lawson's muted trumpet solo on "Saint Louis Blues."

The recording was done "live" at the Atlantic Club in Stockholm and if announcements don't come through loud and clear or soloists fade in and out or the piano isn't miked as well as it should have been, that's "live" recording. The engineer walks into the club and what's there is what he has to work with. The sound would have been better in a studio, but the music wouldn't have been as good without those "live" bodies there to play to and off of. J.K.

CHARLIE PARKER: *Charlie Parker Encores.* [Bob Porter, producer; Jerry Valburn, Jack Towers, transfer and editing engineers; from original Savoy 78 rpm records, 1944 to 1948.] Savoy (Arista) SL 1107.

FATS NAVARRO: *Fat Girl.* [Bob Porter, producer; Jerry Valburn, Jack Towers, transfer and editing engineers; from original Savoy 78 rpm records, 1944 to 1948.] Savoy (Arista) SL 2216.

Performances: **Bebop's beginnings**
Recordings: **True to the originals**

When we bought these Savoy 78s in the 1940s it was to catch a glimpse of the music that *Metronome Magazine* said was going to make all our previous jazz idols obsolete. Now, thirty years later, the music seems tame in terms of the innovations which followed the bop era. Unfortunately, thirty years later a lot of the pioneers are dead. Not just Parker and Navarro, although that would be tragic enough, but these albums are filled with the names of those no longer with us such as Clyde Hart, Bud Powell, Denzil Best, Kenny Dorham and the most important voice among bop arrangers and composers, Tadd Dameron. *Sic transit gloria mundi.*

Many of these tracks have appeared on LPs before. In the early days of Savoy LPs, there were a number of reissues of Parker and Navarro material, organized with little regard for chronology, by Ozzie Cadena. Now here come Bird and Fats, well ordered and programmed in time by Bob Porter who loves the bop era as much as any one person I know, with authoritative notes by Dan Morgenstern for Navarro and J.R. Taylor on Parker. The Navarro album offers at least one version of everything Fats Navarro recorded for Savoy, plus a few alternates. The Parker album is *all* alternates. The master takes can be heard on Savoy SL 2201 which includes the issued takes of all the Parker Savoys. The unissued material comes from sixteen-inch acetate safety copies which Savoy founder, Herman Lubinsky, had the good sense to store away after the recording sessions, preserving their virgin condition and all the music that was played including false starts and breakdowns. There will be at least one more LP of Parker encores from these sources. The sound isn't great but then it wasn't great in the pre LP era of the '40s anyway. The fact that the transfers were made either from tapes for previous LP reissues or from the acetate safetys at least saves the listener the swish and hiss of World War II 78 rpm pressings. At least there's no added echo or stereophonic sound. In a phrase, what the microphone picked up is what you hear.

—J.K.

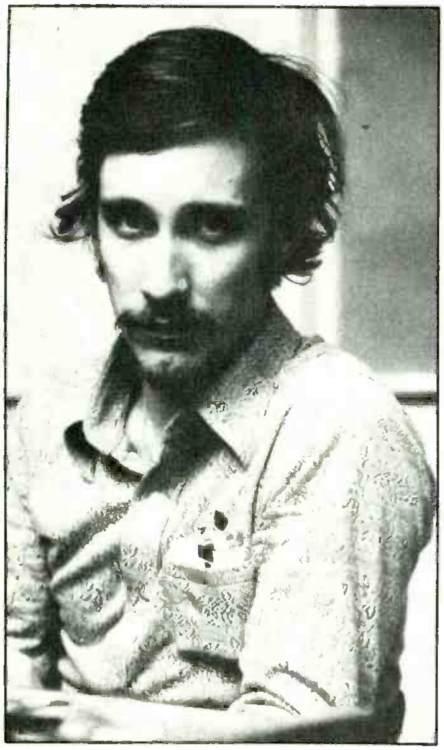
CLASSICAL

SIBELIUS: *Karelia Suite, Op. 11; Lemminkainen Suite, Op. 22.* Radio Symphony Orchestra Helsinki, Okko Kamu cond. [Cord Garben, producer; Heinz Wildhagen, engineer.] Deutsche Grammophon 2530 656.

Performance: **Idiomatic**
Recording: **Appropriate**

The music of Finnish composer Jean Sibelius presents special problems for a recording team. Of course, sound quality will always be a personal matter; just as some concertgoers like to sit close to the stage, others prefer the overall blend of sound heard from a good balcony seat. Sibelius (much like the French Impressionist school) requires a certain indefinable atmosphere, a mystery and suggestive quality that overly close miking—whatever *that* is—frequently destroys. Many well played and conducted recordings of Sibelius' music lack an element of "magic" that might well have been captured by more sympathetic engineering. The shifting rhythms and dark, brooding textures become *too* clear, too matter-of-fact. As I said, it's very personal.

This new D.G. recording is very successful, I think, in conveying a complete Sibelian experience. Although lacking a score for confirmation, none of the instruments seems slighted, the balance is judicious from the point of conducting and engineering, and the textures are clarified to just the right degree, with no sacrifice of atmosphere. Kamu's conducting is sensitive to the emotional and dynamic elements in these two early works and his musicians respond with the commitment one would expect from a Finnish orchestra. My only complaint is in the opening Intermezzo



OKKO KAMU: Captures Finnish sensitivity to Sibelius

of the three-movement *Karelia Suite*, where he begins very broadly, accelerating to the main tempo (which should have been the tempo throughout) and slowing down again as the music trails off quietly; the effect is to render the jaunty horn melody sluggish and unidiomatically Wagnerian.

What D.G. calls the *Lemminkainen Suite* is usually referred to as the *Four Legends*. The name has apparently been changed because Kamu reverts to the order in which the four movements were first performed; Sibelius had reversed the two inner sections late in his life, placing the popular "Swan of Tuonela" second. In whichever order, Kamu's competition is not strong. The Groves recording on Angel is distant and dull; Jalas on London is poorly played; Foss on Nonesuch lacks adequate string body and, like Jalas, is too closely recorded.

S.C.

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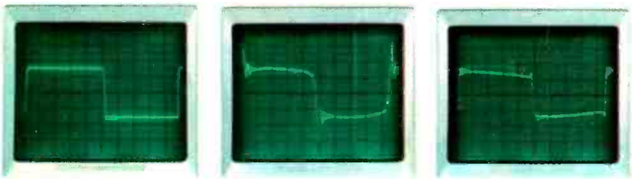

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