PROFILE: LUTHER VANDROSS

MODERN RECORDING & MUSIC

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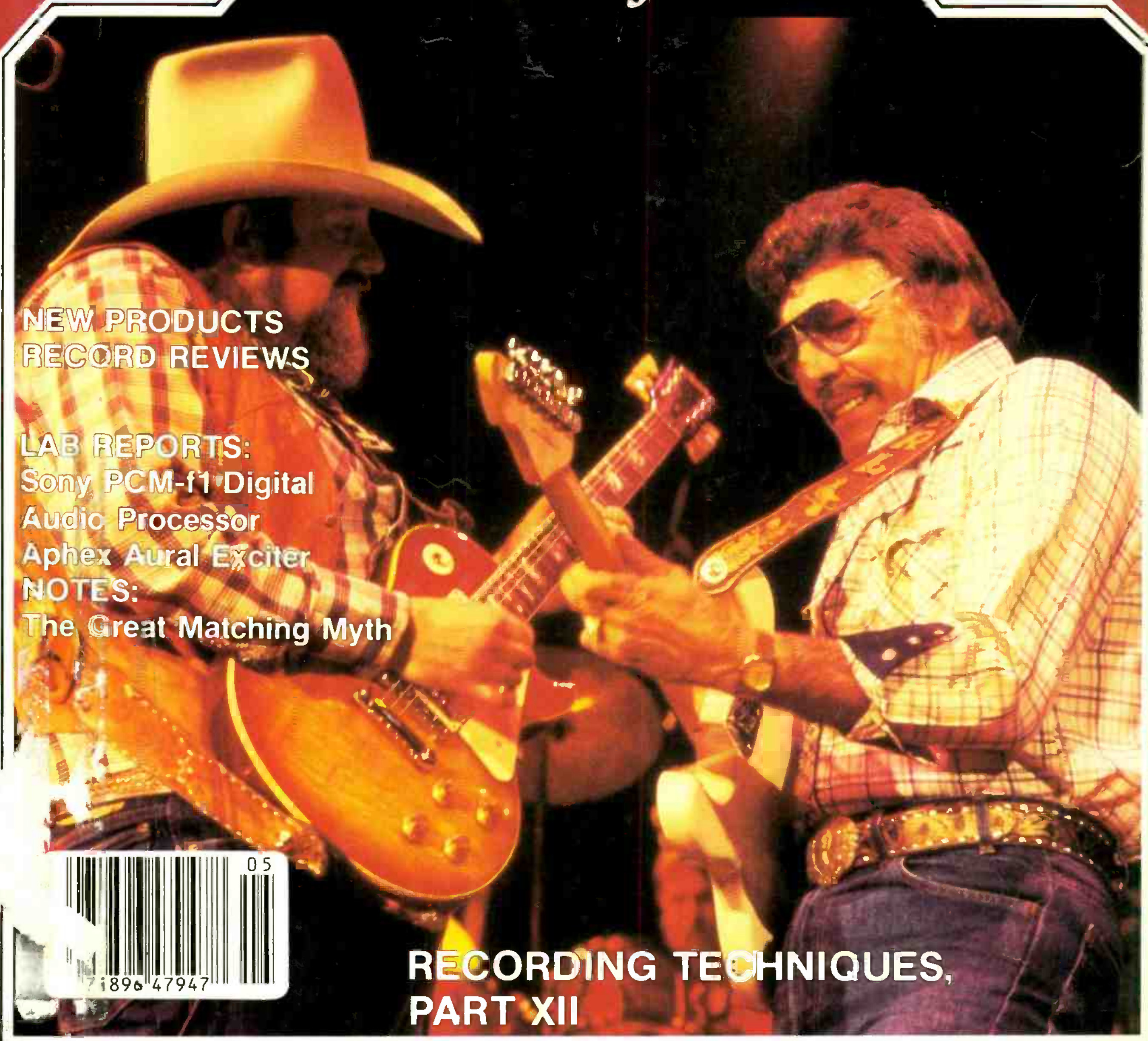
MAY 1983
VOL. 9 NO. 5

Volunteer Jam IX



NEW PRODUCTS
RECORD REVIEWS

LAB REPORTS:
Sony PCM-f1 Digital Audio Processor
Aphex Aural Exciter

NOTES:
The Great Matching Myth

RECORDING TECHNIQUES, PART XII

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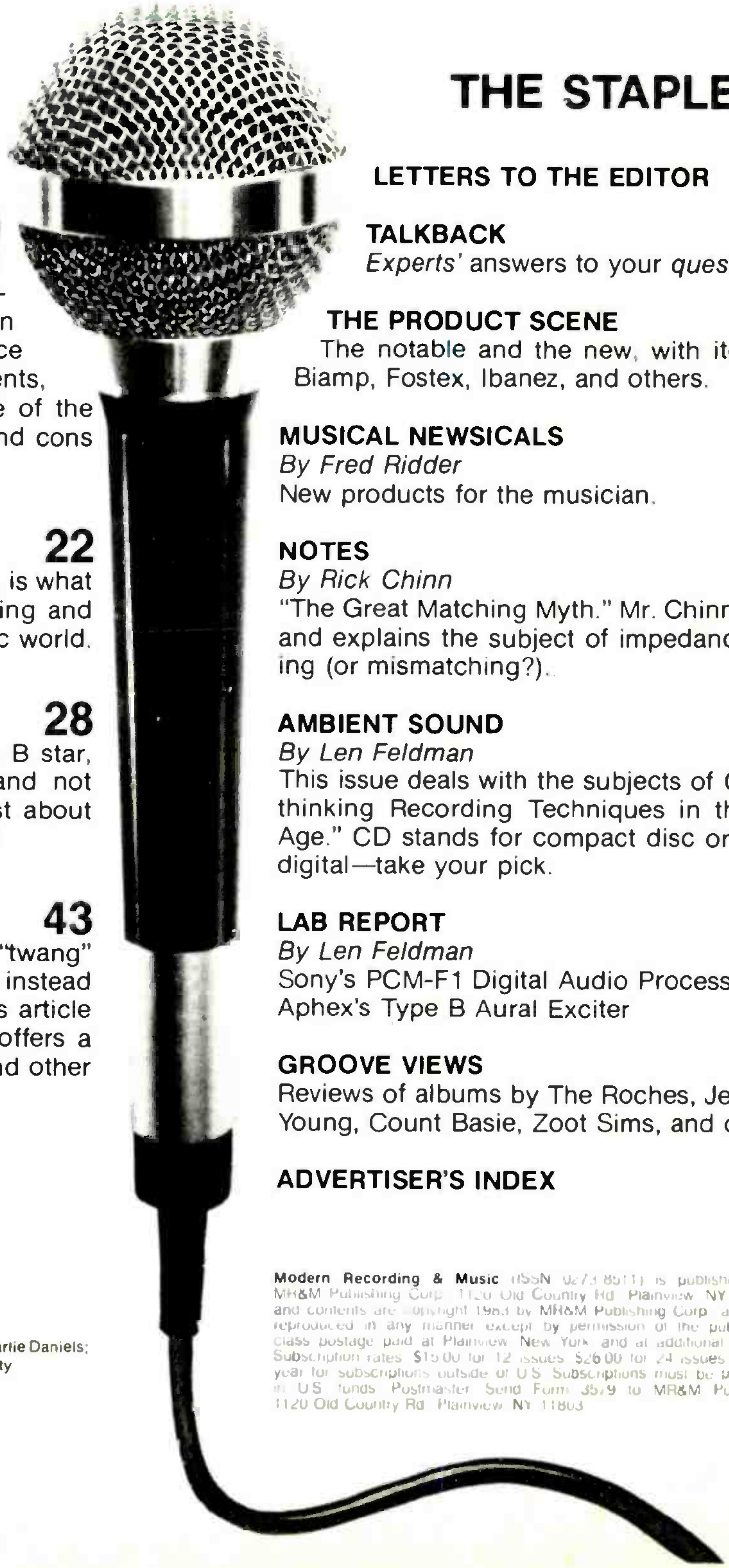

THE FEATURES

THE STAPLES

Cover Photo: Courtesy of Network INK, Inc.
Color Photos of Ray Sawyer with Dr. Hook; James Brown with Charlie Daniels; Box Car Willie, Daniels, Roy Acuff: Courtesy of Sound Seventy
Other Charlie Daniels Photo: Courtesy of Network INK, Inc.
Luther Vandross Photo: Courtesy of Epic Records

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MODERN RECORDING & MUSIC

Editorial and Executive Offices
MR&M Publishing Corp.
1120 Old Country Road
Plainview, NY 11803
516 433-6530

Editorial contributions should be addressed to The Editor, Modern Recording & Music, 1120 Old Country Road, Plainview, N.Y. 11803. Unsolicited manuscripts will be treated with care and must be accompanied by return postage.

Compressor Producers

After reading Bruce Bartlett's very interesting and informative article on compression, I would like very like to know more about which companies produce compressors—and also where they can be bought. I presently own a Tascam 144 portastudio and I feel that using a compressor would be a great improvement on my work. I realize that you may not be in a position to recommend a particular brand, but any general information would be helpful.

—M. A. O'Connor
Victoria, B.C.

We received the following response from Bruce Bartlett:

I would appreciate your comments. The following is a PARTIAL list of companies that make compressors suitable for home studios. Ask them for product literature and for the addresses of local dealers. Be sure to note whether the product requires a separate power supply or is self-contained. Also find out if the connectors will mate with your cables.

dbx, Inc., 71 Chapel St., Newton, MA 02195. (617) 964-3210.

United Recording Electronics Industries, 8460 San Fernando Rd., Sun Valley, CA 91352. (213) 767-1000.

LT Sound, P.O. Box 1061, Decatur, GA 30031. (404) 284-5155.

Orban Associates, Inc., 645 Bryant St., San Francisco, CA 94107. (415) 957-1067.

Ashley Audio, 100 Fernwood Ave., Rochester, N.Y. 14621. (716) 544-5191.

Spectra Sonics, 3750 Airport Rd., Ogden, UT 84403. (801) 392-7531.

Furman Sound, Inc., 616 Canal St., Suite 29, San Rafael, CA 94901. (415) 456-6766.

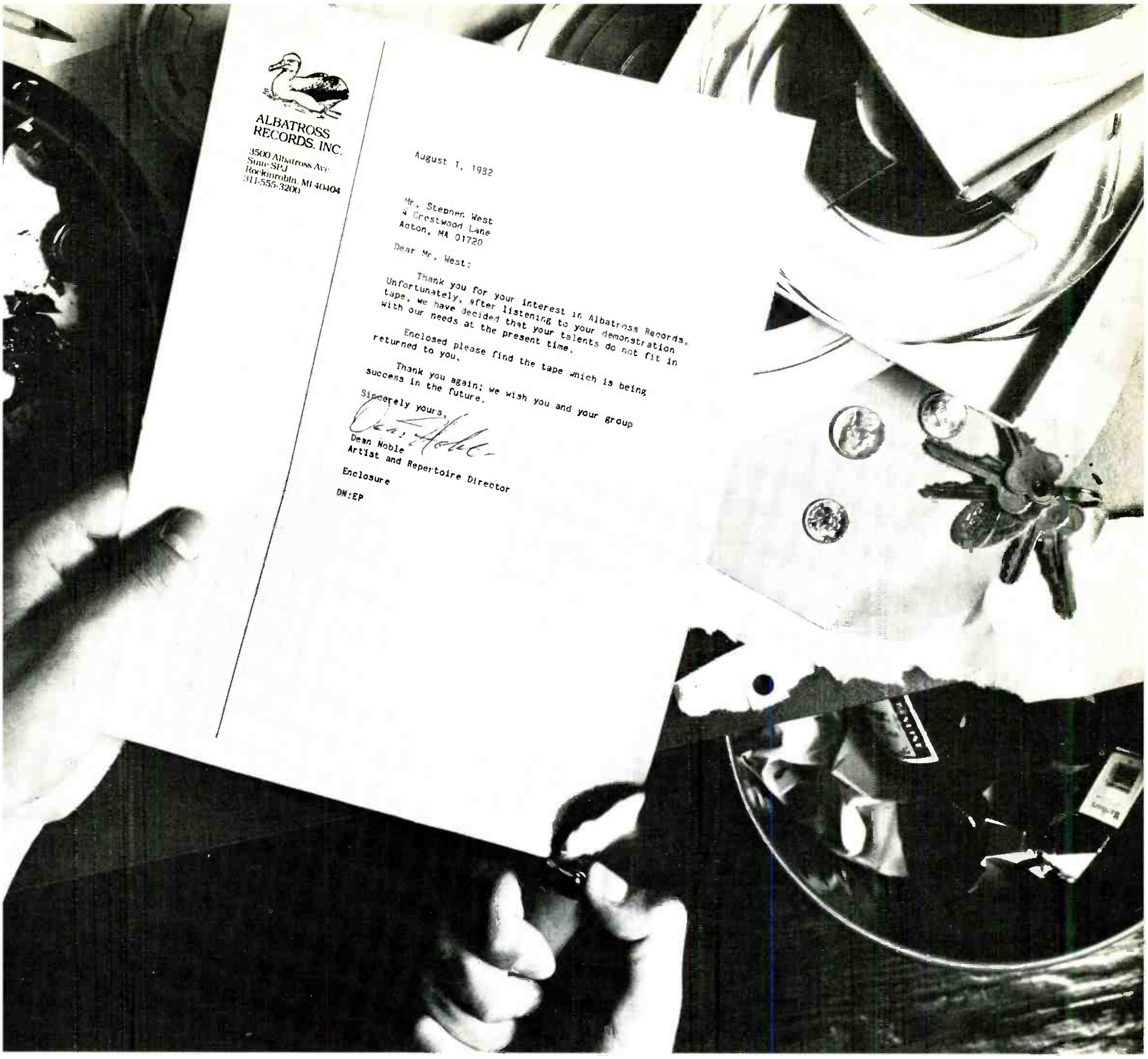
MXR Innovations, 740 Driving Park Ave., Rochester, N.Y. 14613. (716) 254-2910.

Symetrix, Inc., 109 Bell St., Seattle, WA 98121. (206) 624-5012.

Audio Arts, Inc., 5617 Melrose Ave., Hollywood, CA 90038. (213) 461-3507.

Also check out the *Pro Audio Yearbook*, Sagamore Publishing Company, 1120 Old Country Road, Plainview, NY 11803; and the *Modern Recording & Music Buyer's Guide*, published annually.

—Bruce Bartlett
Contributing Editor
MR&M



"Thank you for your interest in Albatross Records. Unfortunately..."

Few things in this world are as irritating as a polite rejection letter.

But maybe one of these days someone will like your demo tape. And instead of a form letter, you'll get a phone call.

At dbx, we know it's the quality of your music that will make you successful. But we can't help thinking that the quality of your tape recording will play a part, too. After all, how will they know how good you sound if the tape doesn't really capture your sound?

So we have a suggestion: dbx tape noise reduction.

For a modest investment, you can add a dbx professional component to your tape system and start sending in demos that sound like the real thing.

Because dbx doesn't just reduce noise. We *eliminate* it. You can record, mix-down and overdub, again and again, without any build-up of tape hiss. Each generation sounds just as clean as the first. And you'll end up with a final mix that sounds better than you've ever sounded on tape before.

Now of course, we can't guarantee that this will make you a hit.

But it will sure make you sound like one.

Visit your authorized dbx professional dealer for a look at our full line of equipment. Or call or write dbx, Incorporated, Professional Products Division, 71 Chapel Street, Box 100C, Newton, MA 02195 U.S.A. Tel. (617) 964-3210. Telex: 92-2522.

dbx® Professional Signal Processing

Circle 12 on Reader Service Card

Circle 14 on Reader Service Card

Three Out of Four *Ain't* Bad!

I am an electronics technician and sound engineer. On many occasions, your answers to questions in "Talkback" have been helpful, but now I have a few questions of my own:

1. Where can I get Carling switches? MXR uses them in effects boxes, but I have been unable to locate an outlet for them.
2. What is the address of Ampeg?
3. Who makes high-quality volume control pots?
4. What is the address of Cambridge Audio, formerly of Cambridge, MA?

Incidentally, I have built both your Dual Limiter and Hot Springs reverb. Both work perfectly, and I use them constantly in my live sound and recording work. Thank you for useful and cost-effective projects.

—M. Compton
Tele-Tech Communications
Yelm, WA

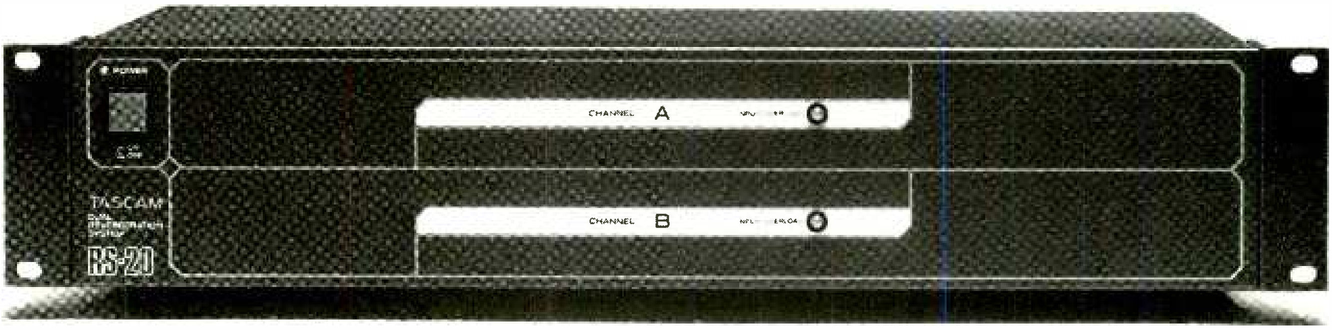
We researched your questions and came up with the following answers:

1. *Carling switches can be found at your local sales office in Seattle: Electronic Component Sales, (206) 883-6690. Their main office: Carlingswitch, Inc., 505 New Park Ave., W. Hartford, CT 06110, (203) 233-5551.*
2. *Ampeg address: Ampeg, 105 Fifth Ave., Garden City Park, NY 11040, (516) 747-7890.*
3. *There are many manufacturers out there. We'd suggest contacting one of your nearby distributors to see what's available locally. J.A. Tudor & Assoc. may be able to help. Their Seattle phone number is (206) 682-7444.*
4. *We don't know what happened to Cambridge Audio.*

(Oh well, three out of four ain't bad!)

NOTE! NOTE!

JBL & Shure Bros., Inc. are pleased to be able to offer prizes for Jim Rupert's design contest (April, 1983). More info next month.



TWEAK FREE REVERB. REVERB. REVERB.

Many reverbs come with level controls, but not the RS-20—it's tweak free. After all, your mixer has echo send and return controls, so why pay twice for the same thing. Just set the rear panel sensitivity switch, and a pair of bi-color LEDs on the front panel help you set the mixer's send level. That's it.

Limiters ahead of the RS-20 drive amps then prevent overdrive or "twangy" spring sounds caused by high energy transients inherent in plucked guitar strings, etc., so the RS-20 remains "squeak free."

This exciting new unit incorporates a proprietary design with three different sized springs on each channel. Here's why: Most conventional single-spring reverbs have poor high frequency response, and those that don't usually compensate by "shunting" high frequencies directly through their reverb with a capacitor, a short-cut that cheats you of high frequency delay. The RS-20's multiple springs extend the frequency range at least an octave above conventional units (without a shunt). Also, three springs let us scale the decay time to the frequency for superbly natural sounding reverb.

You've got to hear it—sound that rivals high quality plates and digital systems, for a whole lot less. Even at twice the price, nothing beats the RS-20 Dual Reverb.

For additional information, see your TASCAM dealer, or write TASCAM Products, 7733 Telegraph Road, Montebello, CA 90640, (213) 726-0303.

TASCAM
TEAC Production Products

DESIGNED BY POPULAR COMPLAINT.

Circle 13 on Reader Service Card

TALK BACK

Errors in Monitor Response

In the November Talkback (page 12-14), I read with interest David Smith's response to T. Young's question concerning passive Time Alignment™ in recording studio monitors. UREI is well known in the industry for use of these principles in its line of studio monitors.

I agree with most of Mr. Smith's response. However, I would like to point out a couple of errors.

The most obvious error is in the schematic given of a "Low-Pass Delay Circuit." This is actually a high-pass circuit. For low-pass response, the inductors and capacitors should exchange positions.

The second error is not so obvious. The graph labeled "Delay" for the all-pass and low-pass circuits shows a slight increase in the delay before rolling off with increasing frequency. If this figure corresponds to the Bessel response he alludes to, then the bump, which is characteristic of higher-Q filters, such as Butterworth, will not be present. The filter will exhibit uniform time delay with no increase up to the roll-off frequency. This response is most desirable for use in time-delay networks. For further information on time delay in crossover networks, I recommend reading "A Three Enclosure Loudspeaker System," Parts I-III in *Speaker Builder Magazine*, Issues 2, 3, (4/80), by Siegfried Linkwitz.

As to the audibility of delay errors, we believe them to be a major factor for consideration in the design of any high quality studio monitor system. We feel that the wide acceptance of our monitors demonstrates that other professionals think so, too.

—Brian R. Oppegaard
Project Engineer, UREI

A Different Drummer

After comparing and researching portable 4-track cassette decks, I have decided upon the Tascam 244 Portastudio. However, I have questions about the deck's compatibility with the gear I presently own.

I'll be mixing down onto a Technics RS-M65 cassette deck through a David Hafler DH 110K preamp and using a Spectro Acoustics 210R graphic equalizer. Also being employed will be a Roland TR-606 Drumatix computer drummer and the various Drum Drop albums for the percussion tracks.

Will the two methods of creating the drum tracks be reproduced cleanly on the 244? Secondly, will the outputs of the 244 be accepted by the line inputs of the RS-M65?

I've had problems with distortion in the past (exclusively in the percussion area) with the RS-M65 as a mixdown deck, using a borrowed Pioneer RT2044 and a Tapco 6001R board. The proper recording techniques were used and double checked, but upon

mixdown, the drum tracks were always reproduced distorted, while the other tracks mixed down cleanly. It took severe equalizing of the drum tracks to take enough off the highs to reduce the distortion on the RS-M65. This is the crux of my problem with the 244. Any help and advice you can give will be much appreciated.

—L.A. Safratowich
Billings, MT

Tascam suggests you take your electronic drum unit to a dealer's location and try it out with a 244 to determine if what you get is what you need. Electronic drum units, in general, are very troublesome to record—and the slower the tape speed, the greater the trouble. Practically no cassette deck will record these drum machines so that they sound the same as the original. In tests at Teac, we've noticed that the various portable recorder/mixers all fail to reproduce drum machines like the original sound. This is due to factors too numerous to describe here, but they are factors common to all recorders that use cassette tape oxide formulations and operate at speeds less than 15 inches per second.

The outputs of the 244 specifically should be no particular trouble for your Technics deck, but using a cassette deck for a mixdown in the first place is asking for trouble. Cassettes just don't have the capability to accept high-level, high frequency sounds such as those produced by percussion transients, partly due to the speed, and partly due to the fact that these tapes are *designed* for low-speed recording.

If you need to make tapes good enough for record pressing, your best bet is to concentrate on the tools appropriate for making such high quality tapes. Although the 244 may be able to give some pretty amazing results at times, it was never intended as a tool for record making, but rather a music notepad for composers and songwriters. The key here is to know if your goals are a match for the equipment you want, or whether you might be expecting too much because of the beguiling but not completely professional capabilities of cassette tape.

—Drew Daniels
Applications Engineer
Tascam Production Products
Montebello, CA

High (Humidity) Anxiety

I have a problem that is being caused by high humidity. I've had Neumanns for several years and have never had any trouble with them (due to the humidity), but I have an ongoing problem with AKGs. I maintain the equipment in an air-conditioned environment, and when I transport it from one location to another I warm it up to the ambient temperature before I take it outside. There is no problem with the deck and the other equipment, yet with the AKGs I have to try each preamp until I find a pair that has no noise. Specifically, what happens is a low-frequency intermittent noise which sounds similar to (and pegs the needle) when you touch the needle on a turntable. I keep the mics in sealed plastic bags in a camera case when I'm not using them. Would there be any advantage in opening up the AKGs and spraying them with something like an ignition sealer? Or, would it help to heat them slightly for a period of time? The temperature range is rated at -20°C to +60°C; humidity ranges from 99 percent at 20°C to 95 percent at 60°C. I've never seen an article dealing with the effects of humidity on electronic equipment. Could you give me any advice covering this problem?

—B.E. Kirkwood
Panama

Your problem *is* probably humidity-related and could be leakage across the printed circuit board and residual solder flux contamination. I'm not real keen on storing the mics alone in plastic bags, because of the possibility of trapped humidity in the bags. Perhaps you could try storing them in the bags with a silica-gel dessicant (like they ship with cameras). If you were to try sealing the printed circuit board, it probably would be desirable to heat the preamp assembly (less the microphone capsule) in a low oven to drive off any residual humidity, then clean the board with a halogenated solvent to remove any flux residue, then spray it with a sealer. I would defer to any comments made by the AKG folks, however.

—Rick Chinn
Contributing Editor
Modern Recording & Music

Recording Techniques Part XII

by Bruce Bartlett

In our last installment, we began our coverage of the monitor system by looking at monitoring philosophy, room acoustics, speakers, crossovers and monitor-speaker requirements. But simply having a pair of quality speakers doesn't guarantee good sound reproduction; you have to install, equalize, and use them properly for best results. Let's start by considering where to put them in the control room.

Speaker Placement

The closer a loudspeaker is to the walls, ceiling, or floor, the more bass it produces. Why? If a speaker is suspended in the middle of a room, it radiates low frequencies in all directions (into "full space"). But if the speaker is placed against a wall, that "full space" is effectively cut in half—the low-frequency energy is now concentrated into half the space, which boosts the lows by 3 dB. Putting a speaker in a corner boosts the bass output even more, by concentrating the low-frequency energy into one-quarter the space. The highs aren't affected much by speaker placement near a surface because high frequencies radiate mainly out front, regardless of where the speaker is placed. Check the speaker instructions for recommended placement relative to the room surfaces.

There are several ways to install monitor speakers, each with their own advantages and disadvantages. The cheapest way is to hang them from the ceiling with chains or put them on shelves at ear height. Unfortunately, these arrangements degrade the frequency response. Low-frequency sounds radiating around the speaker reflect off the rear wall, are delayed, and combine with the direct sound in front of the speaker *(Figure 1)*. This results in phase cancellations or a "comb-filter effect" in the mid-bass region. These reflections also vary the speaker's acoustic loading at certain frequencies, causing a rough response.

One way to avoid reflections from the rear wall is to mount small wide-range monitors about three feet apart on top of the console meter bridge, very near the mixing position. This is called "near-field monitoring" (developed by Ed Long). Since the speakers are close to your ears, you hear mainly the direct sound of the speakers and tend to ignore the room acoustics. This can be an inexpensive monitoring arrangement for small home studios.

Another way to prevent rear-wall reflections is to flush-mount the monitors in the wall. They should be mechanically isolated from the wall by foam rubber, fiber glass, or rubber shock mounts. That prevents sound from traveling through the wall and ceiling to the listener before the direct sound arrives through the air. If flush-mounting is impractical, you can weaken the wall reflections by placing the speakers at least three feet from the rear wall and four feet from the side walls.

The most expensive arrangement, but probably the best one, is the "live end-dead end" room treatment (LEDE™) invented by Don Davis and Chips Davis. The front half of the control room around the speakers is made very sound-absorbent (dead) by applying fiber glass or Sonex acoustic foam to the walls and ceiling. This prevents reflections from the surfaces near the speakers. The rear "live" wall behind the engineer is hard and reflective to provide ambience and increase loudness. This surface is broken up, rather than flat, to diffuse the sound.

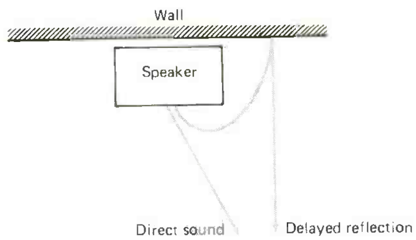


Figure 1. Speaker placement near a wall may cause phase cancellations due to delayed reflections combining with the direct sound.

Actually, an LEDE room is not necessarily all-absorptive in the front and reflective in the rear. It only needs to meet the following criteria:

(1) At the mixing position, the direct sound level should be at least 12 dB above the diffuse sound level, up till 12 to 15 milliseconds after the direct sound arrives.

(2) The listener should sit about 8 feet from the rear wall so that the reflections from that wall are delayed about 15 to 20 milliseconds. This delay makes reflections from the rear wall inaudible, due to the precedence effect (Haas effect). The reflections should be 6 dB higher in level than the diffuse sound level.

(3) After the rear wall reflections arrive, the remaining sound should decay evenly in time.

(4) The room's outer shell is solid and non-symmetrical. The inner shell of acoustical treatment is symmetrical.

(5) The speakers are flush-mounted, time-aligned™ (a trademark of E.M. Long Associates), and isolated from the mounting wall.

One of the benefits claimed for an LEDE control room is that mixes are easier and faster to do because it's easier to hear what's happening in the studio. The overall sound is clearer; stereo imaging and depth are said to be greatly improved; frequency response is flatter; boominess and overhang are reduced, and transient response is sharpened. LEDE-monitored recordings are said to hold up well on many home hi-fi speakers.

In any control room, the speakers are mounted at ear height or slightly higher so that the sound path is not obstructed by the console. A typical arrangement is shown in *Figure 2*. For best stereo imaging, align the speaker drivers vertically and mount the speakers symmetrically with respect to the side walls. Place the two speakers as far apart as you're sitting from them (about 8 feet), aim them toward you, and sit exactly between them.

Of course, wire the speakers in-phase (same polarity) to obtain a sharp center image. Also wire the speakers in correct *absolute phase* as follows: Place a microphone of standard polarity inside a kick drum. Have someone beat the drum while you watch the monitor-speaker woofer. The woofer cone should go out (toward you) the instant the drum is struck. If the opposite occurs, reverse the speaker leads.

To adjust the stereo balance, play a mono musical signal and assign it to Channels 1 and 2 in the console. Adjust the Channel 1 and 2 master faders so that the signal reads the same on the Channel 1 and 2 VU meters. Then, while sitting behind the console midway between the speakers, listen to the image of the sound between the speaker pair. It should be localized midway between the monitors—that is, straight ahead. If necessary, center the image by adjusting the "monitor trim" pots on the console, or by adjusting the volume control on the left- or right-channel power amplifier. Note that an off-center listener will hear the image shifted toward one side.

Power Requirements

Check the loudspeaker rating for recommended amplifier power. A power amp of 150 watts RMS per channel should be sufficient for use with high-efficiency studio monitors. Bi-amped systems require less total power than single-amp systems. For example, if a bi-amped system has a 100-watt RMS amp for the woofer and a 25-watt RMS amp for the tweeter, the peak power can equal that of a 225-watt RMS amp driving a passive crossover.

Some performance requirements for monitor power amplifiers are (1) distortion under 0.05 percent, (2) ability to drive any load, (3) ability to control or damp the loudspeaker vibration, and (4) ability to amplify consistently over long periods. Two valuable features are independent channel-level controls and an accurate indicator for distortion or clipping.

To avoid losing power in heating the speaker cables, put the power amps close to the speakers and use short, heavy cables with thick conductors. The low resistance of these cables allows maximum damping of the speaker by the power amplifier. Number 16 cable is recommended for cable runs less than 25 feet long, #14 for runs from 25 to 50 feet, and #12 for runs from 50 to 100 feet.

Room Equalization

Room equalization involves adjusting the frequency response of the monitor chain to flatten the speaker/room response at the listener's position. Equalization can correct for broad response errors in the anechoic frequency response of the speaker, and partly for unequal absorption versus frequency in the room treatment. Note that equalization is not effective against standing waves, because they vary with the listener's position. Neither is it a cure for poor room acoustics or narrow-band speakers. Work on the room treatment and speaker placement first, then apply as little equalization as possible. If a flat-response speaker is installed in a room with equal absorption at all frequencies, little or no equalization will be needed.

To equalize a monitor/listening room for flat response, you'll need a real-time analyzer (RTA), a pink-noise generator (usually built into the analyzer), a laboratory-calibrated instrumentation microphone, and a

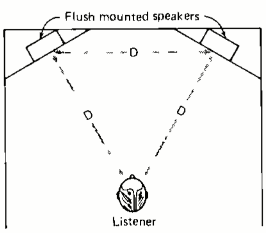


Figure 2. Recommended speaker/listener relationship for best stereo imaging.

1/3-octave equalizer (preferably a graphic type). The RTA and microphone can be rented from a sound dealer.

First, connect the equalizer between the console monitor output and the power-amplifier input (or the active-crossover input if the system is bi-amped). Set the controls of the graphic equalizer to their center (flat) positions. Put the microphone at the listener's position and plug it into the analyzer. Then feed pink noise into the equalizer input. You'll see a frequency-response curve on the RTA screen.

The final response curve after equalization should be flat from 40 Hz up to 5 or 8 kHz, then gradually roll off to about -10 dB at 16 kHz. You'll probably need to do a final touch-up by ear. Rolling off the monitor high-frequency response will make the engineer boost the high frequencies in the mix. That boost is acceptable because most home hi-fi speakers roll off at high frequencies, making the end result sound natural. A mix made on monitors tuned flat up to the highest frequencies is likely to sound dull on most home systems.

Adjust the speaker's mid-range and tweeter controls to get the desired response curve. Or, if the system is biamped, adjust the volume control on the tweeter power amplifier. Finally, flatten the curve using the graphic equalizer. Pull down the highest peaks in the curve first. Don't apply boost if you can help it.

If you need more than about 5 dB of boost or cut at any frequency, the speaker needs to be upgraded or the room treatment needs work. Add bass traps if the curve is raised at the low end. Add fiber glass, carpet, or curtains if the curve is raised at the high end. A sharp dip at a certain frequency may be due to a vibrating wall panel—stiffen it. Some dips may be caused by sound reflections off the console; these can be detected by covering the console with a heavy blanket and looking for changes in the RTA display. Don't remove those dips with the equalizer; instead, build a hood over the console to prevent sound reflections. The hood can be a thick wooden panel angeled toward the speakers (*Figure 3*).

Lacking a real-time analyzer, you may be able to equalize the monitor system roughly by ear as follows: Place a flat-response omni-directional microphone one foot from a person in the studio and record them speaking. Play the recording through the monitor system at a natural level. Equalize the speaker so that it sounds like the same person speaking "live" in the control room near the speaker. This demonstrates the meaning of "accuracy"—the reproduced sound is similar to the original sound source. As an alternative, play recorded music through some top-quality headphones, then through the monitor speakers. Equalize the speakers to sound like the headphones.

Using the Monitors

The listening level during mixdown should be maintained at about 85 dB/SPL—a typical home listening level. As discovered by Fletcher and Munson, we hear less bass in a program that is played quietly than in the same program played loudly. If you mix a program while monitoring at 100 dB/SPL, the same program will sound weak in the bass when heard at a lower listening level—which is likely in the home. So, programs meant to be heard at 85 dB/SPL should be mixed and monitored at that level.

Here's another reason to avoid extreme monitor levels: Loud, sustained sound can damage your hearing or cause temporary hearing

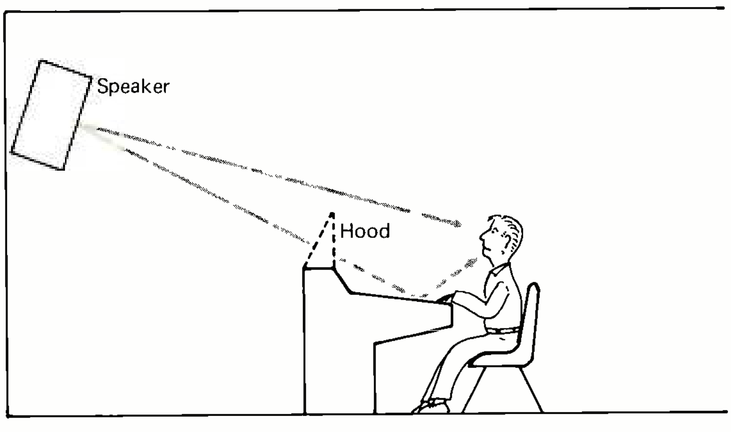


Figure 3. A console hood blocks sounds that would otherwise reflect off the console.

loss at certain frequencies. If you must do a loud playback for the musicians (who are accustomed to high SPLs in the studio), you can protect your own ears by using earplugs or leaving the room.

You can obtain an inexpensive sound level meter from Radio Shack. Play a musical program at 0 VU on the console meters and adjust the monitor level to obtain an average reading of 85 dB/SPL on the sound level meter. Mark the monitor-level setting.

While mixing, monitor the program alternately in stereo and mono to make sure there are no out-of-phase components that cancel certain frequencies in mono. Also beware of *center-channel buildup:* Instruments or vocals that were panned to center in the stereo mix will sound 3 dB louder when monitored in mono than they will in stereo. That is, the balance will change in mono—the center instruments will be a little too loud. You may have to compromise the stereo and mono mixes so that both sound acceptable.

It helps to monitor with small "cheap" speakers in addition to the big, accurate studio monitors. Single-driver mini-monitors simulate the inexpensive car radios and compact stereos that the majority of consumers listen to. First mix the program to sound good on the big monitors, then switch to the small ones to see if anything is missing or if the mix changes drastically. For example, make sure that bass instruments are recorded with enough harmonics to be audible on the smaller speakers.

Don't try to equalize the program so it sounds "hi-fi" over the cheap speakers. If you do, it will sound bad over the accurate monitors. Listeners who buy accurate stereo systems should not be penalized for their efforts! Also, the recording will lack archival value if mixed to sound good only on a colored speaker. A recording monitored on an accurate system will sound better and better on home stereos as they improve in the future.

Before doing a mix, you may want to play some familiar records over your monitors to become accustomed to a commercial spectral balance. But listen to several records, since they vary widely.

Headphones

High-quality headphones are a low-cost alternative to loudspeakers for the home studio. You should be aware of their advantages and disadvantages compared to loudspeakers.

Advantages of headphones:

- Much lower cost than speakers and amplifiers.
- No room-acoustics colorations to worry about.
- Consistent tone quality in different environments.
- Ideal for on-location recording.
- Easier to hear small changes in the mix.
- No time smear due to room reflections.

Disadvantages of headphones:

- May become uncomfortable after long listening sessions.
- The tone quality may not match that of speakers.
- You can't feel the bass notes through your body.
- Bass response varies with headphone pressure against the head.
- The sound is in your head rather than out-front.
- No listening-room reverberation is heard, so you may mix in an inappropriate amount of artificial reverberation.
- For panned signals (or for coincident-pair stereo recording), the stereo spread between ears is less than the spread between speakers.

Even though headphones may not sound like speakers, you can do your mixes over headphones to match commercial records heard over those same headphones. Then your mixes will sound "commercial" over speakers, too.

If you're monitoring as the musicians are playing, use closed-cup or circumaural headphones to block out sounds from the studio.

The Cue System

The cue system is a monitor system for musicians to use as they're recording. It consists of the cue mixer on the console, a cue amplifier, wiring to the headphone junction boxes and headphones. Musicians sometimes can't hear each other adequately in the studio (due to baffles and their own instruments), but by listening over headphones they can hear each other in a reasonable balance. They can also listen to previously recorded tracks while overdubbing.

A suggested cue system is shown in *Figure 4*. A power amplifier connected to the cue output of the console drives several resistor-isolated headphones in parallel. You may want to wire the headphones permanently to the cue lines to prevent theft.

Headphones for a cue system should be

- Durable, with metal-jacket plugs.
- Comfortable.
- Closed-cup (to avoid leakage into microphones).
- Wired mono (following the beat is easier in mono than in stereo).
- Limited bandwidth (to eliminate extreme high frequencies that may cause listening fatigue).
- Capable of producing high levels without burning out.
- All the same model (so that each musician hears the same mix and level).

Although some consoles can provide several independent cue mixes, the ideal situation is to set up individual cue mixers near each musician. Then they can set their own cue mix and listening level. The inputs of these mixers are fed from the console output buses.

Conclusion

Ultimately, what you hear from the monitors influences your recording techniques and affects the quality of your recordings. So take the time to plan and adjust the control room acoustics. Choose and install the speakers carefully; equalize them if necessary. Monitor at proper levels and listen on several systems. You'll be rewarded with a monitor system you can trust.

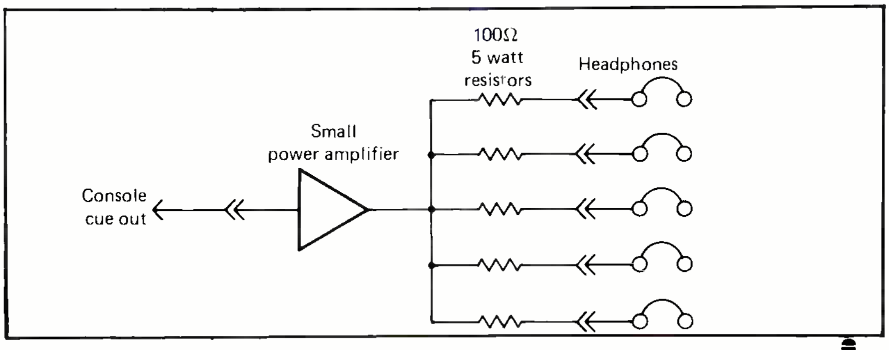


Figure 4. A cue system.

THE PRODUCT SCENE

PHASER/AUTO FILTER

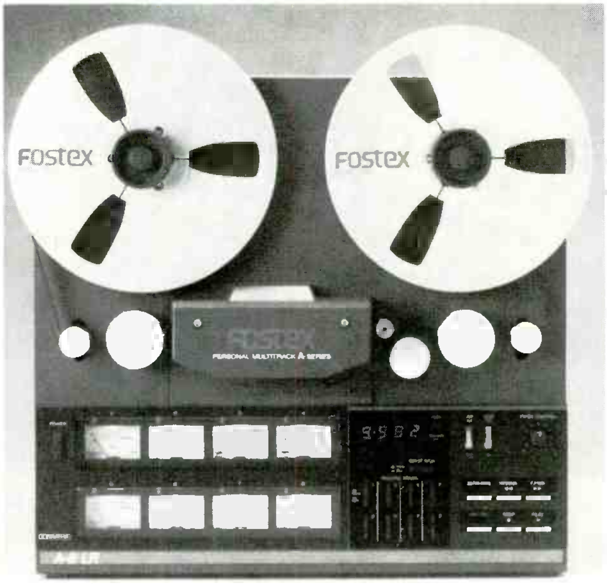
Ibanez has announced the introduction of two new products. The Ibanez PT9 Phaser uses eight phase shift networks that provide deep phasing for guitar, keyboards or bass. The sweep circuit of the PT9 provides adjustable sweep width and speed. The feedback control adjusts the level of phasing intensity.

The Ibanez AF9 Auto Filter is an automatic sliding filter that is triggered by the input signal. It provides three selectable filter types, two slide directions and two slide ranges for a wide range of sounds. The sensitivity control adjusts the trigger threshold and the peak control determines the slide width. The chosen filter's slide action may be placed wherever the emphasis is desired.

Both the PT9 Phaser and the AF9 Auto Filter are housed in zinc die-cast cases and may be powered by a 9-volt battery or an AC adapter. Also featured in both products are an LED indicator, quick-change battery pocket and the Q-1 silent electronic switching system.

Circle 33 on Reader Service Card

FOSTEX INTRODUCES A-8LR



Fostex Corporation of America has introduced the Model A-8LR, 8-channel multitrack recorder/reproducer. This model is basically an updated version of the original A-8, the first commercially available 8-track recorder using quarter-inch tape. The primary difference is that the A-8 can record up to four tracks at a time, while the A-8LR is capable of 8-track simultaneous recording. The new model is designed for special applications such as remote recording. It weighs 29 pounds and the 7½ ips operation yields 45 minutes. The A-8LR is available now at Fostex Personal Multitrack dealers and carries a suggested list price of $2,500. The new suggested retail price of the A-8 is $1,995. ***Circle 34 on Reader Service Card***

TANDBERG's TIA-3012 INTEGRATED AMPLIFIER

Tandberg's new Model TIA-3012 Integrated Amplifier can deliver 100 watts per channel of continuous power into eight ohm loads, at any frequency from 20 to 20,000 Hz, with no more than 0.02% total harmonic distortion. That means that the TIA-3012 can be used with virtually any speaker system. The TIA uses a toroidally-wound power transformer. It has low internal resistance and can deliver high current surges at peak levels. The rectifier used in the TIA-3012 can handle continuous currents of up to 10 amperes and momentary surges of up to 400 amperes. The TIA-3012 uses no electrolytic capacitors in the signal path or in the feedback circuit. The TIA substitutes polyester foil capacitors for electrolytics and ceramics.

The slew rate is 1000 volts/μsec, and each amplifier stage has its own short internal feedback loop.

The TIA is equipped with an input for compact digital (CD) disc players. There are no electrical components (transistors, capacitors or resistors) between this input and the volume control. Because the TIA uses MOSFET power transistors, there is no need for current or voltage limiting of the output.

DC voltage is controlled by Tandberg's new "Thermic Servo Loop" (Pat. pend.), which utilizes heat-difference sensing devices to detect DC offset voltage and to adjust operating parameters so as to eliminate any DC voltage at the output.

The tone control circuitry of the TIA-3012 is passive and consists of 1% calibrated resistors which provide 2 dB steps as well as switchable turnover and total tone control bypassing. The phono preamplifiers employ semi-passive RIAA equalization and conform with the new recommendations for phono equalizations of IEC amendment 4, 1976, specifying a roll-off at least 12 dB per octave below 20 Hz. ***Circle 35 on Reader Service Card***

NEW REVERB SYSTEM

Studio Technology, Inc. has introduced the Ecoplate™, a compact plate reverberation system which retains most of the features of its counterparts, the Ecoplate and Ecoplate II. The Ecoplate III is pre-tuned at the factory and features a new, shock resistant, plate suspension system which assures smooth, bright decay and eliminates tuning problems.

Specifications include: reverb time—variable from .5 to 5 seconds; signal to noise—65 dB; frequency response of reverb—80-20,000 Hz; input—-10 or +4 dBm, 10 kohms unbalanced, standard phone jack; stereo outputs—+4 dBm (+24 bBm max.), 50 ohms unbalanced, standard phone, size and weight—55" x 37" x 9", 98 lbs. The price of the Ecoplate III is $1,696.

Circle 36 on Reader Service Card

PRODUCTION AUDIO MIXING CONSOLE

Tascam has introduced the M-50 mixing console designed to perform three production functions—recording, overdubbing, and mixdown. This 12 x 8 console has multiple inputs per channel, plus reassignable submixes and complete monitoring capability.

The M-50's 12 input channels each have three switch-selectable inputs and an electronically balanced XLR mic input and a tape input. Two channels have RIAA phono inputs for playing effects library or reference disks, two have instrument inputs that serve as direct boxes, and eight have line inputs. The eight main mixing buses facilitate 1-take, 8-track recordings, and simplify normal overdubbing.

Two completely independent. auxiliary stereo mixing systems can be used for performer cues, effects sends, remote feed, or stage monitor (foldback) mixes. Control room and studio monitor facilities include a talkback mic with CR monitor muting to prevent feedback and a slate/tone test oscillator. Stereo solo in-place permits the monitors to be used to check individual channels or whole portions of a mix, while PFL (pre-fader listen) allows for level trim, pre-roll cues, or input source identification before the channel is brought into the mix.

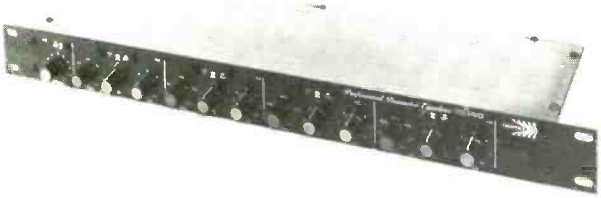
To drive long lines or low sensitivity broadcast equipment, any of the console's outputs can be patched into a pair of balanced differential XLRs that are switchable to +4 dBm or +8 dBm. Signal flow can be changed via the 178-jack rear panel patch bay.

Features also included are a 3-band sweep-type parametric EQ on each channel, and a bridge housing eight output bus VU meters with peak LEDs. Switches on the bridge can select rear panel external meter inputs for monitoring effects, tape machine returns, remote feeds, etc.

The M-50 is designed for mixing in ad agencies, video post production rooms, and multi-media facilities, as well as for small-club P.A., final film assembly, or broadcast on-air mixing.

Circle 37 on Reader Service Card

PARAMETRIC EQUALIZER



Biamp Systems announces the addition of the EQ/140, a single-channel, four-band parametric equalizer, to its line of audio equipment. Engineered to function at low levels of noise and distortion, the EQ/140 has flexible equalization and is adaptable for use in portable or fixed sound reinforcement, studio recording, broadcast, monitor speaker set-ups and home hi-fi systems.

Features include a range of ±16 dB with a deeper notch to −40 dB for feedback tuning on channels 2 and 3. The two end channels can be converted from reciprocal peaking to shelf mode, and there is a user-adjustable Q for the end filters. The EQ/140 also features balanced outputs and inputs and five LED overload indicators activated at 10 dB below clipping. It is 1¾" high in a standard rack mount. The suggested retail price is $299. ***Circle 38 on Reader Service Card***

TWO-WAY LOUDSPEAKER SYSTEM

This two-way Energy loudspeaker "Reference" Model 22 stands 35¼ inches in height (with stand) and features a tweeter design with an on-axis frequency response extending to 45,000 Hz. The Dual Hyperdome tweeter consists of two hyperbollically-shaped sections comprised of a single, chemically-treated, 1½-inch cloth dome. Special flush-mounted foam cancels any unwanted diffraction effects. The 7-inch polypropylene woofer's rubbery suspension (which has been stitched to the polypropylene) eliminates any unwanted resonances. The cabinet is internally braced in two areas to eliminate wall vibrations. The system's port, tuned to 34 Hz, is located at the front. The "Reference" model has uniform dispersion in all forward directions, and units are sold in mirror-image stereo pairs. On-axis frequency response is 30 to 45,000 Hz ± 3 dB. At 30 degrees off axis the response is 35 to 20,000 Hz ± 2 dB. Matching between stereo pairs is within 0.5 dB. System impedance is 8 ohms; sensitivity is 89 dB sound pressure level for a 1-watt input at 1 meter. The price of Energy Loudspeaker "Reference" Model 22 is $1,100, which includes the stand. ***Circle 39 on Reader Service Card***

TWO-WAY MUSIC PLAYBACK SYSTEM

JBL Incorporated introduces the new Cabaret Series model 4691, a compact two-way system specifically engineered for full-range music playback. It is ideal for installation in nightclubs, discotheques, theaters, or any application requiring high acoustic output and efficiency, controlled dispersion, low distortion, and wide frequency response (40 Hz to 20 kHz).

The system incorporates the recently developed 2370 flat-front Bi-Radial horn, a 2425J titanium-diaphragm high frequency compression driver, and an E140 15-inch woofer. The 2370 Bi-Radial horn provides expanded on- and off-axis frequency response in the horizontal plane, with a 90 degree horizontal by 40 degree vertical nominal coverage pattern to beyond 16 kHz. The use of the 2425J compression driver results in extended frequency response, as well as high efficiency and power handling.

A 1.5 kHz high-pass network blends the low and high frequencies, while switchable biamplification inputs are featured on a rear terminal panel. The 4691 may be used alone or in conjunction with the 4695 subwoofer. ***Circle 40 on Reader Service Card***

Announcing the Great Electro-Voice® Sound In Action™ PL Microphone Drawing!

GREAT NEWS— FINE PRIZES AND FREE T-SHIRTS, TOO!

If you're interested in microphones, EV has some good news for you. Between March 10th and June 10th every PL Microphone purchaser receives a free EV T-Shirt (retail value $7.95) just for the asking.

And you can enter the great EV PL Microphone Drawing at the same time (no purchase necessary).

Grand Prize is an EV PL Microphone ensemble of 13 mikes custom fitted in a tour case, lots of accessories and a limited edition EV tour jacket (total retail value over $3,700).

Seventy-five other winners will each receive a free PL Microphone.

HOW TO ENTER

Entry blanks and complete rules are available free at your nearest participating Electro-Voice PL Microphone dealer. You can also get information by writing to Electro-Voice, Inc., Department 615, 600 Cecil St., Buchanan, MI 49107 (enclosing a stamped, self-addressed envelope will help speed a packet to you). Complete the entry blank and mail it on or before June 10, 1983.

a gulton company

FREE T-SHIRTS

Every purchaser of a PL Microphone during the drawing qualification period will receive a free T-Shirt by returning a request form (at your EV dealer's) with the original sales slip and your new mike's warranty card. Your free EV T-Shirt will arrive in 4 to 6 weeks.

TRY OUR PL MICROPHONES

EV is sponsoring the drawing and offering free T-Shirts for one simple reason—we want to show you the great line of eleven PL Microphones.

PL Vocal and Instrument Microphones are accepted worldwide as high quality and reliable tools for working musicians. EV builds them with the expertise acquired from more than 50 years of U.S. production of broadcast, recording and music performance mikes. EV backs their products with solid warranties and expert speedy service.

The full line of EV PL Microphones is presented on the following page. Drop by your EV dealer soon and check out the eleven PL Microphones—and don't forget to enter the PL Microphone Drawing!

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Meet the Great PL Microphones from Electro-Voice®

PL VOCAL MIKES

PL80 Designed with computer-assisted FFT technology, the PL80 super-cardioid has better gain-before-feedback than virtually every other dynamic microphone currently on the market. Its crisp highs and bass-boosting proximity effect enhance a performer's voice without compromising the vocal quality. The PL80 has fast become one of the world's most desired vocal microphones.

PL88L Outstanding performance at an "entry level" price. This dynamic cardioid offers a good gain-before-feedback and voice-tailored frequency response with just the right amount of close-up bass boost.

PL91A This popular cardioid microphone has been refined and redesigned specifically for performers. It features good strong bass boost when held close, superior highs, and an "open" sound over the entire range that makes it a joy to work with.

PL95A As the concert mike of choice by top performers, the PL95A offers superior gain-before-feedback. The wide, linear frequency response and controlled bass boost when held close have made it an international favorite.

PL76B The hottest vocal mike to hit the market in years, this sophisticated cardioid condenser will change your ideas of what a mike can do for your voice. Battery life is an incredible 3000 hours.

PL77B Take the PL76B, add phantom power, a two position bass contour switch, and you have this great vocal mike.

PL INSTRUMENT MIKES

PL20 One of the most coveted microphones in the world, the PL20 is a Variable-D® dynamic super-cardioid created for critical recording and musical sound reinforcement applications.

PL5 A professional instrument mike for super high SPLs, often used for close miking of amplified guitar, bass or synthesizers.

PL6 This super-cardioid is a Variable-D® design which allows directional miking with the characteristic close-up bass boost of directional mikes. This minimizes feed-back problems and unwanted sound leakage from other sound sources.

PL9 EV believes this is the best omnidirectional instrument mike on the market. Very flat frequency response over a wide range. Put inside of drums at high SPLs or for high quality recording and capture it all.

PL11 An instrument mike that can double as a vocal mike, the PL11 is another of the fine Variable-D® family great for pick-up of brass, reeds, and overhead percussion.

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Please turn page for exciting announcement . . .

MUSICAL NEWSICALS

SYNTHESIZER EQUIPMENT

Syntauri Corporation, one of the pioneers in the field of programmable microcomputer-based synthesizer systems, has announced a significant new software package for their alphaSyntauri system. The new program is called Composer's Assistant and provides the alphaSyntauri user with automated transcription and hard-copy score printing. Besides an alphaSyntaurai keyboard (with models ranging from a low-priced 4-octave keyboard up to the 5-octave Composer's Workstation) and its associated Apple II computer system (with 48K of memory, language card, at least one disk drive, and a monitor), Composer's Assistant requires a dot-matrix printer such as the IDS or the Epson MX-80 to produce hard-copy output. In use, the musician or composer performs his piece on the alphaSyntauri keyboard, playing against the built-in metronome (10 to 280 beat/minute range), while the Composer's Assistant program analyzes the keyboard inputs and generates a transcription in conventional music notation. For scoring and orchestrating applications, the system can accommodate up to 16 separate tracks, each of which may be polyphonic, and can display or print them either as an orchestral score or as separate instrumental parts. Composer's Assistant can adjust for performance timing variations, since personal playing styles can vary significantly, and can be made to ignore accidental notes produced during difficult passages. The program resolves to 1/16 notes and rests, has ¾, 4/4, and free time signatures, special triplets mode, and provision for making measure ties and for transposing key signatures. In addition, the system has built-in text editing facilities to allow addition of expression markings, special instructions, lyrics, and chords to the score as displayed on the video monitor or printed on the printer.

Circle 44 on Reader Service Card

Star Instruments, Inc. have introduced an intriguing new electronic wind synthesizer called the Starwind. The new instrument is a self-contained, battery-powered, hand-held instrument weighing only one pound. The unit combines keyboard control of pitch with breath control of loudness and tone color, making the Starwind very easy to play, yet capable of a wide range of expression.

Circle 45 on Reader Service Card

PERCUSSION EQUIPMENT

MXR Innovations have become the exclusive U.S. distributor for the English electronic percussion device known simply as The Kit. The Kit produces the sound of a four-piece drum kit complete with cymbals in a four-pound electronic package. All sounds are played in real time by the musician tapping on touch-sensitive pads; four large pads are provided for kick drum, snare drum, and high and low tom-toms, along with three smaller pads for open and closed hi-hat and a variable crash/ride cymbal. The hi-hat may additionally be driven from a built-in rhythm unit with variable tempo and time signature. Each of the unit's sounds has its own level control for full control of the overall, mixed output from the unit, plus individual outputs are provided for recording application or for external processing or mixing of the sound.

Drummers have always seemed to be at the mercy of their hardware. Recognizing this, Yamaha has introduced Yamaha System Hardware, designed in consultation with many of today's top professional drummers. Emphasis was placed on solid support and easy set-up. For example, the Yamaha boom arm has extremely stable stand legs and a lightweight upper section that does not require a counterweight; additionally, the boom may be inserted as is into the main pipe for instant conversion to a regular cymbal stand with 28-position tilter adjustment. Other innovations include nylon bushings for height adjustment joints (to provide exceptional holding power while virtually eliminating pipe distortion at the clamp), an all-round ball clamp with a resin compound ball for superior holding power, infinite angle adjustment and single-bolt tightening. Yamaha also has an interesting tom base which has a third clamp for a cymbal or cowbell, or for a third tom holder. Like Yamaha Systems Drums, Yamaha System Hardware is divided into three series, each with a different design philosophy and pricing structure. The Nine Series is the top of the line, with double leg bases that are massive and stable enough to stand up to the strongest drumming and most punishing treatment. The Seven Series is built to the same standards of design and construction, but with both the size and the prices scaled down. The Five Series incorporates the best design features of the other two series, but in a lightweight, basic format.

Circle 46 on Reader Service Card

GUITARS

Fender continues to explore the lower end of the electric guitar market by expanding their successful Bullet line of guitars from two models to seven. The two existing models, the Bullet and the Bullet Deluxe guitars, have been redesigned, while three new guitar models and two new basses have been added to the line. The Bullet line is designed not only to be an attractive, professional-quality instrument with the Fender name, but one which is also affordable to a wider market than can afford a Telecaster or Stratocaster. Features common to all Bullet models include Stratocaster-style bodies of solid poplar wood, four-bolt detachable necks of hard rock maple, Telecaster-style headstock, and single volume and tone controls. Two of the guitar models feature aluminum bridge/pickguards; one of these models has a single humbucking pickup with coil splitter switch, while the other has two single-coil pickups. The other three guitar models have Stratocaster-style bridges and laminated pickguards; pickup configurations include twin single-coil pickups, triple single-coil pickups with 5-position pickup selector switch, and twin humbuckers with 3-position selector switch and two coil-splitter switches. On the bass side of things, Fender has long-scale (34") and short scale (30") models, each equipped with a chrome-plated, Precision Bass-style bridge and a split, single-coil pickup with oppositely-phased sections for hum rejection.

Circle 47 on Reader Service Card

New from St. Louis Music Supply is a very pretty Alvarez guitar in which intricate carving replaces the traditional soundhole. The new model is the Alvarez Artist 5051, and features a solid spruce top, mahogany back and sides, and rosewood fingerboard and bridge. The intricate floral pattern is actually carved by a laser, and is designed to admit the same amount of air to the guitar body as a traditional soundhole, thus allowing the same volume and projection from the guitar—although the sound is more reminiscent of an F-hole guitar than a round-hole.

Circle 48 on Reader Service Card

A new cutaway acoustic/electric guitar has been added to the Adamas line from Ovation Instruments. The Adamas Cutaway guitar has the same features as the Adamas and Adamas II models, including a carbon/graphite composite top, Kaman Bar neck reinforcement, a resin-impregnated, reinforced walnut fretboard, and Ovation's patented stereo acoustic/electric pickup and preamp system built into the

guitar's body.

Circle 49 on Reader Service Card

The Guitarras del Brazos line from MCI, Inc. has a new top model, the GBM-100. The new model has a top of either solid German spruce or solid cedar, depending on the musician's preference in tonal color, along with back and sides of solid rosewood. The mahogany neck has an ebony fingerboard and is equipped with gold-plated tuning machines. An ebony bridge is also standard.

Circle 50 on Reader Service Card

Phased Systems have become well-known for their D'Mini series of 2/3 and ¾ size guitars. The latest offering in the D'Mini Series is the new D'Mini Bass, which has already picked up an impressive endorsement from renowned studio bassist Corol Kaye. The D'Mini Bass is available in either regular bass or piccolo bass configurations with a 25½" scale length. Besides being a faster, easier instrument to play, the D'Mini bass weighs only 6 pounds, making it comfortable to play for long periods of time. Mahogany and ash bodies are available, equipped with bolt-on, 22-fret, maple neck, and split single-coil pickup.

Circle 51 on Reader Service Card

GUITAR STRINGS

A new name in guitar strings is a well-known and respected name in guitar parts and hardware, Schecter Guitar Research. Schecter spent two years testing and refining their strings before releasing them for sale; by the time they were done, they felt justified in naming their strings The Finest™. Schecter currently offers electric guitar strings in three gauges (.009, .010, and .011 first strings), although they plan to expand their line to include acoustic strings and bass guitar strings in the near future. Schecter's strings were designed specifically to set up fast yet to hold their tone and last much longer than competitive brands of strings, and this was confirmed in use tests with selected guitarists throughout the U.S. and Canada.

Circle 52 on Reader Service Card

Following up on the overwhelming success of their XL Reds Bass Strings, the J. D'Addario company has introduced a line of XL Reds Guitar Strings which share the same copper-coated, round-wound construction of their bass counterparts. Like the bass strings, XL Reds guitar strings offer an exceptionally bright sound with what is described as a "unique piano-like brilliance"; the sound has also been described as being closer to a pure acoustic tone than any other electric guitar string. D'Addario XL Reds Guitar Strings are available in regular, light, and extra light gauges.

Circle 53 on Reader Service Card

GUITAR AMPLIFIERS

Fender has revived the name of one of their most highly regarded amplifier models for a new line of guitar amps that combine the sound of all-tube circuitry with contemporary, state-of-the-art features. Fender's new Concert Series are 60-watt tube designs that incorporate switchable input channels, an effects patching loop and sophisticated equalization, along with other features. A clean channel is provided for rhythm work, along with a switchable lead channel with total control over the amp's gain structure—thanks to a volume control which sets the amount of front-end overload, a Gain knob for intermediate-stage drive, and the Master volume control which sets the overall level delivered to the output stage and speakers from either input channel. The effects loop provides separate Send and Return level controls to allow optimum matching of signal levels for any type of outboard effects devices; the signal sent to the effects loop is taken after the amp's front end so that a pre-distorted signal goes to the effects boxes or foot pedals. Besides bass and treble controls, the Concert Series amps have a Presence control and a

Midrange control which has a "pull for boost" switch that was designed to increase drive and sustain on the top strings without sacrificing brightness. Other features include reverb (accessible from both the Normal and Lead channels), a line output for recording or PA applications, and a footswitch with LED indicators for channel switching and reverb. Like all current Fender amps, the Concert Series features lock-jointed pine cabinets, 14-ply birch plywood baffle boards and an improved type of Tolex covering said to be virtually tear-proof. Available configurations include 1 x 12", 2 x 10", and 4 x 10" cabinets with either special-design Fender speakers (standard) or EV speakers (extra-cost option).

Circle 54 on Reader Service Card

MICROPHONES

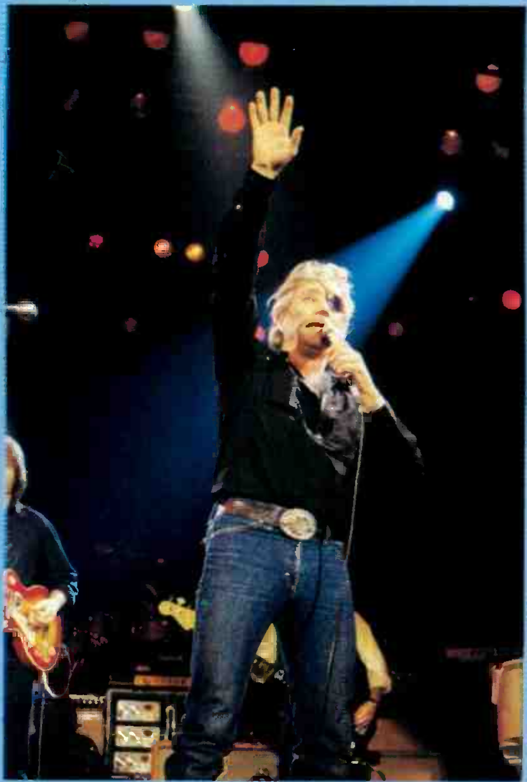
Ibanez recently introduced three microphone models specifically designed for instrument miking applications. Most popular vocal mics are rather poorly suited to instrument miking for the same reasons that they are popular as vocal mics, the upper-midrange presence peak and the low-frequency proximity effect which give clarity and fullness to voices. ideal for acoustic guitar, piano, woodwinds, and drum overhead use. And lest the vocalists among us feel neglected, Ibanez has the IM60 and IM66 models which are rugged dynamic cardioid mics designed for The Ibanez IM70 is a super-cardioid dynamic mic with a lightweight cartridge diaphragm for a combination of fast transient response and high signal handling ability along with ruggedness; recommended uses include snare drum, high tom-toms, brass instruments and guitar amps. The IM76 uses a shock-mounted cartridge diaphragm with a high-compliance edge and double dome, making it suitable for percussion use, particularly on lower-frequency sources such as kick drum and floor toms. The IM80 is a cardioid electret condenser mic with an exceptionally wide frequency response, making it vocal and instrument amplifier miking.

Circle 55 on Reader Service Card

New from Samson Music Products is a cost-effective wireless mic system designated the SMX-1. The system uses a hand-held mic with built-in transmitter with an Off/Standby/Operate switch and an indicator which lights up when approximately one hour of battery life is left. The receiver is AC powered and has LED meters for both RF signal strength and audio level. Frequency response is specified as 30 Hz to 9 kHz, ±6 dB, signal-to-noise ratio is 90 dB, and total harmonic distortion is 1.5 percent at 400 Hz. Two operating frequencies are available, 49.830 MHz (Channel 1) and 49.890 MHz (Channel 5), so that two systems may be used simultaneously. For more demanding applications or difficult installations, a version is available with a dual antenna diversity receiver to help eliminate dead spots in the reception; this system is known as the TR-1020M.

Circle 56 on Reader Service Card

VOLUN



erin morris

TEER JAM IX

"It should be the most confusing show, but everyone is so relaxed that it all goes smoothly. All the people involved are so competent and secure. They don't get in each other's way because they all know their job and are working at their full capacity." This is Johnny Rosen's appraisal of Charlie Daniels' Volunteer Jam IX. Rosen owns Fanta Professional Sound the company that broadcasted the Jam.

It is the longest, most lavishly staged event of its kind, and has become a model of a high energy yet trouble-free festival. Here's a report on how it was all done this year.

Charlie Daniel's first Volunteer Jam was held in 1974 when he gathered a few of his performing friends together at the War Memorial in Nashville to record two live tracks for his upcoming album, *Fire On The Mountain.* Those friends included Dickey Betts, who was then with the Allman Brothers, and Toy Caldwell, Paul Riddle and Jerry Eubanks from the Marshall Tucker Band. After "Orange Blossom Special" and "No Place To Go"—the two album tracks—were cut, the group began to jam. The audience response was so great that Daniels decided to make it his annual homecoming event. The show is "Volunteer" for two reasons: the artists volunteer their services, and Tennessee is nicknamed the Volunteer State.

Past Jams have boasted such guests as Billy Joel, George Thorogood and the Destroyers, Ted Nugent, Willie Nelson, Sea Level, Wet Willie, Bonnie Bramlett and Delbert McClinton.

This year's Jam was held Jan. 22 at Nashville's Municipal Auditorium, its home for the past seven years. The 10,000 fans who filled the venue at $15 a ticket got over eight hours of non-stop entertainment for their money. The Jam has become such an institution that this year's bash, Jam IX, sold out in less than two days—by mail.

Daniels' homecoming benefits such charities as the T. J. Martell Leukemia Foundation and area schools for handicapped children. The event is produced by Cumberland Concerts and Sound Seventy Productions.

Daniels foots most of the bills for the event, including rental of the arena and its downstairs exhibition floor, the artists' travel and lodging expenses, concert security, and food and drinks backstage for more than 2,000 invited guests, most of them connected with the music industry.

To get a Jam show together, Daniels sends out invitations to artists and bands all over the country. This year more than 20 major rock, jazz and country performers accepted the invitation. Their names are kept secret until the performers hit the stage.

This year's line-up included James Brown; The (Dickey) Betts, (Jimmy) Hall, (Chuck) Leavell, (Butch) Trucks Band; Carl Perkins; Johnny Lee; Larry Gatlin and The Gatlin Brothers Band; Quarterflash; Dobie Gray; McGuffey Lane; Steve Walsh and Streets; Dr. Hook; the Winters Brothers; Grinderswitch; Papa John Creach; Woody Herman; Roy Acuff; Boxcar Willie; Tanya Tucker; members of the late Marty Robbins' band, and the Charlie Daniels Band.

Rehearsals were held the Thursday and Friday evenings and Saturday afternoon just before the Jam. Each act (that was available for the rehearsal) was allowed time to run through the three or four numbers it would perform to allow the house sound, the video company and the recording truck to get some idea of where everyone would be working on stage. But a vague idea was all they got, for during the actual performance excited band members had a tendency to forget where they were to plug in. Also, since Daniels loves to spring last minute surprise guests, quite a bit of improvisation was required.

However, with the help of an elaborate intercom system and several video feeds from E. J. Stewart Video Production out of Primos, Pennsylvania, the Los Angeles Record Plant truck (recording the Jam) and Zinn Audio (doing house sound) were able to stay abreast of which instruments were coming in on which channels.

Zinn, which works out of Carmel, Indiana, is the company Daniels generally uses for his road show. Their equipment was supplemented with equipment from Apex Audio in Chelsea, Michigan. John Cooper acted as chief engineer for the house sound.

Two Custom Audio Electronics 36 x 8 x 2 boards were used for the event, each one equipped with three-band, full parametric equalization with high- and low-pass filters. One console was used as a sub-console to the other. The left and right outputs of the board went on through a pair of UREI 537 one-third octave graphic equalizers and then on to a pair of dbx 160s for metering purposes. From there the signal passed through two UREI four-way electronic crossovers and finally on to four dbx 162 compressor limiters for line drive to the power amps on stage.

The four-way speaker system stacked on stage was designed by Zinn. Making up the design were two JBL 2225H 15-inch front horn-loaded low-frequency speakers, two Electro-Voice EVM-12L 12-inch full range speakers, one JBL 2445 4-inch compression driver/pattern control horn and one JBL 2425H 1.75-inch compression driver/pattern control horn. The system has a frequency response of 63-18 kHz ±3 dB. The throats were also built by Zinn.

Six of the speakers were stacked on the left and right sides upstage for front fill, and two were stacked on both sides of the back of the stage for rear fill. In addition to the stacks, a three-way cluster was flown. The amplifiers used to power the speakers were Crown and AB systems.

The effects used for the show were a Lexicon Super Prime Time programmable delay and a Ursa Major space station.

Cooper said that several bands brought their own engineers with them, among them Dr. Hook, Larry Gatlin and The Gatlin Brothers Band, Quarterflash, Steve Walsh, McGuffey Lane and Dobie Gray.

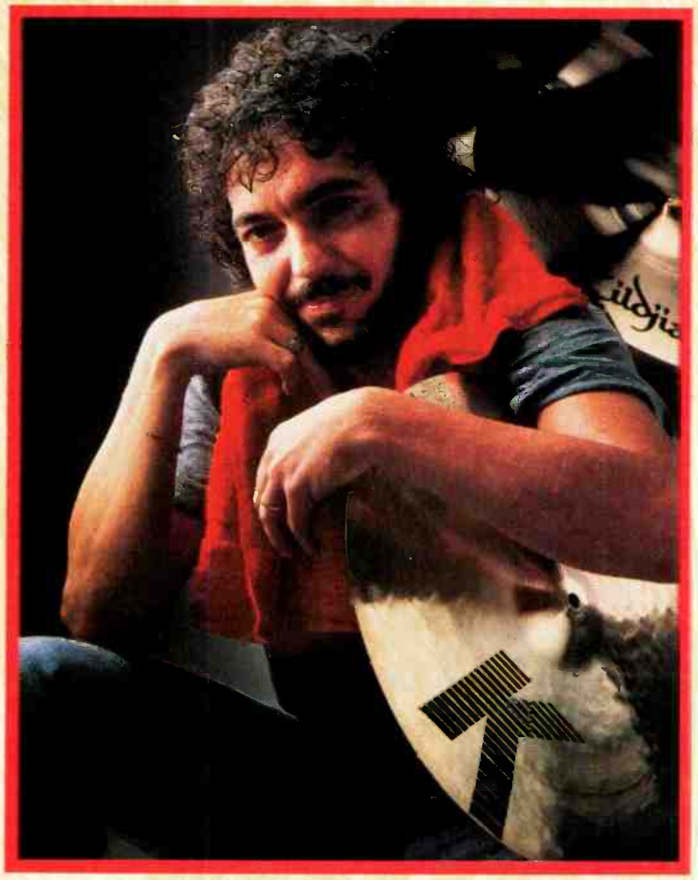
Zinn's Scott Holloway ran the stage monitor system. It employed two Custom Audio Electronics 26 x 10 consoles with two-band shelving equalization. The system was run through external equalization racks, which housed eight UREI 529 or 539 one-third octave all-cut graphic equalizers and four Orban 622B stereo four-band parametric equalizers.

There were three types of monitor cabinets in use, and all monitors were tri-amped. The first cabinet had one Electro-Voice EVM-12L, a JBL 2482 compression driver on a JBL 10-inch lens assembly and a JBL 2405 tweeter. The second used an Electro-Voice EVM-15L and an EV Force 10 10-inch speaker. For the high end, a JBL 2441 driver on a JBL 10-inch lens assembly was used. The third cabinet was used mainly as a drum monitor. It contained two EVM 15Ls, a JBL 2482 on a 10-inch lens and a JBL 2420 on a 10-inch lens. In all, there were 14 monitors on stage.

To cut down on the time needed for set changes, Daniels provided most of the musical equipment on stage. All keyboardists set up their equipment ahead of time and left it there throughout the concert. Two drum kits were on stage at all times; one set was Daniels' drummer's, the other a complementary set provided by Ludwig. Daniels also supplied a variety of guitar amps including Marshall speaker bottoms, Fender Twin Reverbs and Super Reverbs, and Mesa Boogies, so that no player was required to bring his or her own on stage.

Comprising the stage microphone system were Electro-Voice DS-35s

STEVE GADD. HOT ON ZILDJIAN.



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As you can imagine, between his touring and recording, Steve's not the easiest guy in the world to pin down.

But he did stop for a breather the other day and we got a chance to talk with him.

On Practice. "I've been playing since I was a kid. As long as I keep my muscles loose, I don't have to practice a lot every day. When I do practice, I just sort of let things happen naturally and then later on try to work it into my playing. Like on '50 Ways to Leave Your Lover'... I used my left hand on the high hat for the whole section—it was a little thing I'd been practicing and it just worked out."

On Control. "Sometimes I use light, medium and heavy sticks to do the same drills because the sticks affect my muscles in different ways. You have to use your hand and arm muscles differently to control your playing. It's a subtle thing but it helps me tremendously."

On Effects. "After I graduated from Eastman, I played in a rock 'n roll band. It was keyboard, bass, drums and a lot of homemade stuff. I bought 6 big artillery shells, sawed them into different lengths and hung them on a rack that I built. I'd use them for the free sections in the music."

On K's. "Art Blakey gave me my first set of K. Zildjian's a long time ago. I love the feel of them. There's something about the way the stick reacts to the surface...it almost becomes part of the cymbal. They're not cold or edgy. They have a very warm and deep feeling. They've got real character. I use a 20" Ride and an 18" Crash Ride with 14" Hi Hats for recording and live sessions."

On A's. "I love to use A. Zildjian's when I play rock 'n roll. When I want to play louder, I add a 16" Thin Crash and an 18" Crash Ride for a full crash sound. The bells on the A's really project the sound in a clear natural tone."

On Zildjian. "Zildjian to me is the foundation. I play Zildjians because that's what's in my heart. I love the sound, the feel, the history...I love the quality and the status of a Zildjian."

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for vocals; a Shure SM-57 for the Leslie organ; Shure SM-57s for all electric guitar amps; Sennheiser 421s for percussion instruments and horns, and an SM-57 for harmonica.

Both drum sets were miked the same—Electro-Voice RE-20s for the kick drum; Shure SM-57s for the snares; Sennheiser 421s for the toms, and AKG 451s for the high-hat and as overheads.

All keyboards were run through Sescom direct boxes, and basses were run through a Westlake Audio direct box.

Each set change took only two to five minutes. There were approximately eight sound people on stage between sets and an additional two to three roadies from each band. Also assisting were four union stage hands. Wayne Smith was the stage manager for the Jam; he coordinated the set changes with two assistant stage managers.

The signal was split three ways at the stage and went directly to the house consoles, the monitor consoles and to the Record Plant recording truck, sent in from Los Angeles to tape the Jam.

Co-producing this effort was Daniels' record producer, John Boylan, and chief engineer, Paul Grupp. Working with Grupp was a crew of four from the Record Plant and Michael Sanderson, Daniels' chief engineer.

Sanderson's duties involved mixing all the tracks to two-track and sending that mix to the radio broadcast truck, the video truck, and to various spots throughout the building. Sanderson also assisted Grupp throughout the evening.

The Record Plant crew included Jack Crymes, the maintenance technician, and Bill Freesh, who took cues over headphones from Zinn Audio on stage to let Grupp know who was plugging in where. Mark Eshelman worked on stage and had a headphone system directly linked to Grupp at the board and to Freesh's headphones. Eshelman made sure that the tie-in with the sound company was working sufficiently and also took notes on which instrument or vocal was plugged into which line. Gary Singleton worked inside the truck, making certain that the two 3M Model-79 24-track tape machines were always getting the signal clearly. Only one machine taped at a time, however. It was Singleton's duty to see that when one machine began to run low on tape, the other machine was turned on. This was to ensure that one full take of any song was taken down on one or the other tape machine. Grupp uses Scotch 206 tape because he says the sound is always consistent and it is the hardiest tape he has found.

Record Plant brought several of their own microphones to the Jam since their requirements differed somewhat from those of the house sound system. An AKG-452 and an Electro-Voice CS-15 were used for overhead mics on the left and right drum sets, respectively. Also, microphones were placed in various spots around the hall to pick up sounds for the live recording. These included two Neumann U-87s in the back of the hall; two Sennheiser shotguns pointing outward from each side of the stage to pick up audience noises, and two RCA 77 ribbon microphones pointing inward at stage left and right to pick up stage ambience.

E. J. Stewart Video ran two video sends through the Record Plant's Sony closed-circuit video system so that Grupp and associates could see what was happening on the stage and didn't have to rely solely on verbal instructions. Record Plant also brought two of its own video cameras and placed them at strategic points around the stage. With the four video sends, the Record Plant crew was able to see what was happening at all points on the stage at all times.

There was a total of 66 inputs coming into the Record Plant truck. Using an API console (designed by Record Plant) and a Soundcraft 16 x 8 console, Grupp mixed the inputs down to 23 tracks. He used the 24th track for the SMPTE time code that the video company was sending. This code would later be used to synchronize the mixed audio tape of the Jam to the edited video tape.

The Soundcraft board was used mainly to pre-mix the horns, for stage and audience ambience mikes, and for various percussion instruments.

Monitors in the truck were JBL 4320s and a pair of Yamahas for backup.

No effects were used on the audio recording of the event. Explained Grupp, "First of all, there are not enough tracks for any effects, and since so many things are changing at once, it would be suicide to add more things to adjust."

Fanta Professional Sound of Nashville handled the live radio broadcast for the event. Fanta took a two-track feed from Record Plant's recording truck and integrated commercials and backstage interviews with artists. Johnny Rosen, Fanta's owner, was in charge of the broadcast.

The show was aired over the "Volunteer Jam Network" of six stations throughout Tennessee: WRVU-FM and WWKX-FM, Nashville; WZXR-FM, Memphis; WIMZ-FM Knoxville; WSKZ-FM, Chattanooga, and WBGY-FM, Tullahoma. Each station had a home-base announcer as well as a DJ on-site at the Jam.

Aiding communication among the broadcasters was a very sophisticated earphone system. The DJs received a live radio mix in one ear and special instructions in the other. In addition, they were able to speak to their cohorts back at the station. Guests being interviewed also had earphones over which Rosen could issue instructions.

In the Fanta truck, five Ampex

tape recorders were kept running at all times. Two took down the live music, two recorded the completed radio show and one had the pre-recorded spots.

The broadcast was not delayed in any way for bleeps. Whatever happened on stage was what went out over the airwaves. There was a set of lights (red, yellow and green) on stage to make Daniels aware of the status of the broadcast.

No problems were encountered with the broadcast. However, the radio signal had to go through WRVU-FM, the Vanderbilt University radio station, before going on to WWKX-FM. The signal was hard to get through the campus station because they don't have a full-time technical staff. That problem will be eliminated next year, when a technician will be placed on duty specifically for the broadcast.

At press time, there were no specific plans for the syndication of the radio broadcast.

John Boylan, Charlie Daniels' record producer since the "Million Mile Reflections" album, says that there will probably not be an album made specifically from the Volunteer Jam IX. He notes, however, that for the 10th Anniversary Jam, there will probably be a series of albums issued that will be compilations of past Jams and will be in separate sleeves. They will be divided up and marketed according to musical format. "In the past, the albums have been too eclectic," Boylan observed. "When a hard-core country fan hears a Crystal Gayle cut from the album and goes to buy it but finds that Ted Nugent is on it also, we end up losing some sales."

Boylan says he picked the Los Angeles Record Plant to record the concert because he always mixes there and prefers the API boards and JBL monitors their truck is equipped with.

He explains that there are many differences between recording a regular live concert and the recording of the Volunteer Jam. First, he says, if the live recording is of one band, he will record the act several nights playing at one location. This, he adds, allows plenty of performance cuts for an album. However, with the Jam there is practically no rehearsal time, and only one chance for a take of any song.

Secondly, the Jam music is by-and-large improvisational. The only schedule adhered to is the one that states when an artist goes on stage. After that, anything can happen. Surprise guests take the stage, artists appear with other artists, band members interchange and technical problems add their own variations.

"The Volunteer Jam is the most difficult live recording situation I've ever been in," Boylan asserts. "Mistakes can't be overdubbed because the house ambience would then be missing."

Boylan budgeted the recording of the most recent Jam at a low $15,000—to cover the costs for tape, engineers and the truck. Since it is a charity function, artists are not paid for their time. However, the American Federation of Musicians is paid when anything from the taping is used.

Paul Grupp summarized the feelings of most people who worked at this year's Jam when he said, "It went really well this year, particularly compared to a couple of years where we had a lot of problems. We've finally gotten to the point where we've done about as much planning as we can do. We have all the right equipment now; there were no problems at all."

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jeff tamarkin

The notion of overnight success in popular music has long been exploited. It's a popular method of hype to say that so-and-so just "came out of nowhere" and "took the music world by storm." The truth, however, is that instant successes often become fly-by-nights, flashing brilliantly like a comet and then quickly burning out.

To many observers, Luther Vandross might seem like the latest in overnight sensations, because he did seemingly come from nowhere to score with a platinum debut album, record a stunning followup, and produce a smash LP for Aretha Franklin. But the truth is that Vandross, a 31-year-old New York City native, has been employing himself as a background singer and jingles singer since 1975, when David Bowie asked him to sing on one of his records. Actually, Vandross's singing experience goes back even further than that, to his high school days. His is the voice of experience, his success as singer, live performer, producer, writer and arranger is not something that comes or goes overnight.

Vandross's debut album for Epic Records, *Never Too Much* (1981), came only after years of the singer/producer turning down offers from other companies. Vandross had established himself in the world of background and jingle singing, and decided not to accept a recording contract unless he could write his own material and produce his own music. Labels, having had unfortunate incidents with other performers requesting similar clauses in their contracts, understandably balked. Epic finally gave in.

Vandross grew up musically in the 1960s; his favorite singers were women— Diana Ross, Aretha Franklin, and Dionne Warwick specifically. Perhaps that female influence is best reflected in the soft touch of Vandross's music. He eschews the tough, macho approach common to many male R&B singers and applies the dynamics and element of class that is native to Ross, Warwick and Franklin. Coupled with his sensitive songs, his studied sense of perfection in production and his ability to arrange his music, Vandross's obvious vocal talents couldn't miss once exposed to the masses. And they didn't—Vandross has been called by the *New York Times* "this decade's most promising all-around pop soul musician."

His appeal extends to the concert hall, where both men and women have been known to scream for him. Women admire his vulnerability, and many men are thrilled, even if subsconsciously, to have an artist among them who is not afraid to sing about the emotions that they too experience. His singing is relaxed, and his audience relaxes along with him. Perhaps it was a simple matter of good timing, but Luther Vandross's time has come.

Following his successful debut, Vandross realized one of his lifelong dreams ("only in my *wildest* dreams," he corrects) by producing Aretha Franklin. Her album, *Jump To It*, has been hailed by critics as the Queen of Soul's best since the '60s. The public apparently agreed, sending both album and single to the top of the charts. Vandross also produced Cherly Lynn's hit album, *Instant Love*, before recording his own second solo album, *Forever, For Always, For Love* on Epic. Following the release of that album, Vandross met in NYC to converse with *Modern Recording & Music*'s Jeff Tamarkin. What emerges is a portrait of an artist who not only knows where he came from, but also where he's going.

Luther Vandross

Modern Recording & Music: Let's start at the beginning. When and why did you start singing?

Luther Vandross: Metaphorically speaking, it was gravity—I gravitated towards it. We had a piano at my house and I started playing it at age three. I knew all the words to the Baby Washington records. It stuck with me; it was what I wanted to do.

MR&M: So you weren't one of those kids who got forced into taking piano lessons and couldn't stand it?

LV: I took lessons and did all the normal stuff and didn't like it. I taught myself the chords, although I didn't know the names of them until later. Next thing I knew I was starting to see the Shirelles on the "Clay Cole" TV show and all that, and that's how it started.

MR&M: Did you take formal singing lessons?

LV: I never really did, no.

MR&M: Did you pick up style and inflections from other singers?

LV: No. Well, of course I had influences. But really, my style comes from listening to myself and trying to do better; I'm very critical of myself.

MR&M: You've often said that your favorite singers were women—Diana Ross, Aretha Franklin, Dionne Warwick. Why were you attracted to female vocalists?

LV: It's such an intangible thing; it's like trying to figure out why your libido gets aroused by a certain kind of person. All I know is that they did it for me—the Supremes on the "Ed Sullivan Show," Dionne, Aretha on "Shindig" in the '60s, the Blossoms—they just did it to me.

MR&M: Was it because you related to what they were singing about?

LV: No, no. It was music and sound, not lyrics; it was absolute communication on a non-verbal level.

MR&M: The reason I'm asking about the lyrics is because a lot of critics have commented that your lyrics are almost the opposite of the macho sensibilities found in a lot of contemporary R&B. You seem to be more sensitive towards women.

LV: I don't know where the inspiration for my lyrics comes from, or why I might choose to write story lines that may be vulnerable or real. I think it's just an effort to be real. On my new album there's a song called "Once You Know How," and the first line is "One day I was cryin'." A guy said to me, "You're not really gonna say 'One day I was cryin',' are you?" And that got on my nerves and I said, "Well, if I wasn't gonna say it, I'm surely gonna say it now." So I did and that probably typifies how I felt all along.

MR&M: That's a brave approach for a male singer to take today. I'm sure there are a lot of other men who are going to react to that line the same way that guy did. They can't handle those emotions coming from a man.

LV: I'm not insecure about that. That's a real part of male life; you think men don't get hurt? Oh, my goodness. Not that chivalry has to be dead or anything like that, but you can't box emotions into categories that are convenient. We're all people and we're all given the same access to emotional situations. I've been hurt, just as any woman has been. Not *like* any woman, but *as well as* any woman.

MR&M: Have you ever run across a situation where a guy will come up to you after a concert and say, "Hey Luther, I can't deal with your kind of lyrics."?

LV: Never, because it's not presented in a questionable manner. To

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the contrary, I get guys who out-scream their girlfriends at my concerts, who jump up higher and acknowledge in an even more extroverted manner what's going on. I speak to people; I'm not fighting for women's rights *or* men's rights.

MR&M: Do you think that growing up in New York City contributed to your style or the way you approach music?

LV: Definitely. I think there's something to the intensity of the winters, and just the tightness because Manhattan is so big.

MR&M: Did you ever use your music as a hiding place because of the violence and intensity of the city? Do you withdraw into your music?

LV: Let me tell you, I was unaware of those things until recently. I was just oblivious to all the periphery, all the crime. I was just writing and singing and listening to Dionne Warwick. That's all I cared about. I was blind to everything else. No, I didn't do it as an escape.

MR&M: Now that you are more aware of the problems in a city like New York, would you want to use your music to speak out against it, like Grandmaster Flash has, for instance? Or would you like to stick to human situations?

LV=LV

LV: I don't want to commit myself to answering that. Who knows what I'll wake up resenting tomorrow? Isn't what you want to tell people a result of what your experience is? From day to day I like to assess what's happening with me. Who knows? I might run for president in 1990. But I seriously doubt it.

MR&M: Is there a theme that you try to convey throughout all of your songs, or is each one taken on an individual basis?

LV: It's real individual, from song to song. I wouldn't go from rape to love to social injustice in one album, but the mood of the music is first with me.

MR&M: Do your songs come more from personal experience or observation?

LV: More observation; listening to the voices inside my head. If I'd have lived through all that I write about. Lord have mercy! They'd have to write the male counterpart to *Sybil* about me. I'd have 50 personalities! If I'd lived through "A House Is Not A Home," and "You Stopped Loving Me," it'd be a little intense. It's not that I write fiction, but it's closer to that than to being autobiographical.

MR&M: When you're standing onstage and you look out into the audience, do you see men, women, both? Do you see a whole, or individuals?

LV: I love that question; I'm gonna tell you why. What I do is to do what I do, and let people decide for themselves whether or not they're being reached. You can't sell me a vacuum cleaner unless I want one, no matter what your sales tactics are. Now, when I play, the fact that they came indicates that they want to hear a certain amount of what I have. I do it with sincerity and it gets done real. Some nights I'm much better than I am on other nights, but that's the way I prefer to approach it.

MR&M: Does the audience recognize if you're in a bad mood?

LV: I'm not talking about mood. I'd never bring that onstage. I'm talking magic, chemistry.

MR&M: But if you've had a lousy day, won't it affect your performance?

LV: Yeah, it can make you fabulous. Remember, artists release our tension—that's where we get it out. I want to get butterflies when I walk onstage. I never want to get it totally right. That doesn't mean I don't want to feel it. The things that happen during the day don't necessarily show up onstage. Then again, I don't know if you (the audience) will be able to recognize when something's affecting me in a negative way. I am a pro and one of the rules of being a pro is to not go below a certain standard. Everyone has their peaks and valleys, but you can't let the valley go below a certain point or else the alarm goes off.

MR&M: Now that you're successful, do you ever look back at when things were rough and want to say to someone who may have screwed you once, "Hey, look, I made it."?

LV: No, and there *have* been circumstances in the past that have angered me. I'm glad I'm doing this at 31 instead of at 18 like I thought would happen. I thought I'd leave high school and make *Never Too Much*. I know more now and I'm glad to have the experience I have. I've seen first-hand the biggest names in the business cope with their success in a variety of manners. I've seen Bette Midler's success first-hand, standing there with her, on tour, in the studio. The same with Bowie, Roberta Flack. I've been able to add it all up and say, "Well, I wouldn't do that." I've become, in a way, astute at handling a career as it happens. I'm thankful for that.

MR&M: Did you ever get discouraged to the point of saying "I've had it; I'm going to find a regular job."?

LV: Never "I've had it," although I've gotten discouraged. But I've always been so one-track-minded about my career, even when I was in school. In retrospect, I can say that even school was just an obligatory thing to finish, just to keep my mother cool. I enjoyed it and was a good student until the end, when my grades slumped. When I discovered I could sing, that dominated my psyche. If I had to rely on my grades during my senior year to graduate, I'd still be there. I've paid dues, just like everyone has to, but I tell you, I had a ball. I did the 9 to 5 thing too, and I had little singing experiences that didn't add up to much, but then David Bowie started my career. He overheard me accidentally, and asked me to put down on tape what I was singing and I was on his *Young Americans* album. That gave me the boost to say, "OK, do it."

MR&M: You recorded two albums for Atlantic which didn't do very well.

LV: Right, but those were with a group [Luther]. It took time for me to realize I could do this by myself. That group was a crutch for me. I needed the group then, but I don't now.

MR&M: How did you find the confidence to go on your own?

LV: From working with people like Bowie, who was super encouraging. Same with Roberta Flack, and the producers when I did jingles. It started accumulating and I knew I was worth something. Bette Midler called me at four in the morning and said she wanted me to come to L.A. That showed me something.

MR&M: Singing is something that comes naturally, but where did you learn to do arranging?

LV: It comes naturally, just like singing. I always had a knack for it.

MR&M: What about producing? That's something that has to be learned.

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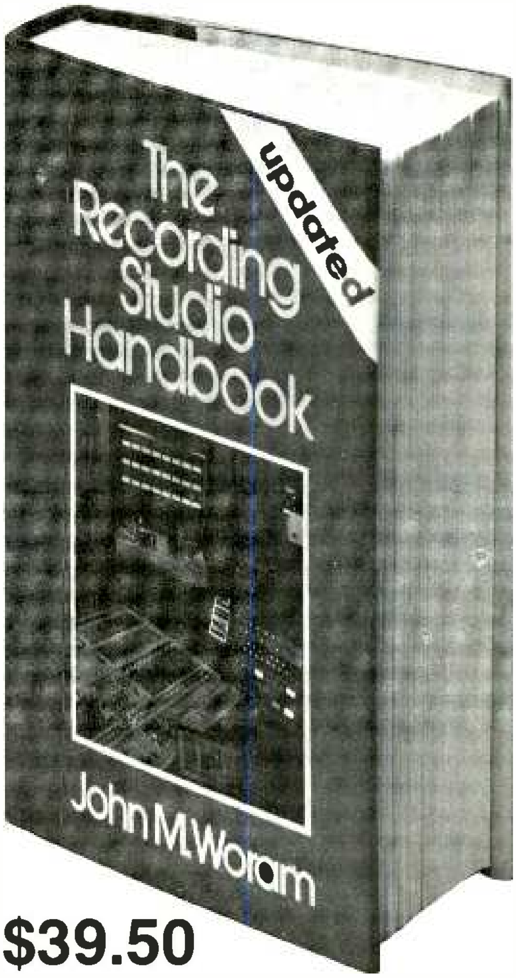

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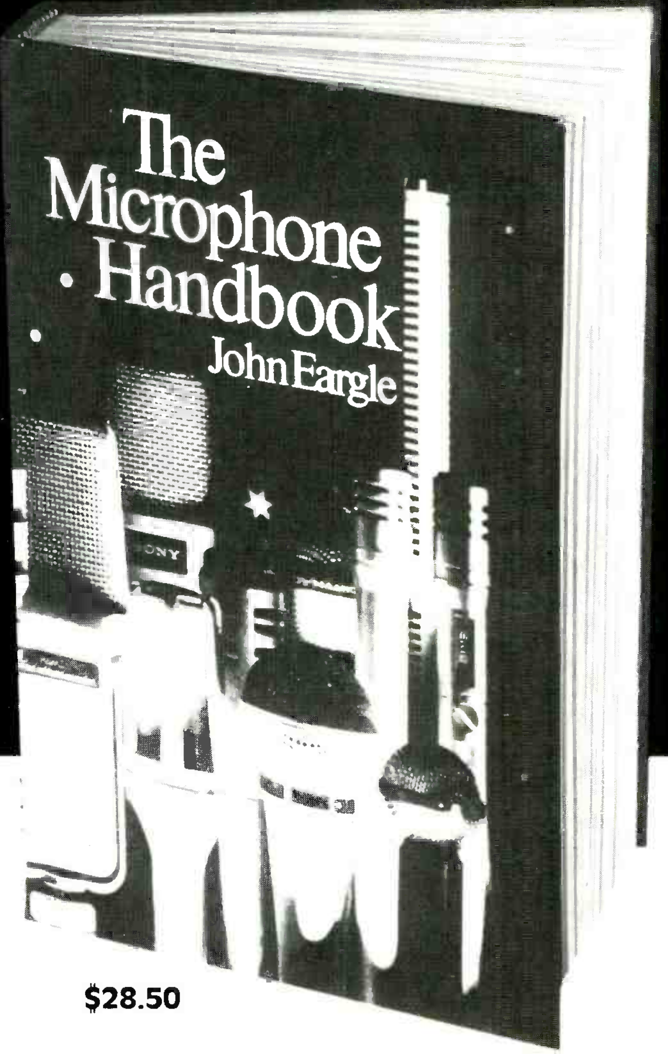

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LV: That was learned during my jingle and session career. I would do something one way, and if the producer changed the sound of it, I'd ask him about it. I'd ask why the group sounds thin, or why the soprano was dominant when she wasn't dominant when we recorded it. He'd say that maybe he changed the EQ, and then after a while I knew what to ask for. I learned which microphones fit what we were doing. I had to learn all these things, and I'm glad I did. Now I know what to ask for.

MR&M: How did you become involved with singing jingles?

LV: It was at a Quincy Jones session where I met Leon Pendarvis. They started calling me for sessions.

MR&M: Was that satisfying, to sing about products?

LV: It was real satisfying, because it was a challenge. It always interested me to maintain my own artistic standards yet give somebody exactly what they're asking for. It's a skill that you have to develop, and everybody can't do it. Every good singer is not a good background singer, either.

MR&M: What makes a good background singer?

LV: First of all, a good voice. Your instincts about music have to be good, but you have to be able to compromise your instincts so that you can sing with other people. You can be Miles Davis, but I don't know if Miles Davis can be in a marching band.

MR&M: Moving along to your current activities, you must explain how you managed to convince a major record company (Epic) to let you produce, arrange and write all the material for your own debut. That's almost unheard of today.

LV: I must admit that that's what kept me from having a deal all that time. The other labels didn't want me to produce or write for myself. Jerome Gasper and Larkin Arnold at CBS heard the tape and liked what they heard. Also, I'd done guest lead vocals on records by Change and Bionic Boogie, which brought me to the attention of a variety of people in the industry. So they were all interested in the voice, but not in the gamble of whether I could produce myself. They'd been burned several times by people who wanted to produce themselves and didn't come across with the goods. I was already making a good living as a jingle singer, so I decided I'd hold off. Finally, Jerome and Larkin said OK, let's try it. And the public was waiting. I feel that I deserve it, but I say that in a humble way—I feel it because I've worked for it.

MR&M: How do you prepare for your recordings? Do you do a lot of planning or work things out in the studio?

LV: I do a lot of homework, preproduction work. I basically know what I'm going to do before I go in.

MR&M: Is it difficult balancing your roles? For instance, when you're singing, are you subconsciously worrying about the sound that's going down on tape?

LV: Absolutely, but it's no conflict to have the different roles; it's fun. There's slight anxiety there, but that's to be expected.

MR&M: Where does the engineer fit in? Do you leave him any work?

LV: Michael Brauer engineered the second album. He gets good sounds on the instruments, so once we get the sound I like, it's just a matter of if it's played correctly, it'll sound fine.

MR&M: Do you ever require an outside opinion?

LV: Sure, I often solicit outside opinions. But once again, the bottom line is taste. So an outside opinion is fine if it's based on something other than taste.

MR&M: You've also branched out into producing other artists. Aretha Franklin's album, *Jump To It*, was her biggest record in years. How did you come to produce it? You must have been in heaven doing that.

LV: It was the most rewarding thing. She was fabulous. I'd done an interview with *Rolling Stone*, and the last question they asked was, "What do you do next?" I said I'd wrestle Bruno Samartino for a chance to produce Aretha Franklin. It was said in many a true word, but spoken in jest, if you know what I mean. But then Aretha and Clive Davis [president of her label, Arista] read that, and said, "Whoa, who is this?" (laughs) And they called me. I did three songs at a time; they kept testing the water.

MR&M: Having listened to her music since you were a kid, did you know beforehand what you wanted to do with her?

LV: Yeah. Maybe not note by note, but I did know conceptually.

MR&M: What did you try to bring out in her voice? Was there something that other producers were missing?

LV: There's nothing to bring out in Aretha Franklin. The way you produce Aretha Franklin is to make the setting right for her. To say that I had to bring anything out of Aretha's voice would be like a movie director saying that he applied the makeup to Marilyn Monroe's face. No; what he did was made sure that the palm tree cast a certain shadow and that the staircase was immaculately clean, and that the brass on the bannister shined perfectly. For Aretha, that means giving her the right material, and if you do that well, it means finishing a record in three days.

MR&M: Who else would you like to work with?

LV: Dionne Warwick—I'm getting that chance now. Diana; I've also worked with her, doing some background vocal arrangements. I'd love to work with Bowie again. I'd love to make a fabulous R&B rock record with him.

MR&M: I'm curious about the songs you covered on your two albums: "A House Is Not A Home" is a Bacharach-David tune, and "Since I Lost My Baby" is an old Temptations number. Why those two?

LV: Those songs just sit well with me musically. I don't care what a song says lyrically, but if it doesn't sit with me musically I don't want it. Those two songs just do. I'm an interpreter; some day I might do an album of only cover songs.

MR&M: How would you describe one of your recording sessions?

LV: OK, contained hysteria. We always have a good time; we've never had a session that's a drag. They're all tension-free.

MR&M: I'm afraid to ask, but when is the last time you had a vacation?

LV: I had three weeks near the end of the summer. That's it.

MR&M: What do you like to do after a day of recording or touring?

LV: I like to go home and watch "The Jeffersons," "The Honeymooners," "Alice," "One Day At A Time." I just love to lay back with a copy of my favorite magazine. I love going to the wrestling shows. I love that. I don't go to clubs or anything like that. I'm such a recluse it's amazing. I have to be coerced to go out. Sometimes I like to buy a new record just to see what's happening. I'm not married, and that's all I'm gonna say about that.

MR&M: How do you find the time to do all you do?

LV: I don't know, but it gets done.

MR&M: Ever get any sleep?

LV: Yeah, I sleep for days.

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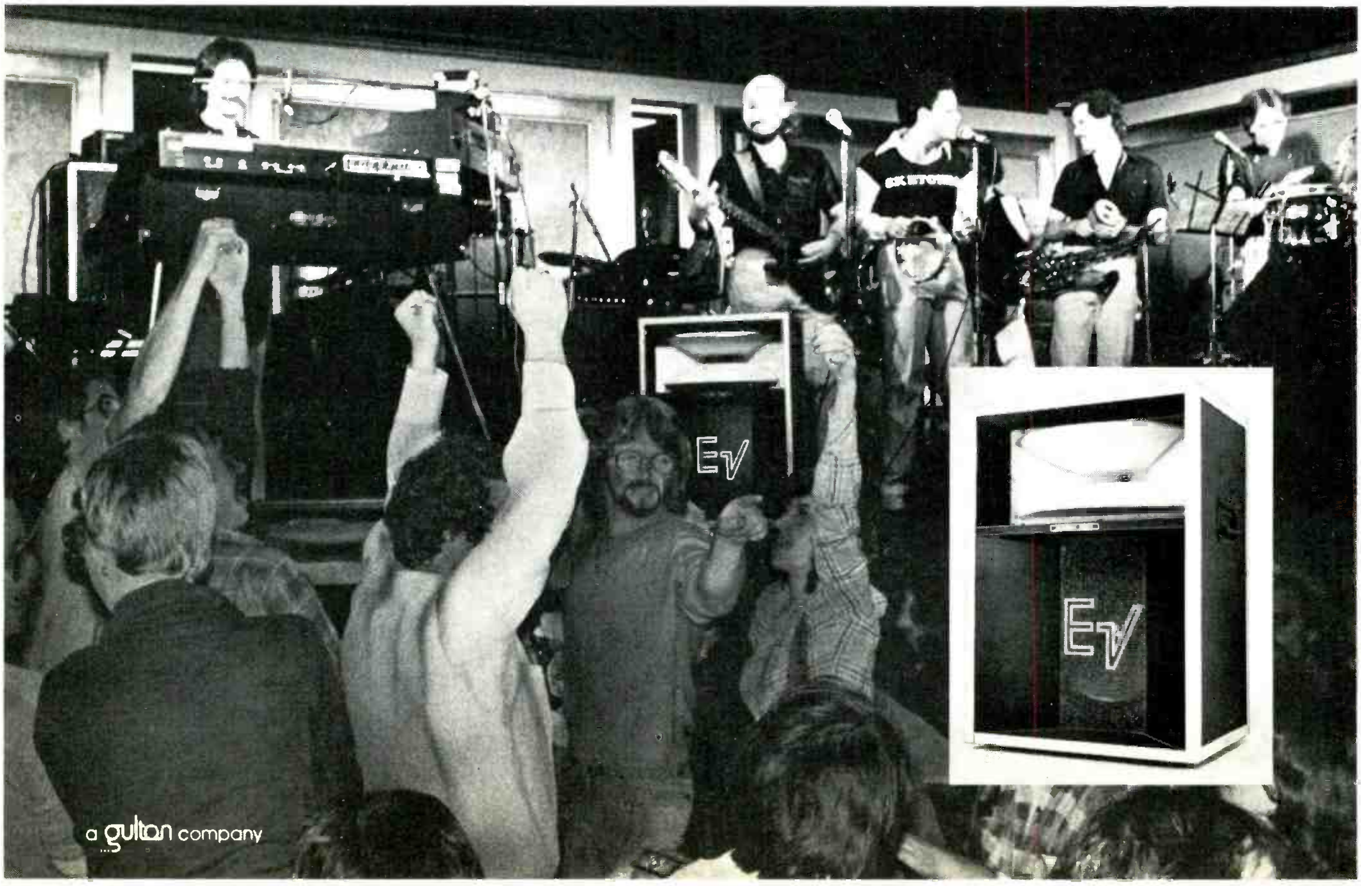

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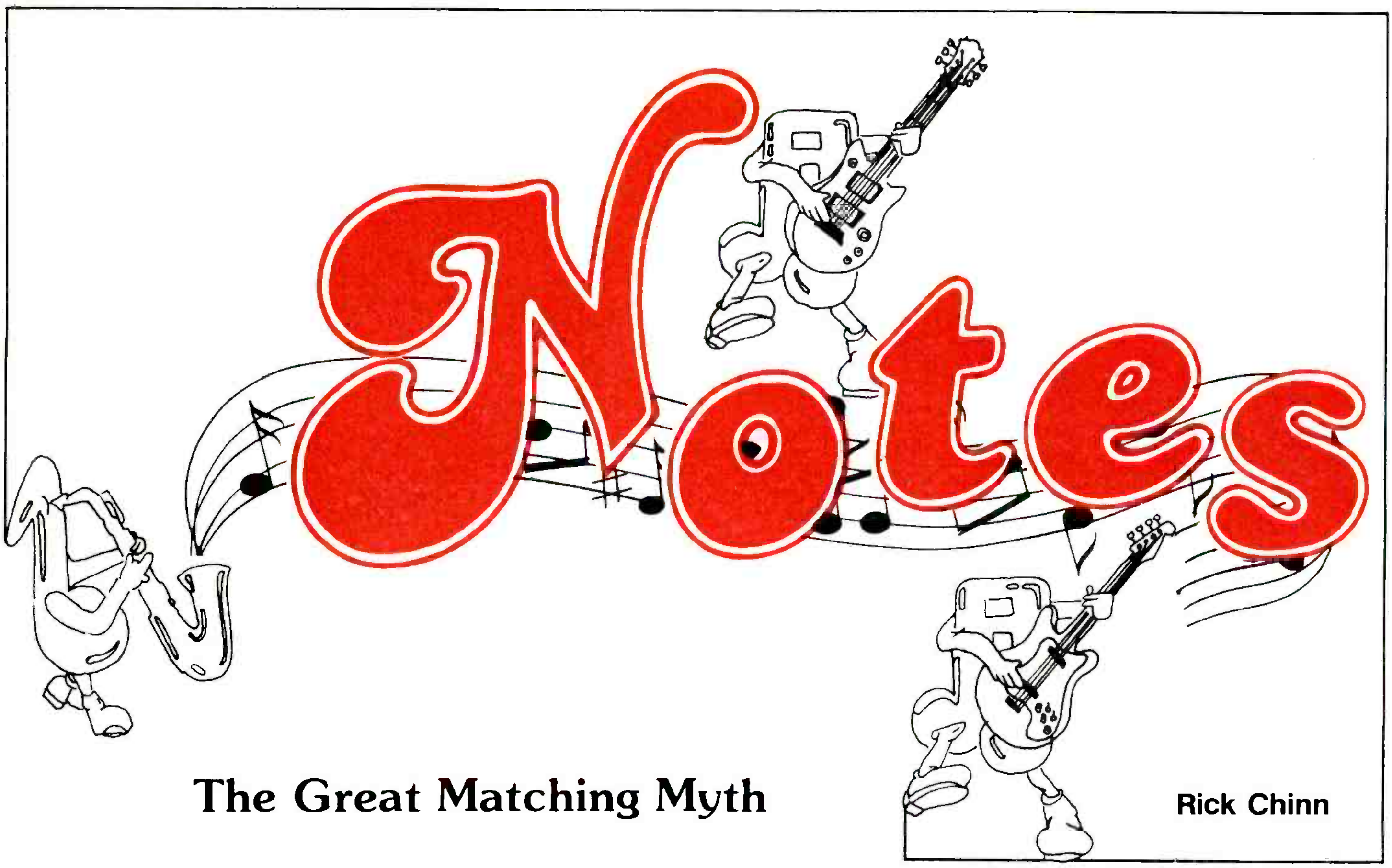


The Great Matching Myth

Rick Chinn

I think that one of the most misunderstood subjects today (among the musical audio crowd) is that of impedance matching (or mismatching). I have heard impedance matching blamed for a host of ills, yet many of the complaints were unjust. I say this as a result of answering queries, going to trade shows and just plain meeting folks over the last several years. I have sometimes wondered whether or not some of these people had impedance on the brain. Many articles and papers have been written in the last few years on this subject. In spite of this profusion (confusion?) of literary wisdom, the musical public still flounders when confronted with the subject of impedance matching (or mismatching) and dealing with a new type of matching: voltage or level matching. Fear not! As you will soon see, the real problem is *level* matching, not impedance matching.

Thus...with pen in hand (actually, typewriter on desk), I will make my stab at cutting through the fog and perhaps blowing away some of the myths that have been accumulating over the last ten years.

Warning: There is going to be some math sprinkled about here. Impedance matching or mismatching is just about impossible to explain without using math. If the numbers are over your head, try reading ahead. I'll always try to wrap it up without the numbers.

A Little History

I guess the problem started back in the old days (circa Alex Bell), before amplifiers (repeaters) were available for making up losses in long telephone circuits. The original telephone transmitters were voice-power-operated. This meant that you got the most volume when you transferred the most power to the telephone line. This usually meant *yelling* into the transmitter. The other thing the telephone folks discovered was that things got even better when the telephone line impedance equaled the transmitter impedance. Naturally, this concept of matched impedances was taken to be the most desirable condition and Presto: we have impedance matching.

As time went by, this belief in matching impedances was furthered by the development of amplifiers, filters, equalizers and other devices that had 600 ohm input and output impedances. Furthermore they didn't work right unless you matched them—*input and output!* The 600 ohm figure was the result of the broadcast industry adopting the standard of 1 milliwatt in a 600 ohm line (which equals 0 dBm) in May, 1939.

Tube amplifiers and loudspeakers didn't help matters much either. The vacuum tube, being a somewhat-high impedance device (at least compared to the loudspeaker), was basically incapable of operating a loudspeaker directly. Again, since maximum power needed to be transferred (more watts into the speaker makes it louder), some means of making the amplifier think that the loudspeaker was an OK load was necessary. Enter the output transformer. Its task was to "transform" the loudspeaker's impedance into something more palatable to the output tubes in the amplifier. Now if you didn't match (and I do mean *match*) the speaker impedance to the amplifier's output impedance, the output tubes did not "see" a "correct" load and would not operate as the design engineer intended. Since many guitar players persist in using tube-type amplifiers, we must still be cognizant of impedance matching for their sake.

As you can see, history hasn't been much help in changing our way of thinking. So far, everything we have talked about is still impedance matched.

What Is This Thing Called Impedance, Anyway?

The *CAMEO Dictionary* says: "the total opposition to the flow of alternating current (AC) in an electrical circuit. Impedance is measured in ohms."

The 1976 *Radio Amateur's Handbook* says: "when a circuit contains both resistance and reactance, the combined effect of the two is called impedance, symbolized by the letter Z."

I say: Impedance is to an AC circuit what resistance is to a DC circuit. A DC circuit results when the frequency reaches 0 Hz. Both impedance and resistance are measured in ohms.

The second definition given mentions reactance. Simply stated, reactance (also measured in ohms) is the AC resistance of an inductor (coil) or capacitor. The reactance of an inductor or capacitor varies with the applied frequency. We say that the reactance is frequency dependent. At DC, or 0 Hz, the reactance of an inductor is 0 ohms and the reactance of a capacitor is infinite. As the applied frequency increases, the reactance of the inductor increases while the capacitor's reactance decreases. This property of frequency-dependent resistance or reactance makes possible things like crossover networks, equalizers, and filters.

In an AC circuit, when you look at the impedance you are concerned with two quantities: resistive impedance and reactive impedance. The resistive portion is always there, regardless of the applied frequency. The reactive portion is the part that changes value with frequency. The total impedance of a circuit is the Pythagorean mean (remember the Rule of Pythagoras?) of their values, or:

$$Z_{total} = \sqrt{resistive^2 + reactive^2}$$

in an ideal circuit, there is no reactive component to the impedance. That is, the impedance is the same for DC and AC. In most audio circuits, at least those that don't involve loudspeakers, the reactive component of the impedance is small enough to be ignored. These are the types of circuits that we will deal with throughout the remainder of this discussion.

The Maximum Power Transfer Theorem

Earlier, I mentioned that if you want to see maximum Power in the load of a circuit, you must match the impedances. Let's see why.

In *Figure 1*, a simple series circuit, R_S represents the impedance of the load. The battery delivers 10 V, regardless of the current drain. This rather unusual battery is known as a perfect voltage source (it's perfect because the voltage does not vary with the load). This concept will be important later.

Now, fix the value of R_S at 8 ohms, and vary the value of R_L. Use Ohm's Law to compute the Power delivered to R_L. Remember that power is measured in watts and represents actual work being accomplished. In this case, the work being accomplished is that R_S and R_L are getting slightly warm.

R_S	R_L	R total	L total	P_S	E_S	P_L	E_L
8	1	9	10/9	9.88	8.89	1.23	1.11
8	2	10	10/10	8.00	8.00	2.00	2.00
8	4	12	10/12	5.56	6.67	2.78	3.33
8	8	16	10/16	3.13	5.00	3.13	5.00
8	16	24	10/24	1.39	3.33	2.78	6.67
8	32	40	10/40	0.50	2.00	2.00	8.00

As you can see, the power, P_L, developed in R_L is at its maximum (3.13 watts) when R_L equals R_S. When R_L is very low (1 ohm), the power developed in R_S is quite high, with this power dropping as R_L increases in value. Also note the voltage dropped across the source R_S and load R_L. Where the impedances match there are 5 volts across each. But, as the load impedance increases, the voltage across the load also increases.

Hmmmmmmm. If I want maximum watts, make the load equal the source impedance...but...if I want maximum volts, then the load should be as large as possible, compared to the source. Interesting. We will definititely make use of this concept later.

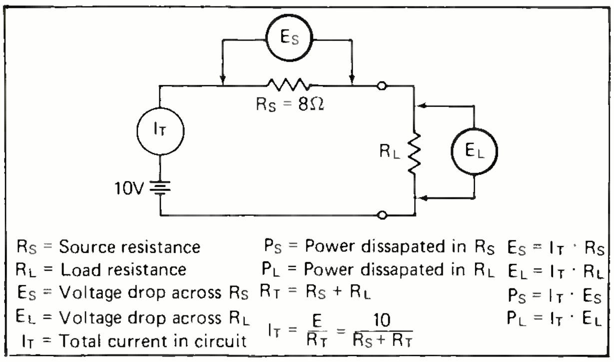


Figure 1. A simple series circuit.

Voltage or Level Matching

Simply stated, level matching is: insuring that the source has adequate level or voltage capability to drive the load with the desired amount of headroom, and that the source is being operated at a signal level that allows an acceptable signal-to-noise ratio.

A circuit's maximum output level is determined by the voltage on which the circuit is designed to operate (the internal power supply voltage), and by the minimum impedance into which the circuit is designed to work. Clipping occurs when the input signal voltage times the amplification factor is greater than the circuit's maximum output voltage. If the load demands more current than the source can deliver, the same thing happens. When either of these occur, the waveform is truncated, chopped off, or **clipped**.

When the circuit is pushed beyond its limits you hear the harsh sound of clipping distortion. (Most guitar players like this.) Inputs also have inherent signal level limitations which are also determined by the power supply voltage available to the input circuit and the amount of amplification provided by the input circuit. If the input circuit consists of a pot (i.e. volume control), then it is essentially overload proof.

A by-product of operating a circuit at room temperature (as opposed to absolute zero or -460 degrees Fahrenheit) is *Noise*. Noise in this context sounds like hiss, or white noise. It is *not* hum, buzzing or other extraneous non-musical type signals. When you examine or measure the residual noise output of an amplifier, you are measuring its noise floor. The noise floor is the value of the noise output, measured in dBv (referenced to 0.775 V).

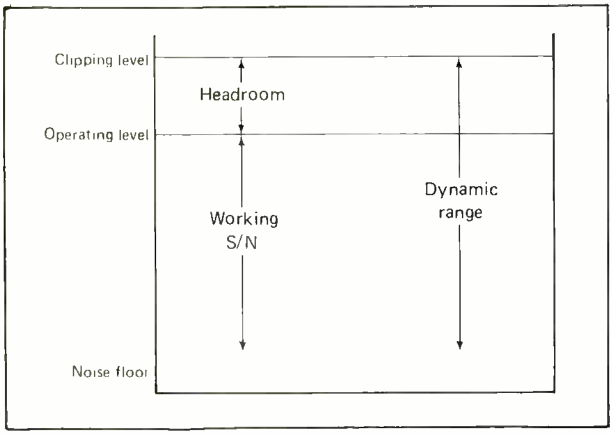


Figure 2. Signal-to-noise ratio, headroom and normal operating level.

Decibel Notation

Throughout this discussion, we've made constant reference to the decibel. Named in honor of Alexander Graham Bell, the decibel is the logarithimic ratio of two quantities (deci- or one tenth and -bel, short for Bell). This may sound like so much gibberish, but using the decibel gives us a means of comparing two quantities in a fashion similar to the way our ears perceive them.

Consider: How much more powerful is an amplifier that delivers 1000 watts from one that delivers 100 watts? First, a formula: (Use this when the two quantities you are comparing are measured in watts.)

$R = 10 \log (P1/P2)$
where: R = ratio in dB,
log = common (base 10) logarithm,
$P1$ = one quantity,
$P2$ = other quantity.

Plugging in some numbers:
$R = 10 \log (1000/100)$
note: it is customary to arrange P1 and P2 so that their ratio is greater than 1.
$R = 10 \log (10)$
note: $\log 10 = 1.0$.
$R = 10 \times 1.0$
$R = 10$ dB
Conclusion: 1000 W is 10 dB greater than 100 W.

If the quantities you are comparing are measured in *volts*, then the formula changes slightly.

$R = 20 \log (V1/V2)$
note: it is customary to arrange V1 and V2 so that their ratio is greater than 1.
where: $V1$ = one voltage,
$V2$ = other voltage.

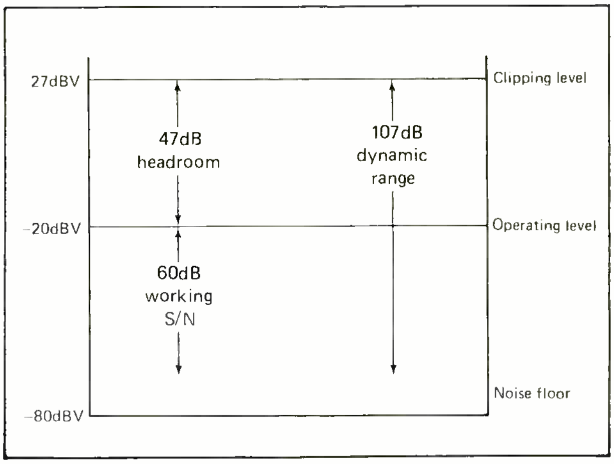


Figure 3. A practical example of Figure 2.

Consider: A mixer is able to deliver 22.4 volts. How much greater is this than 1 volt?
$R = 20 \log (V1/V2)$
$R = 20 \log 22.4/1)$
note: $\log (22.4) = 1.35$
$R = 20 (1.35)$
$R = 27$
Conclusion: 22.4 V is 27 dB higher than 1 V.

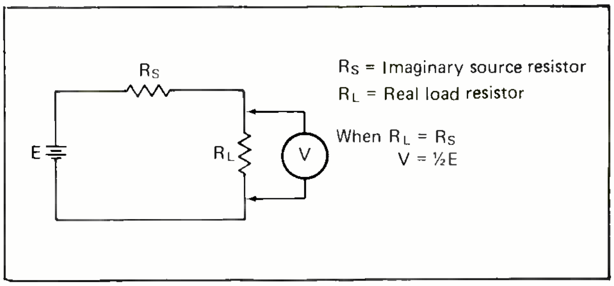


Figure 4. When r_L = r_S the impedances are matched.

When dealing with decibels, it has become customary to fix or reference one of the quantities of the ratio V1/V2. The standard most common today is 0 dBm = 0.775 V. This rather odd voltage represents one milliwatt (0.001 W) developed across a 600-ohm resistor. Anytime you see the term "dBm" it is referring to the one milliwatt (0.775 V across 600 ohms) reference.

If you see the term dBv, the reference is still 0.775 V, but the one milliwatt requirement is gone. Thus, the term dBv has no impedance reference. Rather, it is measured open circuit, with no load. Another reference used today is the dBV (note capital V). Here, the reference voltage is 1 V. Beware comparing numbers stated in dBV and dBv as they are really 2.2 dB apart.

When using decibel notation to analyze a sound system, begin with the input signal level. Add the gain of all amplifiers (in dB), subtract the loss of equalizers, faders, or other loss causing devices. The answer is the output level from that combination of components.

Example: A microphone with an output level of -40 dBv is connected to an amplifier with a gain of 50 dB. A fader is connected to the amplifier's output terminals and is set for 10 dB loss. What is the overall output level?

(-40) + 50 - 10 = 0 dBv.

Signal to Noise Ratio, Headroom and Normal Operating Level

The difference (subtract the numbers) between the noise floor and normal operating level is called the *Working Signal-to-Noise Ratio*. The difference between normal operating level and clipping level is called *Headroom*. The difference between the noise floor and clipping level is the *Dynamic Range*. Sometimes the dynamic range is incorrectly referred to as the signal-to-noise ratio (it gives a very spectacular looking specification). *(Figure 2)*

Example: An equalizer has a rated output noise floor of -80 dBv below 0 dBv with a +27 dBv maximum output level. Its nominal operating level is +4 dBm.

Under these conditions, the working signal-to-noise ratio is 84 dB and there is 23 dB of headroom. The dynamic range of the unit is 107 dB.

Question: What can you expect if you operate this unit at a minus 20 dBv signal level?

Answer: The working signal-to-noise ratio is 60 dB, with 47 dB of headroom. The dynamic range is still 107 dB. With a working s/n of 60 dB, the equalizer will be somewhat noisy, though probably acceptable *(Figure 3)*.

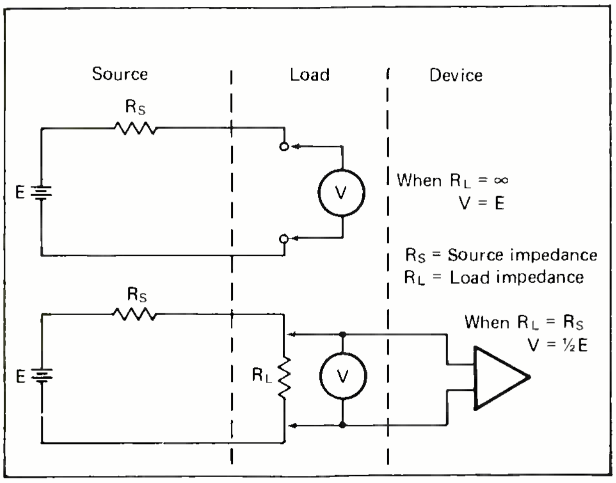


Figure 5. If $r_L = r_S$, then when r_L is connected, the meter will read 6 dB lower.

Source and Load Impedance

The source impedance of a device is the value of an imaginary resistor connected in series with the output of that device. This resistance determines the operating impedance (the impedance that the circuit operates at) of any given circuit. If you connect a real resistor across the output terminals of this device, you will cause a voltage drop. The voltage across the output terminals will drop by 50 percent or 6 dB, when the resistance of the real resistor is equal to the resistance of the imaginary resistor *(Figure 4)*. When this occurs, the impedances are said to be "matched."

The load impedance of a device is the value of an imaginary resistor connected across the device's input terminals. If you connect this device to a source whose source (or output) impedance is equal to the first device's load, or input impedance, you will have a circuit where the impedances are said to match. If you poke around with a voltmeter, you will find that the voltage developed across the input terminals of the load is one-half of the output of the source with no load (6 dB less). The diagram in *Figure 5* should make this concept clearer.

Let's put some numbers on this. The source is an equalizer with a 600-ohm output impedance. The load is an amplifier with a true 600-ohm input impedance. If the voltage measured at the equalizer output terminals is 1 V, then the voltage across the amplifier input terminals, when connected to the equalizer, will be 0.5 V, or 6 dB less *(Figure 6)*.

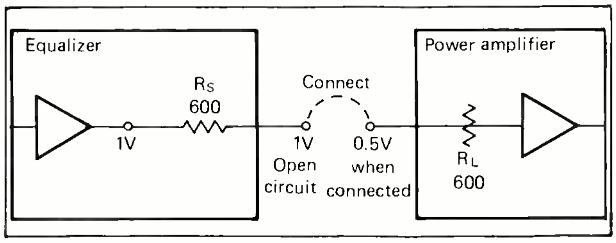


Figure 6. An example of matched impedances.

To Match—Or Not to Match

If the load impedance connected to a source is greater than the source's impedance, then proportionally more voltage will be delivered to the load. If the load is about 10 times greater than the source, the source is said to be "bridged" by the load. In this case, almost the entire available output voltage is delivered to the load.

Resorting to numbers again, the source is now an active equalizer with a source (output) impedance of 60 ohms. The load is the same amplifier as above, with an input impedance of 600 ohms. When the two are connected, we find that the voltage delivered to the amplifier's input terminals is only about 0.8 dB less than when the amplifier isn't connected. The amplifier input "bridges" the equalizer's output *(Figure 7)*. Thus,

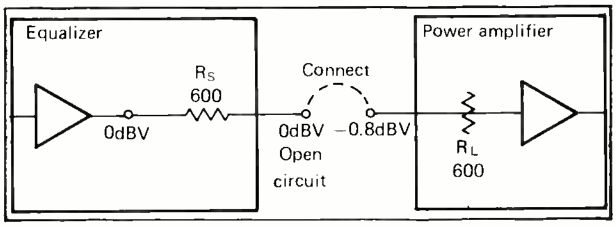


Figure 7. An example of bridged impedances.

almost the entire output voltage capability of the equalizer is available to drive the amplifier input.

Matched or Unmatched?

Looking at the specification sheet for the amplifier discussed above, we find that it requires 0 dBv (about 0.775 V AC) to drive it to full output. Thus, it should not matter whether or not it operates in a bridged or matched circuit. (In reality, there is a very small chance that it might.)

The important thing here is: When it is all connected, can the source deliver 0.775 V to the amplifier input terminals?

If our source has a maximum output capability of +18 dBv, then a maximum of +12 dBv will reach the input terminal of the amplifier when impedances are matched. If the amplifier bridges our source, then +17.2 dBm will reach the input terminals. In either event, it should work fine.

Mismatching the Wrong Way

So far, we have only talked about two conditions: 1) The source impedance matches the load impedance, 2) The source impedance is one-tenth the load impedance.

What happens if the load impedance is one-tenth the source impedance? For instance, a 6000-ohm source connected to a 600 ohm load. First of all, because of the voltage divider action (of source and load), the available output voltage of the source is greatly reduced. In this case, almost 20 dB less voltage will be developed across the load than is available from the source. Thus, with a maximum output of +18 dBv available (open circuit), -2.0 dBv will appear across the load. This is less than required for full output capability. As you can see, it will work (that is, there will be sound), but not very well. Furthermore, if the source (output) impedance of the device represents actual circuit impedances (as opposed to a resistor in series with the output terminals), this type of excessive loading may cause distortion.

With this, I give you the first rule of impedance mismatching. Don't connect sources to loads less than their source impedance. Remember too that the source impedance is not necessarily the same as the minimum load impedance that it can drive.

With most modern audio equipment, it is the *voltage* across the input terminals that determines whether or not things will work.

Minimum Load Impedance

As mentioned briefly above, the source impedance of a device is not necessarily equal to its minimum load impedance. What this says is that some sources do not want to be "matched"...ever. What happens is this: Any given source has two limits to its electrical output capabilities. These are represented by the maximum output voltages and maximum output current. If the current capability of the source is not sufficient to supply the demands of the load, the source may react by reducing its output voltage or perhaps by blowing up. Let's see.

The power company supplies 120 V AC to our homes. Notice that they don't tell us the impedance of the generators supplying the power, they just tell us that it is 120 V AC. The amount of current (amperage) that it is possible to draw is limited by the circuit breaker or fuse connected into the circuit. When you exceed the capability of the breaker or fuse, it blows and things come to a stop. If there were no fuse to stop the excess current, imagine what would happen if the wires shorted out. If the resistance of the short were say, 1 ohm, then a current of 120 amperes would flow. Since the current rating of most household wiring is 20 amps or so, this overload will not go unnoticed. Chances are there will be smoke and flames. The 120 V AC line is a close relative to a perfect voltage source.

Robert Buontempo

Making a Plate Reverb Unit

The weakest link in a small recording studio is usually the reverberation system. Yet sometimes this can be just what makes the difference between a demo- and a master-quality recording.

This article presents plans for making a plate reverb unit, which won't require any electronics other than your mixer and a headphone amp. (If you don't have these items, you're not ready for a reverb plate anyway.) The construction cost will be between $300-$500, a lot less than the $2,500-$8,500 commercially-available units. The article will describe how to find and evaluate the materials needed, construct the frame, mount and tune the plate, fit the driver and pickups, and add dampening to the plate. It concludes with some "tricks" and techniques for enhancing plate sound.

For the complete do-it-yourselfer, diagrams, photos, and a parts list are included. A kit providing a pre-constructed frame; selected, cut, reinforced, and drilled plate; mounting hardware; driver, and tuning cassette tape is also available from the author. The kit may help facilitate the project by eliminating the hardest parts: locating and evaluating materials and having the custom metal working done.

Almost everyone with a knowledge of recording is familiar with spring reverbs, or at least with their sound. Most low-end or semi-pro reverb units are based on the spring principle, as are most musical instrument amps or accessories with reverb. That "spring sound" can range from excellent to "under water," depending on the unit and the way it is used.

The reason spring units sound the way they do is because that is exactly what they are—springs. There are usually several rows of them, possibly with two or three strung in a series. Just like the springs on your screen door, they will "twang" or "boing" when plucked. However, instead of being plucked, the reverb springs are excited at one end by a driver and "mic'ed" at the other end of a pickup...and so is the twang and boing, especially on transient material.

Although some designers have used tricks to smooth out their sound with excellent results (Craig Anderton's "Hot Springs," Oct., 1980 *MR&M*), they may still have spring characteristics inherent in their sound, as well as a limited bandwidth, especially at high frequencies (8 kHz+).

Plate reverb has none of these drawbacks, although it can go from sounding like a true concert hall to an oil drum being banged with an ax in the subway, again depending on its application and who's using it.

Typically, the plate is a large (one by two meters, or 39.37 by 78.74 inches) sheet of steel suspended in a tubular steel frame. In theory, the plate simulates a large concert hall or church with a decay time (the time required for the intensity of the reverb to diminish by 60 dB) of approximately five seconds at approximately 500 Hz. A driver attached to the plate excites it, and as the sound waves travel through it, the plate flexes. The plate's motion is then picked up by one or two contact mics and added to the dry signal at the mixer. Transients do not twang or boing, but behave much as they would in a reverberant room, sounding smooth and natural. As an additional feature, incorporating a damping plate to change the decay time of the reverberated signal can be included in the design.

It was at the Broadcast Technical Institute in Nurenburg, and later at the Institute for Broadcast Engineering in Hamburg, West Germany, that the first reverberation plate using these principles was developed. EMT (in Germany) patented and made the only available units until the patents ran out a few years ago. Since then, several American and foreign companies have come out with newer units. The plans presented here are of a hybrid unit that can be optimized to the design of any of the commercial units you may favor.

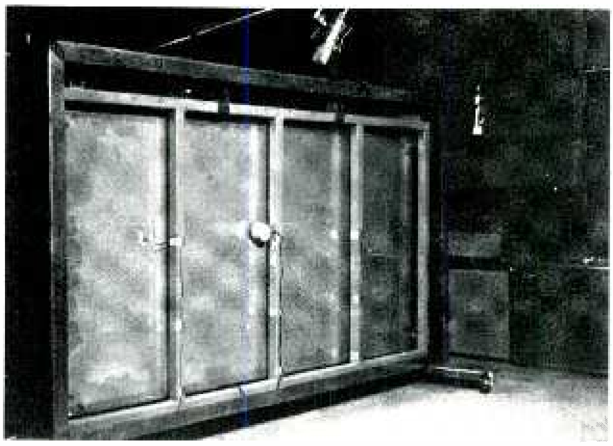


Figure 1. The finished reverberation plate. Note the oil stains; a thin coat of very light machine oil has been applied to the plate to prevent rust and corrosion. (photo by Mark Demcovitz)

Construction of the Unit

As mentioned in the introduction, the design of this unit will incorporate your mixer and cue (headphone) system as all the electronics that are required. We will mostly concentrate on the construction of the mechanical system and the transducers—the frame, plate, driver, and pickups.

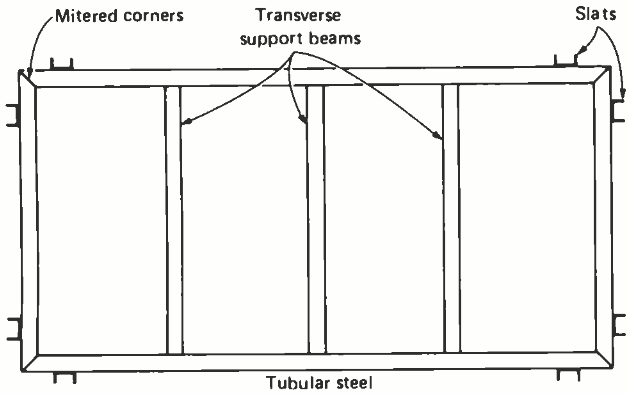


Figure 2. The tubular steel frame is reinforced with three transverse support beams.

Selection of Plate Materials

This is probably the most critical of all the steps involved in the process, so be careful. The plate is actually "the instrument" used for the reverb, so it should be chosen as if it were a fine acoustic instrument. EMT uses a one meter by two meter cold-rolled steel plate approximately 1/64-inch thick. Lawson, who manufactures "The plate" (LP1 and LP2), uses basically the same size plate, but it's a little thinner. On the other hand, some manufacturers use stainless steel. The Ecoplate by Studio Technologies uses approximately the same gauge in stainless, while DB Cassette of Sweden, who manufacture the

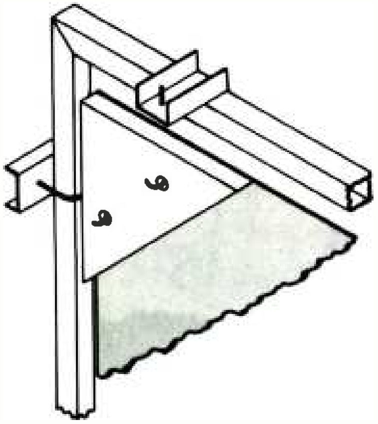


Figure 3. Corner detail, showing triangular reinforcement plate welded in place.

Stocktronics Plate, use a stretched hardened piece of cold-rolled stainless approximately 0.03 inches thick. The question of what kind of steel to use is totally subjective. Reasons claimed for using stainless steel include consistency, high density, and the fact that it's tarnish proof, while regular steel users claim smooth, more natural sounding reverb and a less "metallic" decay. Only you can decide what sound you prefer.

Befriend your local steel warehouse owner, bring two associates, and prepare to listen. Most steel sheets come in 3-foot wide sheets; this is close enough to one meter for our purposes. The length, however, is usually eight feet long and cutting charges to make it six feet might be added to the price of the steel. Some places also have minimum orders, so try to buy your plate and frame materials from the same source to save added expense.

If the owner of the shop will allow—and it's worth a healthy tip to have him help you out—have your two friends hold the sheet of steel horizontally as tight and

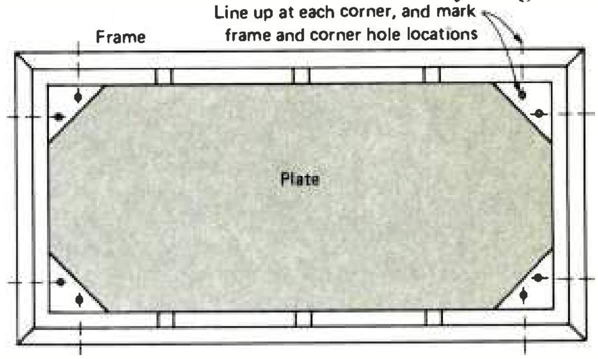


Figure 4. With the plate carefully positioned on the frame, both the frame and the plate may be marked so the holes will be properly aligned with the hooks.

still as possible, such that it doesn't "thunder." Tap it in the center with a key and listen for a "sizzle" and long decay in the high frequency, as opposed to a "clangy" sound. The delicacy and length of the high frequency decay are what you are really after, since the bottom and mids can be dealt with more successfully by tensioning. Try several pieces of different types until you find what you want. Be selective and take as much time as possible, because this is the heart of your system and you must be happy with it.

Including the cutting, the steel sheet should run between $50 and $100, depending on the type you choose. Reinforcing the corners by spot welding a triangular piece of steel on each one is the recommended procedure. For corner-cutting by the cost conscious, however, it's not totally necessary, since it could run $25 to $50 additionally. But, it really *should* be done if at all possible because the plate will be put under heavy tension and holes will be drilled in those corners later in the plate-preparation procedure. The holes should be drilled after the frame is completed so a more "custom" fit may be made.

The Frame

The frame is simply 1 to 1½ inch tubular steel—either round or rectangular—shaped in a rectangle and welded together at the (preferably mitered) corners. The frame should be reinforced by three transverse beams (*Figure 2*). Near both sides of each of the four corners (eight all together), weld flat pieces or slats of steel, which may be channeled for extra strength. These should extend 1½ to 2 inches beyond the frame, and be about 1 to 2 inches from the corners (*Figure 3*). Holes will also be drilled in the center of each of the slats. To determine the exact placement of the holes in the slats and in the plate, as well as the exact measurement of the length of the tubular steel for the frame, you must make sure the plate and the frame line up together. Make the *inside* measurement of the frame 1 to 1½ inches larger than the dimensions of the plate. Then lay the plate on top of the frame. On the plate, mark the eight spots where the holes will be drilled. Then mark the frame where the eight slats will be welded. Next, mark the slats where the holes will be drilled (*Figure 4*). When all the holes are drilled and the slats welded in place, paint the frame to stop rust and corrosion.

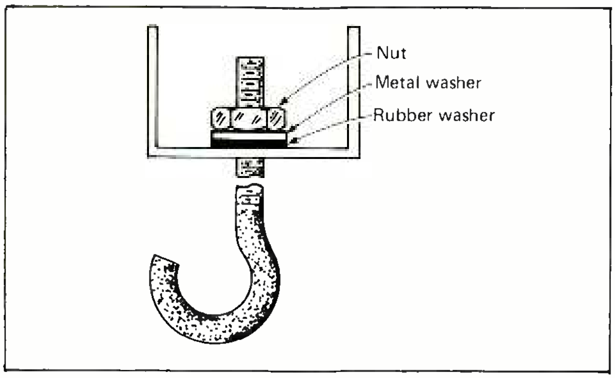


Figure 5. Detail showing correct positioning of suspension hook.

The next step is to suspend the plate in the frame. EMT uses spring clips that hold the plate in place and are also used to determine tensioning. These are weak, and often snap. One of the improvements made by most plate manufacturers is to use stronger, heavier clips or hooks. Ecoplate uses clips similar to those that secure fiber straps used on packages. We will use simple, tempered, hardened-steel hooks, threaded on their shafts. If the hook is plastic-coated, and hard-rubber and metal washers are used, the plate and the frame can be totally isolated as far as direct metal-to-metal contact goes. To suspend the plate, you will probably need help getting the hooks through the holes in the plate. Slip the shaft of each hook through the holes in the slat; thread the washers and a nut of the correct size on the hook shafts, and hand-tighten all nuts (*Figure 5*). The plate can now be suspended from the frame.

Now comes the subjective and fun part of the project—mounting the driver and pickups, and tuning the plate.

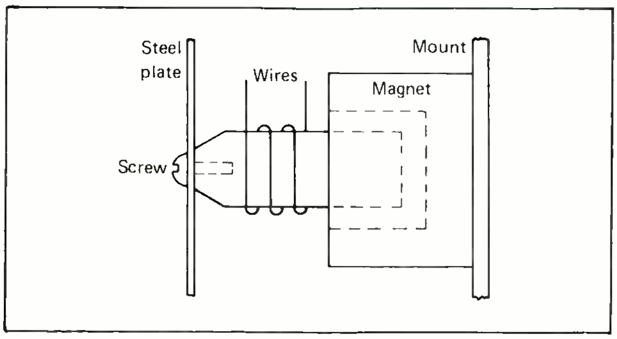


Figure 6. Driver detail, as used on several commercially-available plate systems.

The Driver

EMT, Ecoplate, and Lawson all use similar drivers. A bullet-shaped metal moving-coil "slug" is screwed into the plate. Two wires carrying the signal go to the coil and it is suspended in a large, heavy, circular magnet (*Figure 6*). It is important to be sure the moving coil assembly does not rub or touch against the sides of the magnet. The coil and magnet are aligned using a plastic alignment disk. The procedure is delicate, and transporting the unit sometimes misaligns the driver/magnet assembly.

Stocktronics uses a wire rod attached to the plate on one end and to the voice coil of a speaker on the other. It can be moved with no realignment, since there is plenty of "play" in the movement of the rod, and this is restricted to within limits by a rubber guide.

The system we will use is similar to both, but unique unto itself. It is also one of the main reasons that this plate can be built so reasonably. This design uses an Acoustic 2000 driver—similar to what used to be offered as a "coneless speaker" several years ago. Whatever the driver is attached to becomes the "sounding board" and vibrates enough to reproduce sound. Therefore, if screwed into a door, it would become a "speaker." The Acoustic 2000 (see Figure 7) is an improved version of the coneless speaker. It offers an excellent frequency response from 30-20 kHz. It can safely handle 35 watts rms and 100 watt peaks, and has a built-in crossover with a replaceable capacitor to change the crossover frequency. Best of all it is simple to use and install and is reasonably priced (list $89.95-$79.95 through kit offer).

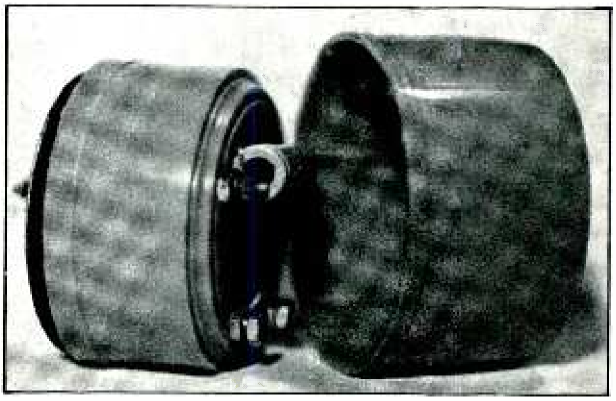


Figure 7. The Acoustic 2000 driver, showing cover (right), crossover capacitor and input terminals. (photo by Mark Demcovitz)

To install the driver, simply drill a small hole, the size for the screw on the driver, in the center of the plate 2½ inches off to one side of the center of the beam of the frame. (For optimum placement, see our review of this project in next month's *MR&M*—Ed.) Screw the driver in the side of the plate, with the frame reinforcements toward you, about half way until tight. Attach a speaker cable to the two terminals on the driver. Be neat and run the cable down the reinforcement with "ty-raps" or tape. Now the fun begins. Move the plate into your studio. Make sure the plate is standing upright. Connect the other end of the speaker cable attached to the driver to the output of your cue (headphone) amp. Put on a tape with a steady snare-drum track or a constant vocal track. Send only the selected track to your cue and, *voila*, the signal will be heard on the plate. Assuming it is the snare track, what you should hear is a thunderous snare sound similar to "Bridge Over Troubled Water" or "The Boxer" by Simon and Garfunkel (although I think they used an elevator shaft for their reverb chamber). But, anyway, congratulations! You have a working plate reverb! Now comes the real fun part—using your opinions, taste, and ego to get it to sound just the way you want. This will require your choice of pickups as well as tuning and equalization.

Pickups

All the commercial units use Piezoelectric pickups or accelerometers. These are basically contact mic'/pickups and are available from dozens of manufacturers. As a matter of fact, you probably already own one, or at least know someone who does. Some examples of available units are Barcus-Berry, "Hot Dots," and the "It" by Frap, Shadow Pickups, Countryman Associates, etc. Some pickups need no preamp and can be plugged into the echo return(s) on your. Some have their own preamps, but these tend to be rather noisy. An MXR Micro-amp or similar FET preamp would be a good substitute if needed because it uses a TLO 81 or similar low-noise, high-slew FET for excellent quality. You can also return the output of the pickups through two mic' inputs on your mixer if you have any modules free. Or, you can use a direct box or transformer to a line input. These are some of the variables that you must work out depending on the mixer you own and the pickups you use. Try as many pickups as you can borrow until you find one that you like and that easily interfaces with your mixer. For a mono reverb, place the pickup near one of the side frame reinforcements. Experiment by moving the pickup up around, and down, on both sides of the reinforcement, until you find the spot you like. Then secure the pickup by epoxy, wax, putty, or whatever the pickup manufacturer recommends. Run the pickup wire down the reinforcement, again using "ty-raps" or tape. For a stereo unit, do the same thing on the other side (*Figure 8*).

Tuning the Plate

With the pickups in place, the plate itself now comes into focus for tuning. In theory, think of the plate as similar to a drum head—the tighter it is, the higher the pitch. Also, correct tuning means all the lugs are equally tensioned. So, start by holding the hooks suspending the plate in the frame with a pair of "vice grips" or similar pliers, and tighten the nut on that hook with a ratchet wrench. Do this evenly around all eight hooks. How do you know when the plate is tuned? Good question. You don't, really, because every manufacturer used their own method for tuning. EMT ships units pre-tuned except for four nuts which are supposed to be tightened by exactly ¼ turn when installed. Most independent EMT servicemen will tell

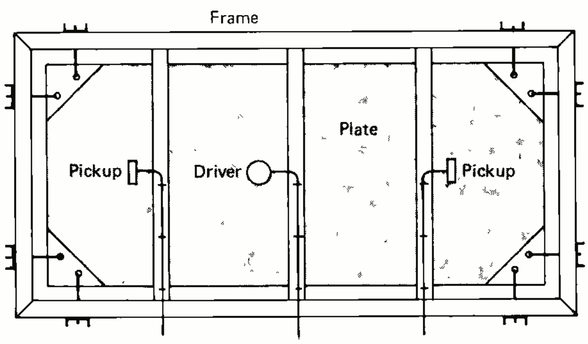


Figure 8. The finished reverberation system, with driver and pickup(s) in place.

you to tighten the plate until a spring suspension clip breaks, and then replace it and tighten until ⅛ of a turn before it breaks again! Lawson ships its units pre-tuned—no adjustment necessary. Ecoplate supplies a spring gauge and specifies pushing the gauge against the plate at all eight tensioning points until there is approximately 150 pounds of pressure at each point. As a total contrast, Stocktronics uses no tuning at all, claiming the steel is pre-stretched/tuned during manufacturing. In fact, their plate is simply suspended by six springs in a very light aluminum frame.

Which method is correct? Any/all/none, depending on your point of view. One thing is certain, though. If you like the way it sounds, it's right. So I suggest tuning by ear. Remember, the tighter the plate, the more highs and the less bottom. It is usually better to over-tension than under-tension. Also, listen for "flutters" or "beats" (like two slightly out-of-tune guitar strings) on the decay of the reverb, and even-out the tension until they disappear. EMT warns about the "oil can effect," a very metallic sound that is heard on an obviously out-of-tune plate. What I suggest is to find an existing plate you like in a studio near where you live. Rent an hour of time, and bring along a tape of various tracks—snare alone, drum kit, congas, tambourine, voices, piano, and run it through the plate. Record the reverb return on one track of a two-track or cassette, and your original dry signal on the other. Bring it back to your place and pan the dry signal to the center of your monitors, and the reverb send from their plate on the right. Send the dry signal to your plate, return it to your mixer, and pan it to the left. Now you can directly compare your plate to theirs, and tune and equalize until the sound of yours equals or betters theirs (subjectively)! (A tape like this is available in the kit offer. It is made from an EMT 140 ST tube unit—the "classic plate" sound. Notice the deep, smooth base, and crisp, sparkling highs—as well as smooth decay. Sends chills up your spine, huh?) Use your ears and you can't go wrong.

A Case for the Plate

Theoretically, you are done—but you really need a place to put your unit and something to put it in. The best place would be a separate quiet room or closet so that no outside vibrations will affect the plate. Even so, a case for the unit is suggested. The case is simply a wooden box that the frame can sit in. EMT and Ecoplate use pressboard, Lawson uses plywood, while Stocktronics has only paneling. The frame can be placed in the case on rubber feet, or better yet, suspended in the case using rubber straps with hooks, such as those found in automotive stores for holding down luggage. The straps can be wrapped around the frame and the hooks hooked to holes or eyelets in the case. This way you can literally pound the case with little vibration. Eyelets can also be put on the outside of the case on each side so that rods can be inserted for carrying. If you are only using the plate during mixdown, the studio isn't a bad place for it. It probably has the best isolation from your monitors and has easy access to your mic' inputs and headphone jacks. The case only has to be a few inches bigger than the entire unit on each side, unless you plan on using the next step—damping.

Damping

The decay time for the reverb as it now stands is approximately 5 seconds at 500 Hz. This is fine for most applications, but can easily be altered by fitting a damping plate. This can be a piece of plywood the same size as the plate and covered with an absorptive material (such as compressed fiberglass, Styrofoam, foam rubber) that can be moved closer to or farther from the plate to alter the decay time of the reverb. EMT, Lawson, and Ecoplate all move the damping plate in parallel to the steel plate, from almost touching (⅛-inch) to 6-8 inches away. This is accomplished by forming a parallelogram type set-up where two metal arms attach to the frame and to the damping plate so that when the damping plate is moved, the arms travel sideways and move it closer to the steel (*Figure 9*). Stocktronics simply hinges their Styrofoam damping plate at the bottom and then pulls

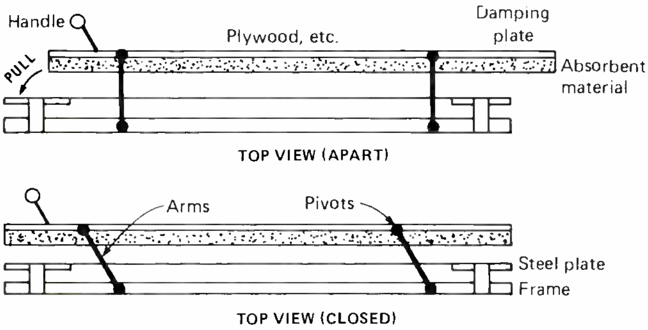


Figure 9. A parallelogram-style damping plate.

the top closer to, or farther from, the steel, claiming this gives a more uniform frequency response in the decay characteristics. A handle or lever on the damping plate facilitates moving it. It can also be remote-controlled using servo motors and cams, but this is beyond the scope of this article. The choices of materials, method, or even use of damping at all is left up to you.

Plate Tricks

Using equalization will help you get the reverb characteristics that you are after much easier than tuning alone. In fact, all the commercial units have some sort of equalization in their electronics, either a bass cut-off on the pickup amps, a high-frequency boost to the drive signal, or both. EMT cuts the bottom at 80 Hz, but many engineers use a 700 Hz high-pass filter to accentuate the top. If you have a few equalizers to spare, it would be a good idea to patch one to the send and one to each return. This will allow you to match the sound of almost any of the commercial plates or any plate sound you have heard. In fact, Studio Technologies has released a product with a parametric equalizer on the send and one on each return, as well as a delay and noise gate, all in one unit. The delay and gate help with some plate "tricks" you can use for some of the popular effects you have heard.

• DELAYED SEND—Take the send to the plate and first put it into your delay line. Use a full-bandwidth setting so that you don't lose any top end. The effect is that you're in a large hall where the first reflection isn't heard until milliseconds after the initial dry signal. The longer the delay, the bigger the hall. A good example of an extra long pre-delay is the reverb on the snare at the end of "It Keeps You Running" by the Doobie Brothers. You hear the snare hit first—and the reverb later. Sort of "*boom . . . cha*!" This will also bring out the deficiencies of a unit, and if you try it on a twangy spring, the time delay doesn't let the program mask the boing of the snare transient. But with a plate, this is no problem.

To shorten the decay without damping, a noise gate comes in handy. Placed on the return, the release time can be shortened. When the attenuation and threshold are properly set, the decay will be gradual and smooth, only shorter. If the controls are set to dramatically attenuate the decay, it can be rhythmic. For example, if hand-claps are done on the downbeat, the reverb decay can end sharply and completely on the upbeat.

You can also gate the send to the plate such that you only reverberate certain signals. For example, if you want reverb only on the snare track, and it wasn't gated when recorded, gate the snare track to the send, and you will only get the reverb on the snare beats, not on any tom toms, bass drum, or cymbals that might have leaked onto your snare mike.

Experiment and you can get any sound you've heard, and some you haven't.

"So, if I use one plate reverb with a lot of top end and a gate for the snare, and another one with a lot of bottom for 'thunder toms,' and one more with a long pre-send delay and high frequency boost for that 'sizzly vocal' sound; maybe one for the strings...with maybe a little flange on the return...and maybe one more...."

Parts List

1 Steel Sheet (of choice)
1 Tubular Steel (for frame)
1 Driver (Acoustic 2000)
1 or 2 Pickups (of choice)
8 Hardened, Tempered, Threaded Steel Hooks (Rubber coated if possible)
Each with rubber washer, steel washer, and nut.

Optional

2-4 Rubber Straps for suspending frame
Wood (of choice) for Case
1 Damping Plate and Absorbent Material (of choice) with Handle
A kit containing:

A pre-made, painted frame, selected steel sheet, all mounting hardware, an Acoustic 2000 driver, and tuning cassette tape is available for $400.00—plus shipping. Shipping is C.O.D. (please specify if you wish a quote on shipping price before kit is shipped, since shipping will vary by distance). The driver and tuning tape only are also available separately for $85.00, shipping included. NJ residents please include applicable sales tax.

SEND TO: Home Grown Studios
P.O. Box 531
Cranford, NJ 07016

Ambient Sound

By Len Feldman

Rethinking Recording Techniques In the Digital Age

I have now officially entered the digital audio era. I own a new CD player. I used to call this digital audio disc player a DAD player, which seemed to me like a catchier acronym for the laser-tracking, fast-spinning turntable/player combination that is used to play those amazing little 4¾-inch discs. Now I see that the many companies who already manufacture these players, as well as those who plan to, have switched to calling them CD. This more abbreviated acronym can stand for Compact Disc or Compact Digital; take your pick. Who am I to argue with the corporate giants of Europe and Japan? CD it is!

In addition to owning my very own CD player and about a dozen discs (more about those discs in a moment), I have also checked out a total of a half dozen PCM processors in my lab over the past year. In the course of checking out those PCM processors, I have also made a fair number of PCM digital recordings, hooking up the PCM processors to my videocassette recorder. Now that I own a CD player, and some digital audio discs, I can even transcribe those discs onto tape for a true test of a PCM processor's dynamic range and signal-to-noise capabilities. My complaint in the past had been my inability to find appropriate source material (other than a live music group) for putting these PCM processor/VCR combinations through their paces.

Sony Corporation is about to announce a new PCM processor. Similar in performance characteristics to their earlier-introduced PCM-F1, this new processor, tentatively called the PCM-701 (now *why* didn't they call it PCM-F2?) will sell at not much more than half the price of the older PCM-F1, or at around $1000.00! All that's been omitted is battery operation and the microphone preamps. Since almost every serious user of the PCM-F1 that I know used a mic mixer with it anyway, the omission of the mic inputs is no great loss. As for the portability afforded by battery operation, most users of the PCM-F1 generally powered it from an AC power supply anyway. In that sense, the PCM-701 is actually more convenient than the portable PCM-F1, in that the power supply is self-contained.

What I'm getting at is that for approximately $1500.00 ($1000.00 for the processor, around $500.00 or so for a suitable VHS or Beta VCR), you can now produce audio recordings with better specs and, in my opinion, better sound than has been possible using the most sophisticated and expensive professional analog studio equipment. That being the case, you can be sure that it won't be long before serious amateur recordists as well as professionals are going to switch to analog. Unfortunately, my own limited but intense recent experiences with digital recording and digital discs have taught me that the transition is not going to be easy. If you are a recording engineer and think that all you're going to have to do is substitute a PCM processor and a VCR for your existing half-track mastering reel-to-reel machine, think again! If your recording efforts are confined to a good analog cassette deck hooked up to a stereo system, you're in for some surprises when you substitute a PCM/VCR combination, too.

Headroom Re-defined

The first thing I learned about digital recording is that my previous experience with record level meters was of no help whatsoever. Most of us are accustomed to letting musical peaks push meter readings well above the 0 dB mark. If we're talking about professional reel-to-reel tape decks, it's not unusual to allow peaks to push those needles up to +8 dB or even higher. With home recorders (especially cassette decks), more conservatism has been the order of the day, but we still don't panic if a peak comes along and pushes those LED peak indicators up to a +3 dB or so. Today's better grades of tape don't saturate as quickly as did the tapes of yesteryear. Furthermore, because so many cassette deck makers con you into thinking you've got a lot of headroom by setting 0 dB at well below 200 nWb/meter (Dolby B calibration level), we've all learned to ignore that arbitrary 0 dB on such machines and to push levels higher for better signal-to-noise ratios.

In the case of digital recordings, all of these "rules" don't apply. The 0 dB mark on the digital processor's metering system is now *really* the absolute maximum recording level that the system will handle. Push levels further than that and you run into a real "brick wall" kind of signal clipping. The system has simply run out of bits in its 14-bit or 16-bit coding system. Simply put, in the case of a 14-bit system, when the meters read 0 dB, the recorded signal amplitude is encoded as the binary number "11111111111111." In binary encoding of this sort, you have 14 places to put either a zero or a 1. When all of the places are occupied by 1s, that's equivalent to the highest signal amplitude that the system can support. If the incoming audio signal gets any louder, the instantaneous sample will still show up as "11111111111111." Think about it a bit (sorry!) and you realize that all higher amplitudes will be recorded as "14 ones," and will get played back as a constant-amplitude, flat-topped signal totally analogous to amplifier hard clipping. The lesson to be learned here is that your *average* record level, as indicated on peak-reading meters, is going to have to be around -20 dB or even lower. That is very hard for an experienced recordist to accept. It was for me, but until I learned that there could be no exceptions, I kept running into that brick wall.

Now, assuming that you really want to take advantage of the available dynamic range of a PCM recording system and *not* ride gain or insert any electronic compression when making a live recording, you may have to back off even a bit further. If you have been using any form of professional compression in studio work as a matter of course, you're going to be surprised at the new peak-to-average level ratios you encounter during a live recording. They can easily exceed 30 dB above average levels, and being able to faithfully capture such peak excursions without running into the limits of the system is what really distinguishes digital sound from even the best analog recordings during playback.

Playback Considerations

Whether you intend to play back digital recordings over a home audio system or through a monitor amplifier/speaker system in a recording studio, you will have to rethink the reproducing chain, too. Even if your average monitoring or playback level draws only 1 watt of power per channel from your playback amplifier, a 20 dB peak on top of that means 100 watts per channel. And a 30 dB peak above 1 watt, for those who have forgotten, is—are you ready?—1000 watts per channel! The amazing thing that I found out when I started to play back digital program source material is that such wide dynamic range reproduction does not, in and of itself, sound all that loud. The peaks are often very brief and do not change the overall average listening level by very much. They simply add a naturalness of sound which can not be duplicated in any compressed, analog version of the same program material. But however similar the overall average loudness levels seem to be compared with analog source material, your amplifiers and loudspeaker will surely know the difference.

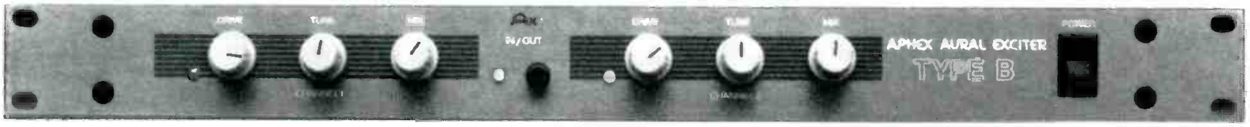
If your amps are underpowered, you'll hit severe clipping more often than before, even at what seem like moderate listening levels. If your speaker systems can't take the kind of peaks we're talking about, either because of voice-coil limitations or cone excursion limitations, you'll know that soon enough, too. For starters, I'd strongly suggest fast-blow fusing of all speaker lines once you convert to digital recording. If present instructions regarding speaker fusing suggest using a slow-blow fuse of a given amperage, I'd use a higher amp rating on the fuse, but opt for the fast-blow type. A speaker that can handle 100 watts of power for a few seconds can handle much more than that for a fraction of a second. More often than not, the kinds of high-powered peaks we're talking about in the case of digital sound are of very brief duration. As for amplifier power, the power race of a few years ago, which has since abated, may soon occur all over again—this time with good reason. Short-term clipping can be quite audible, especially to an experienced, critical listener. It would be a pity to go to the trouble of converting everything to wide-dynamic-range digital sound only to have it recompressed, in the worst possible way, by amplifiers that can't handle the new dynamics.

Broadcasters Need to Take a Hard Look At Digital Sound, Too

If you are an FM station engineer or a recording engineer working for an FM station you too are going to have to do some hard rethinking of broadcast practices once you get involved with digital program source material. At the very best, your station is able to provide 70 to 75 dB of dynamic range to close-in listeners tuned to your stereo signal. Listeners somewhat farther away don't do that well, since stereo signals don't provide nearly as good "quieting" as mono signals once you get far from the transmitter. That being the case, what are you going to do about the 90 dB or so of dynamic range that's available on the new digital source material? If you send it through a compressor, why bother with the digital discs and tapes at all? If you don't, you're going to overmodulate if you keep average modulation levels where they are now. If you try to back off on average modulation to accommodate the new, greater peaks, your average loudness will be much lower than that of your competitors, while the softest passages of program material will be buried well beneath the background FM noise heard by your audience. Right now there is no ideal solution other than to set new standards for FM broadcasting in light of the oncoming digital resolution. Are you listening, FCC? (Probably not—Ed.)

LEN FELDMAN

Aphex Type B Aural Exciter



General Description: The Aphex Aural Exciter is a signal processing device which, according to its inventor and manufacturer, is supposed to increase presence, clarity, and intelligibility, enhance stereo imaging, offer greater perceived loudness without adding extra power, and reduce listener fatigue. In addition, it requires no special decoding, since is is a single-ended device. If that sounds like the greatest thing since sliced bread, take heart and read on.

The operating principle of the Aphex Aural Exciter is really quite simple (despite the fact that its inventor claims to have worked for a couple of decades before perfecting the idea). As illustrated in the block diagram of *Figure 1*, the main signal is fed from input to output and undergoes no change whatsoever. All signal processing is done via a secondary outside loop and the resulting processed signal is added to the main signal in small, but user-variable amounts. The signal processing that occurs in this side chain consists essentially of two steps. First, the signal is passed through a tunable high-pass filter, whose cutoff frequency can be set from around 1 kHz to 5 kHz. Such a filter also introduces a phase shift which increases with decreasing frequencies, reaching its maximum shift below the filter cut-off frequency. Following this filtering, the side-chain signal is metered for correct input level to the next stage. That stage is a harmonics generator which creates low-order harmonics of the incoming signal. The resulting signal then goes to a mixing control where it is added at low level (typically 20 dB below the main signal) to the main signal. Perceptive readers will realize at once that another way of describing this signal is to say that the Aphex Aural Exciter adds harmonic distortion to predetermined sections of the musical spectrum; and that's exactly right! That having been said, please reserve judgment and don't run away from this report just yet.

Control Layout: The Type B Aural Exciter is a two-channel device, and therefore major controls are duplicated for left and right channels. These include a DRIVE control which sets input level based upon the colors indicated by a nearby multiple-color LED (green means too low a signal level, yellow is just right, with occasional red flashes on peaks acceptable as well), a TUNE control which sets the cut-off frequency of the high-pass filter stage, and a MIX control which determines how much of the processed signal gets back into the main signal path.

Controls common to both channels include a power on-off switch at the extreme right of the unit and an IN/OUT pushbutton at the center of the panel which enables you to compare reproduced sound with and without the Aphex unit in the signal path.

The rear panel of the Type B Aural Exciter is equipped with standard ¼-inch single circuit phone jacks for inputs and outputs. Input impedance is 47 kohms while output impedance for each channel is low (150 ohms). An operating level switch on the rear panel allows for a 10 dB shift in optimum input levels. Shifting this switch does not change the gain of the device, but merely optimizes the circuitry so that the aforementioned multi-colored LED will provide the correct indications when setting levels via the front panel drive control.

Test Results: There are few standard measurements that are applicable to the Aphex B Aural Exciter, hence our rather abbreviated table of VITAL STATISTICS for this product. Input-to-output frequency response of the device is essentially flat (within 1 dB) from 5 Hz to 85 kHz. If you turn down the MIX control to its minimum setting (and don't add in any harmonics of the fundamental test signal), inherent harmonic distortion of the pass-through signal path of the device is extremely low: 0.003 percent at 1 kHz, 0.0045 percent at 20 Hz and 0.0054 percent at 20 kHz. SMPTE-IM distortion measured only 0.004 percent under these test conditions. Of course, if you start mixing in the low-order harmonics

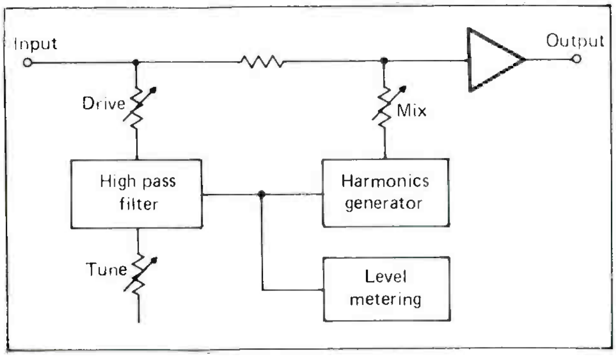


Fig. 1. Block diagram of one channel of Aphex B Aural Exciter.

generated by the processing section of the device, harmonic distortion rises rapidly, but that's what it's supposed to do if you define these additions as distortion. With the MIX control wide open (you'd probably never turn it up that high in actual use), distortion of a frequency within the pass band of the high-pass filter rose to a high of nearly 3 percent. All distortion measurements were made at an output level of +10 dBm. Signal-to-noise ratio referenced to 1 volt output was 89 dB, A-weighted. The Aural Exciter consumes about 7 watts of electrical power.

In order to convince ourselves that we understood the working principle of the Aphex B Aural Exciter, we decided to do some spectral analysis of an output signal both without and with signal processing mix. *Figure 2* is a scope photo of the spectrum analysis over the range from 20 Hz to 20,000 Hz (logarithmically swept). For this analysis the MIX control of the Aural Exciter was turned fully counterclockwise (no processed signal added to the main signal). The test frequency was around 2 kHz, and the TUNE control was set so that the high-pass filter passed all frequencies above 1 kHz. This insured that our 2 kHz test signal would be processed when we turned up the MIX control to its maximum (clockwise) setting.

As you can see from *Figure 3*, the signal certainly did get "processed." Now, in addition to the main signal spike seen at around 2 kHz on the 'scope display, we also see a rather large second-harmonic distortion component at 4 kHz, and a smaller third-harmonic distortion component at 6 kHz. Vertical sensitivity of this display is 10 dB per division, so the second-harmonic distortion component added by the processor is 30 dB lower than the fundamental tone being fed to the device; the third-order distortion is about 55 dB lower than the fundamental. These distortion components, taken together, work out to be a harmonic distortion percentage of 3.17 percent, providing good confirmation of our early maximum distortion reading observed using a distortion analyzer.

General Info: Dimensions of the Aphex Aural Exciter are 19" wide (rack mountable) by 1¾" high by 6" deep. Weight is 4½ pounds. Price: $495.00.

Comments: I have to confess that when it comes to audio reproduction I am something of a purist. Before I started to dig into the workings of the Aphex Aural Exciter I was, frankly, turned off by the notion of deliberately adding distortion to an otherwise clean signal to achieve an effect. After I thought about it a while, though, it occurred to me that if, indeed, some overtones of music and speech are lost in the multi-generation recording processes which we use today, then what would be so bad about adding back some overtones to restore some of the crispness and live feeling of the master tape or even of the live performance itself? I was well aware of the fact that we can tolerate rather large amounts of harmonic distortion during music reproduction simply because we don't always know whether the electronically-generated harmonic components are actually distortion (signals not present in the original performance) or whether they are, in fact, overtones (that word sounds so much nicer than distortion, doesn't it?) that are, or were, part of the original program material. The same, of course, is *not* true of other forms of distortion such as IM, to which we are much more sensitive simply because the IM components are *not* harmonically related, or multiples, of the fundamental tones of the program.

That being the case, I concluded that if a judicious amount of low-order harmonic distortion added to a music signal does indeed help to brighten the sound and increase intelligibility of speech, and do all those other things claimed for the Aphex system, then there's really nothing sinful about using such a device. My own listening tests confirmed that the effects described really do occur; there's no doubt about it! I found the Aphex B to work particularly well when listening to AM radio stations, which are notorious in chopping off the higher frequencies of program material. Some old cassettes and 8-track cartridges still in my possession were brightened up nicely (and, yes, made much more intelligible and crisp in the speech and vocal portions of programming) when the

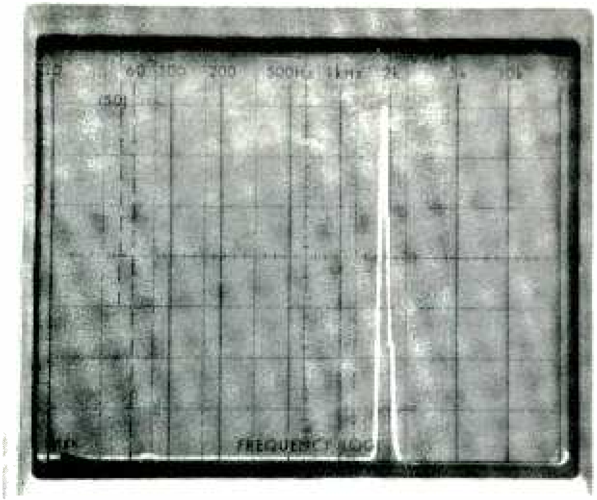


Fig. 2. With "mix" control at minimum, 2 kHz test tone fed through Aphex B Aural Exciter appears as a pure output signal on spectrum analyzer.

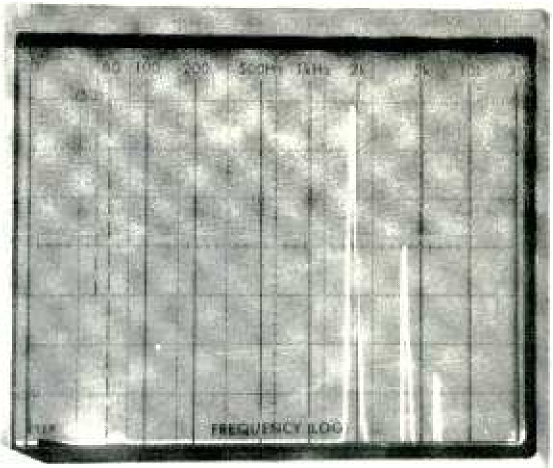


Fig. 3. With "mix" control at maximum, second and third harmonic components are added to the output signal by Aphex B Aural Exciter.

Aphex B was inserted into the signal reproducing chain. In short, the Aphex B is a clever little gadget which, if used in moderation, can improve the apparent sound quality of certain kinds of programs. I must stress the word moderation here, because if overused, it begins to sound like just what it is—a distortion adder.

Having acknowledged that this device is worthwhile in certain applications, I can't sign off without voicing one philosophical objection to the way in which the Aphex B is being promoted. In an earlier magazine article which talked about the inventor (Curt Knoppel) and the invention, the principles of the Aphex system were said to be analogous to holography in the case of light. There was talk of using "psychoacoustic principles to actually recreate these missing details, restoring the natural brightness, presence, and clarity to your sound. The sound will seem to jump out of the speakers. The stereo image spreads making listener and speaker location less critical." Even the words used to create the acronym Aphex—Aural Perception Heterodyne Exciter caused me some discomfort. I guess I just don't care for that kind of hype. Included with the material sent to me concerning the Aphex Aural Exciter was a copy of the U.S. Patent (4,150,253-April 17, 1979) which was granted to inventor Knoppel. The title of the patent, at least, tells it like it is (though I did find several errors in the diagrams accompanying and forming a part of the patent). It is called "Signal Distortion Circuit And Method Of Use." I guess the U.S. Patent Office doesn't go for any psychoacoustic mumbo jumbo—and neither do I.

APHEX TYPE B AURAL EXCITER: Vital Statistics

SPECIFICATIONS	MANUFACTURER'S SPECS	LAB MEASUREMENT
Frequency Response	+0, -½ dB, 10 Hz-50 kHz	5 Hz-85 kHz, -1 dB
THD (20 Hz to 20 kHz) (mix at minimum)	0.02%	0.003% @ 1 kHz 0.0045% @ 20 Hz 0.0054% @ 20 kHz
SMPTE IM Distortion	N/A	0.004%
S/N (Mix at minimum)	100 dBv	89 dBv
Filter Tuning Range	1 kHz to 5 kHz	Confirmed
Operating Levels	0 dBm & -10 dBm	Confirmed
Dimensions	19" w. x 1¾" h. x 6" d.	Confirmed
Weight	4.5 lbs	Confirmed
Power Required	9 watts	7 watts
Suggested List Price $495.00		

Circle 42 on Reader Service Card

Sony PCM-F1 Digital Audio Processor

General Description: The Sony PCM-F1 is a portable PCM (digital) audio processor that can operate from its own self-contained battery power or from an external (supplied) AC power adaptor. It may also be powered by an appropriate associated Beta format VCR, though in terms of digital audio recording, it will work just as well with a VHS-format VCR.

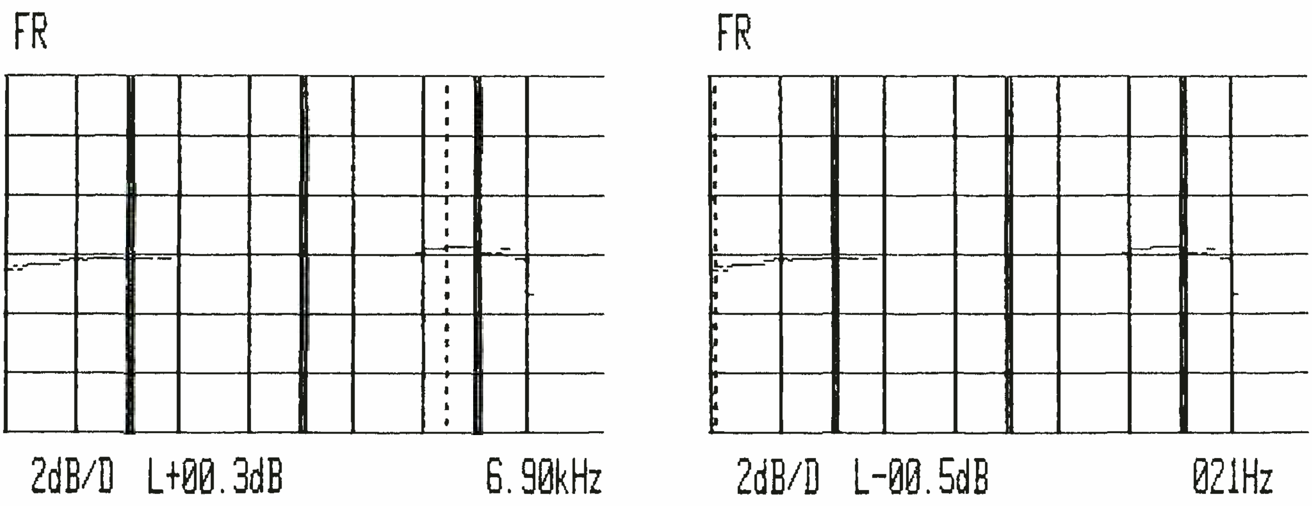
As most readers probably know, the EIAJ (Electronic Industries Association of Japan) agreed some time ago upon PCM encoding standards to be used with VCR transports. (It seems that Japanese manufacturers have a much easier time agreeing upon standards than we do over here.) The standard involves a 14-bit sampling code. On the other hand, the Sony-Philips Compact Digital Disc System involves a 16-bit code which, among other things, provides somewhat greater dynamic range and lower distortion. Since the Sony PCM-F1 uses the same D/A chip that is used in their new CD Disc player, it was a relatively easy matter for them to incorporate both 14-bit and 16-bit digital formats in this remarkable PCM processor. If you want to play a PCM tape made on some other processor, simply flip a switch to the 14-bit mode. If your recordings are to be part of your own in/out loop (recorded as well as played on the PCM-F1), use the 16-bit mode for marginally better results.

A description of the front panel controls will give you some idea of the sophistication of this PCM processor. At left is a power on/off switch. Just below are a headphone attenuator switch and a headphone jack with which you can monitor material being recorded as well as playback. Much of the left section of the panel is devoted to expanded-scale, LED record/playback level meters, calibrated from -50 dB to 0 dB, with an OVER RECORD indication beyond 0 dB. Below the left and right channel meter scales are several indicator lights. A MUTE light illuminates if the associated VCR is mistracking or if there are excessive dropouts; EMPHASIS lights up automatically for tapes made and played on the PCM-F1 (tapes made on other machines without pre-emphasis will automatically be played back correctly and the EMPHASIS light will not glow). Tapes made with a "copy inhibit" code will cause a COPY PROHIBIT light to come on on the PCM-F1. A TRACKING light comes on when tracking is adjusted on the associated VCR. The right-channel meter scale doubles as a tracking adjustment indicator when a METER pushbutton at the right of the panel is depressed to choose that function.

Controls along the right section of the front panel include left and right channel record level controls, a momentary record-mute button, the meter function button, peak hold and reset buttons for the meters, and a battery check button. Miniature toggle switches below the touch buttons choose muting on/off, copy on/off and mic/line inputs. Mic input jacks are located at the lower right corner of the unit. Line in and line out jacks are on the rear panel, along with video in and out jacks that connect to the associated VCR, and a COPY OUT jack. With the front panel copy switch activated, an error-corrected signal is available in digital form at this output for copying from one digital tape to another. The 14/16-bit selection switch is also located on the rear panel.

Whether you use a VHS or Beta type recorder with the PCM-F1, Sony suggests that you stick to the 2-hour speeds and don't use excessively long or thin tapes. Specifically, if using a Beta machine, stay with the Beta-II speed rather than the Beta-III speed. In the case of a VHS machine, it should be operated at the SP speed for best results.

Test Results: Generally speaking, our otherwise excellent Sound Technology 1500A Tape System Tester was not up to the task of measuring the performance of this magnificent recording system. Its own residual noise and distortion were simply not low enough to provide meaningful and reliable readings of the PCM-F1's performance. That was not the case for frequency response, where the accuracy of the test instrument is known to be within ±0.1 dB. *Figures*

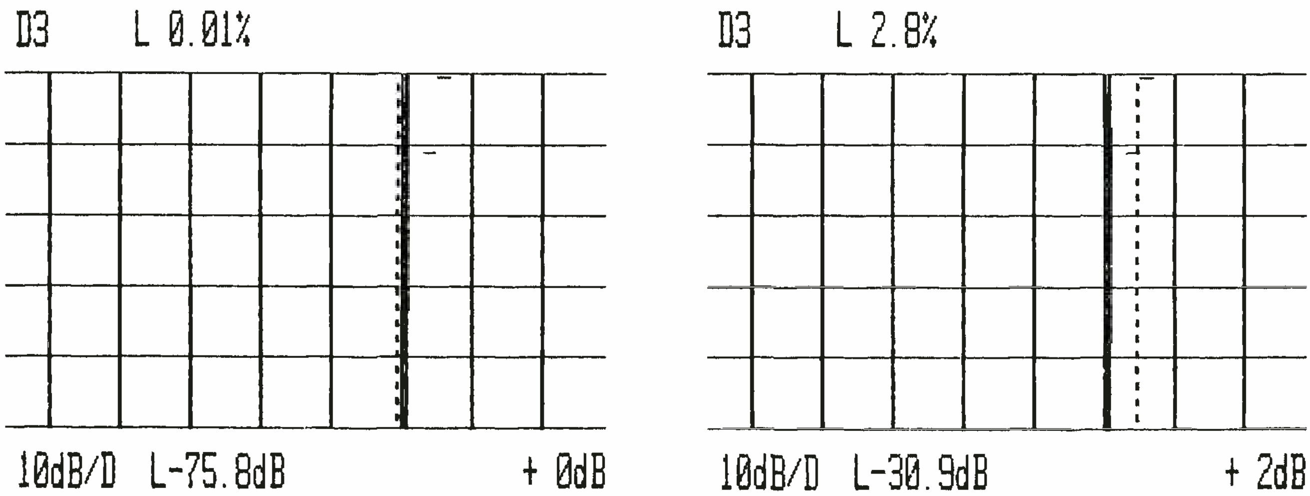


(A) (B)

Fig. 1. Frequency response at 0 dB record level, Sony PCM-F1, 14-bit mode, with cursor set to show maximum deviation from FLAT RESPONSE at treble end (A) and bass end (B) of spectrum.

1A and *1B* are identical plots of record/play response. In both cases, the vertical sensitivity scale has been expanded to 2 dB per division (as opposed to our usual 10 dB per division) to allow us to see very small deviations from perfectly flat response. The only difference between the two displays is that in *Figure 1A* the dotted-line vertical cursor has been moved to show maximum deviation from flat response in the treble region (in this case, +0.3 dB at 6.9 kHz), while in the case of *Figure 1B*, the cursor is positioned to read out maximum deviation in the bass region (−0.5 dB at 21 Hz). Bear in mind that these plots were made at a maximum record level (0 dB on the PCM-F1 meters). Try getting that kind of "straight-line" response with any cassette deck or even any open-reel deck you know, unless you are operating at 15 ips or faster.

In *Figures 2A* and *2B* there is a dramatic illustration of the so-called "brick wall" effect of digital recording. The two displays are identical, except for positioning of the dotted-line cursor. In *Figure 2A*, where the cursor is set at 0 dB record level, third-order distortion reads 0.01 percent (probably lower, actually, since 0.01 percent is the lowest reading obtainable on this test equipment). Moving up to +2 dB, distortion quickly rises to 2.8 percent, and the additional "blip" seen near the top of the diagram indicates that at +3 dB we would read closer to 10 percent distortion. The lesson to be learned is that you should *never* exceed 0 dB recording levels in a PCM (digital) audio recording system. Interestingly, if we switched to the 16-bit mode (thus far all measurements were made using the EIAJ standard 14-bit mode), distortion decreased somewhat



(A) (B)

Fig. 2. At 0 dB record level, third-order distortion is below the measuring limit of the test equipment (A). Distortion rises rapidly, however, when exceeded, reading nearly 3 percent at +2 dB record this level is level (B).

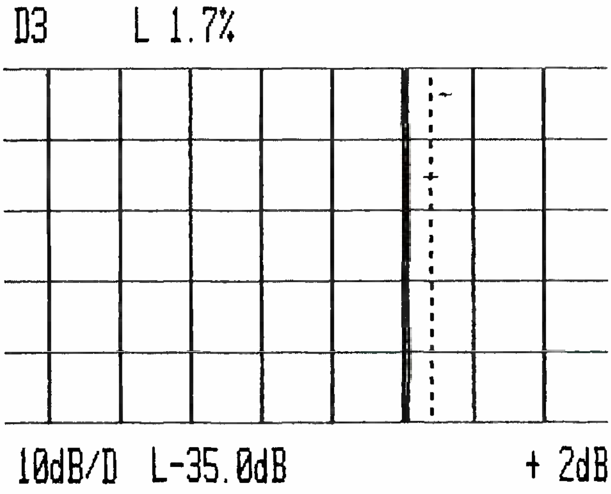


Fig. 3. Slightly lower distortion was observed at +2 dB record level when the 16-bit mode was employed.

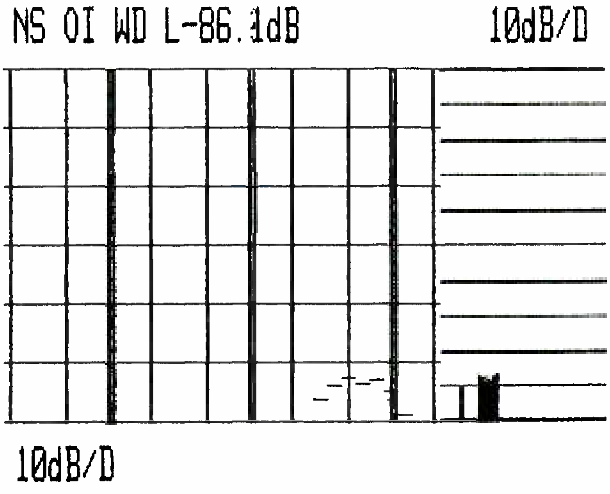


Fig. 4. S/N analysis, referenced to 0 dB record level. showed a reading of 86.1 dB, but this may be limited by residual noise of test setup.

at the +2 dB level to 1.7 percent as illustrated in *Figure 3*. At recording levels of 0 dB or lower, however, any difference in third-order distortion between 14-bit and 16-bit operation would be audibly insignificant.

CCIR/ARM weighted signal-to-noise ratio (referred to 0 dB record level) measured 86.1 dB in the 14-bit mode, as illustrated in the noise analysis diagram of *Figure 4*. We suspect that the true S/N was even a bit better, since the same reading was obtained even with the power to the PCM-F1 turned off! Actually, if you wanted to use the same sort of reference levels that are used in testing an analog recorder (i.e., the level at which 3 percent third-order distortion occurs), you could add another 2 or 3 dB to the S/N reading for a total of 88 or 89 dB!

We tried to measure wow-and-flutter but, as expected, it was too low to register on our test equipment, if it existed at all. Bear in mind that our tape tester can measure to well below 0.01 percent, and yet it did not register any reading when testing the PCM-F1.

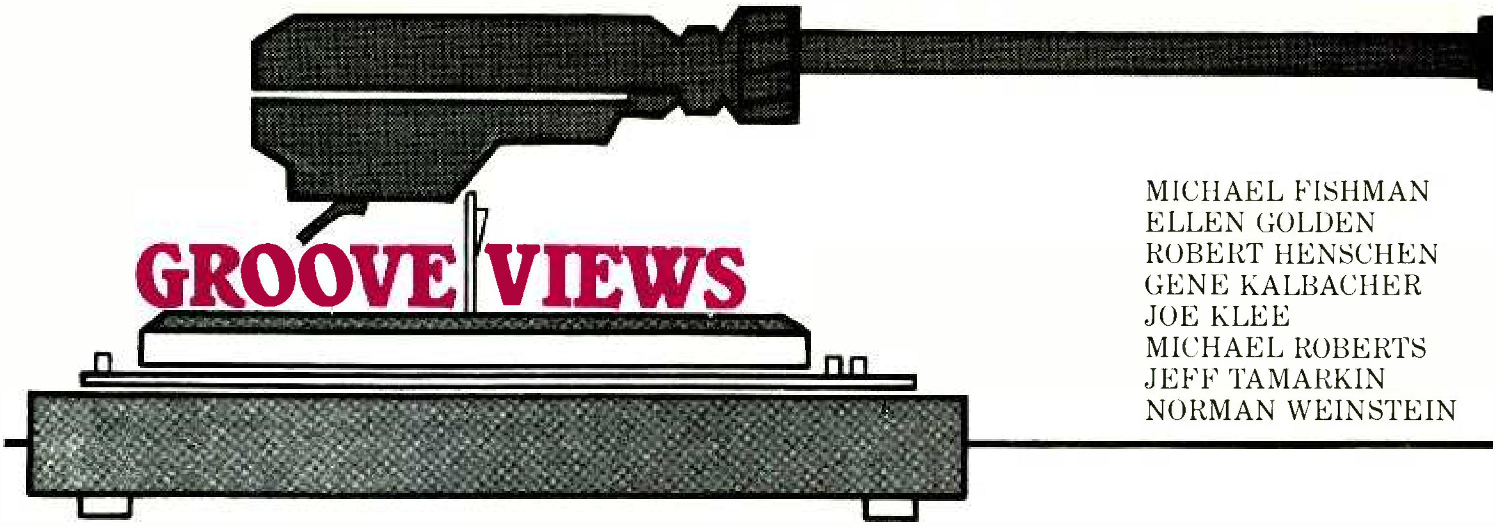
Comment: The first time I tested a PCM processor (a couple of years ago) my biggest problem in dealing with the subjective aspects of the test was to find source material good enough to record in PCM. I ended up using dbx-encoded discs, despite the fact that they do introduce minor artifacts. Now the situation has changed. CD discs are here (at least for a few of us), and I was able to transcribe a half-dozen of these discs currently in my small collection onto video tape, using the PCM-F1. When I played these tapes back I could not distinguish their sound quality from that of the CD discs from which they were made. For the serious recordist, a processor such as the PCM-F1 offers an important alternative to a two-track reel-to-reel mastering deck. For remote recording work it is far and away the best thing going today. Just imagine setting out with a small mixer, a few high-class microphones, a PCM-F1 and a portable VCR and being able to produce an on-site recording of a live performance that is completely free of wow-and-flutter, background tape hiss and, for all practical purposes, distortion-free.

Are there any disadvantages to this approach? Of course there are. Post production editing is not simple. While it's rumored that the PCM-F1 can be interfaced with some of Sony's professional digital editing equipment, from a price point of view that's a bit like the tail wagging the dog. And, as yet there are no plans for a low-cost editor to go with the PCM-F1 that we know of. Still, when you remember that it wasn't so long ago that you had to pay $5000 or more for any kind of PCM processor (not counting the required VCR), Sony's PCM-F1 can only be regarded as a true technological breakthrough.

SONY PCM-F1 DIGITAL AUDIO PROCESSOR: Vital Statistics

SPECIFICATIONS	MANUFACTURER'S SPECS	LAB MEASUREMENT
Signal Formal	NTSC TV Standard	Confirmed
Coding Format	14-bit (EIAJ) or 16-bit	Confirmed
Frequency Response	10 Hz to 20 khz, +/-0.5 db	20 Hz-20 khz, +.3/-.5 db
THD (14-bit model)	Less than 0.01%	Less than 0.01%
Dynamic Range: (14-bit mode)	More than 86 dB	More than 86.1 dB
Wow-and-Flutter	Below Measurable Limits	Confirmed
Mic Input Sensitivity	0.435 mV	0.45 mV
Line Input Sensitivity	95 mV	100 mV
Line Output Level	95 mV	100 mV
Video Output Level	1.0 V p-p	Confirmed
Headphone Out Level	50 to 775 mV	Confirmed
Power Consumption	17 watts	15 watts
Dimensions (w x h x d, in.)	8.46 x 3.15 x 12	Confirmed
Weight	8.8 lbs	Confirmed
Dimensions, Power Supply (w x h x d, in.)	4.2 x 3.15 x 12	Confirmed
Weight, Power Supply	6.6 lbs	Confirmed
Suggested Price (Including Power Supply,	Model AC-700): $1900.00.	

Circle 43 on Reader Service Card



MICHAEL FISHMAN
ELLEN GOLDEN
ROBERT HENSCHEN
GENE KALBACHER
JOE KLEE
MICHAEL ROBERTS
JEFF TAMARKIN
NORMAN WEINSTEIN

POPULAR

THE ROCHES: *Keep On Doing.* [Robert Fripp, producer; recorded at Blue Rock Studios, New York City, June 1982, by Craig Leon, assisted by Ken Tracht.] Warner Brothers 23725-1.

Performance: **Terrific, top-notch, splendid**
Recording: **Splendid, top-notch, terrific**

Lest anyone thought the Roches' first album was a fluke, there was a second album that was within a hair of being just as good. And if anyone needs any more convincing, here is a third album that combines the best elements of the first two. Alas, if only someone were listening.

Why do the critics continue to praise this trio of musically precocious women? How could you *not* like an album that opens with a pristine a cappella version of "The Hallelujah Chorus" and closes with a slightly newer song entitled "Jerks on the Loose."

Actually, this album represents a more fully realized vocal display than either of the first two releases. Fripp called the first release an example of "audio verite," because of its uncluttered liveliness; this effort maintains the same uncluttered approach, but it doesn't sound quite so rough around the edges. And those Roche voices are right up front, with very subtle guitar, synthesizer, percussion, bass and "devices" (played by Fripp) providing a soft cushion of instrumental substance.

The voices, by the way, have lost none of their impact. Maggie Roche continues to amaze with her alto (near-baritone on "Hallelujah Chorus"); Terre effortlessly reaches the upper ranges of the material; Suzzy provides the necessary mid-range vocal instrument for a substantial three-part harmony that rivals the old Phil Spector/Brill Building "wall of sound" of the late 1950s, early 1960s.

Nor have the sisters lost any of their wacky sense of humor—in addition to "Jerks on the Loose," we are treated to "The Largest Elizabeth in the World" and "Sex Is for Children." And on "Want Not Want Not," the Roches and Fripp have concocted a vocal ping-pong match, with voices bouncing back and forth between channels.

True, some of the songs sound similar to earlier Roche efforts. "Losing True" has the same feel as the earlier "Quitting Time" and "Hammond Song," for example, and the offbeat "The Largest Elizabeth" is reminiscent of "We" off the first album and "Nurds" off their second album. And following their flirtation with the traditional Irish ballad, "Factory Girl," on the second album, the Roches turn to a David Massengill song, "On the Road to Fairfax


The Roches

County," as their homage to a traditional-sounding folk ballad.

The album is filled with nice touches by both the performers and the producer. On more than one occasion the vocal line takes unexpected turns and swerves, either in the lead or the harmony, and Fripp invariably inserts just the right quirky instrumental embellishment to complement the music.

In fact, rarely does this album fall short. If only because it is so demanding of the listener, "Sex Is for Children" might be the weakest cut, because it is hard to follow the three talking lines that proceed simultaneously, and the instrumental accompaniment is fuzzy (perhaps deliberately so). In other cases, however, a near perfect blend of voice, instrument and engineering has been achieved. It's almost as if the Roches and Robert Fripp were made for each other.

Three years ago I wrote in these pages, in a review of their first album, that regardless of the accolades, sales of albums by the Roches probably wouldn't be too impressive. But now, as then, reviews such as this serve "as a message to (1) the public, to realize that some extremely fine music is still being written, and (2) the Roches, to keep up the good work." They have kept faith with their considerable talent and their singularly appealing musical direction. They merit attention and support. SR

JESSE COLIN YOUNG: *The Perfect Stranger.* [Michael James Jackson, producer; Dave Wittman, mixing engineer; additional recording by Jeff Dodge, Barry Davis, Toby Scott and Cathy Masters; basic tracks recorded at Santa Barbara Sound, Santa Barbara, Calif., and Sunset Sound & Wilder Brothers, Hollywood by Jim Nipar.] Elektra 60151-1.

Performance: **Muscular, yet smooth**
Recording: **Clean, crisp**

Jesse Colin Young has one of the great tenor voices in all of American pop music. No Neil Young whine, just a smooth, controlled voice not without its depth of feeling. And yet stardom and commercial success have never been his.

So now comes the appropriately titled *The Perfect Stranger*, as if to describe his own position in pop music. He helped define the California folk-rock sound, and has now released a quintessential California pop-rock album—yet he still may remain relatively unappreciated.

Young might be a tricky commodity to record, because of the occasional softness of his voice, but Michael James Jackson seems to have done an admirable job here. Young's voice can be heard clearly and distinctly, even when there are choruses. He isn't lost behind mushy vocal lines or blurring instrumentals; he remains in the forefront at all times.

Jesse Colin Young

Young is also surrounded by some particularly stellar accomplishments this time out. Many vocals are shared with Timothy Schmidt, and the real hit from the album, "Fight For It," has Carly Simon singing with Young. The sidemen include bassist Michael Porcaro; guitarists Robben Ford, Buzzy Feiten and Mark Goldenberg; drummer Russ Kunkel, and pianist Bill Payne. The music itself seems to have traveled some distance from Young's previous releases: Five tracks were co-written with Wendy Waldman, one was co-written with Michael McDonald, and two were co-written with Danny O'Keefe.

We hear crisp drumming, steady bass and fluid guitar lines throughout the record. "Long Nights Coming" is just one example, with Young and a chorus backed by some nice lead guitar breaks and a slide guitar by Rick Vito, as well as rhythmic keyboard work by Bill Cuomo. This could have been a Fleetwood Mac song, so well integrated are the vocal and instrumental lines, and so dense are the instrumental breaks.

There is an overall airiness to the sound here, as if the recording sessions were infused with pure oxygen; it is practically seamless, with no rough edges. But there is also a lean and muscular sound to Young, perhaps tied to the grittiness of some of the material, that is somewhat surprising. You won't find a song here such as "Light Shine" or "Song for Juli"; you will find some ballads and some medium rockers, and some humor as well; Young and Danny O'Keefe's "Night School" is a great sendup of 1950s-1960s greaser-rock.

Few, if any, technical shortcomings are evident. The balance is struck nicely between singer and arrangement, and the resulting effect is relaxed, perhaps even laid back. However, you won't find the rural-to-suburban Jesse Colin Young of years past; from the dark leather jacket of the album cover to the black sleeveless T-shirt in concert, this is an urban Jesse Colin Young. It's good to have him back, even with the changes. SR

GROVER WASHINGTON, JR.: *The Best Is Yet To Come.* [Grover Washington, Jr., executive producer; various co-producers; Pete Humphreys, Richard Alderson and Kendall Brown, engineers; recorded at Sigma Sound Studios-Starship One, Philadelphia, PA, and Rosebud Recording, NY, NY.] Elektra 60215-1.

Performance: **Pop-jazz and gentle funk**
Recording: **Clear and bright**

At this point, it is hard to tell whether multiple saxophonist Grover Washington, Jr. adapts his composing to his production scheme, or vice versa. Either way, the Philadelphian's purity of tone and natural sweetness are given a clean production sheen on this fourth LP for Elektra Records. And if the tunes aren't stirring, they aren't offensive, either.

The Best Is Yet To Come is, in every way, a collaboration among Washington and his brain trust—arranger-conductor William Eaton and percussionist Ralph MacDonald—each of whom contributes to the songwriting. What's more, this LP sports lead vocals by Patti LaBelle, Bobby McFerrin and Cedric Napoleon. The results are salutary in the first two cases because LaBelle, on the title track, and McFerrin, on "Things Are Getting Better," are allowed to provide their own backing vocals. Each singer is equal to the task. Napoleon's nondescript lead vocals, however, are rendered totally ineffectual by a chorus of background singers on "I'll Be With You."

Grover Washington, Jr.

The title track finds Washington on tenor. His playing here is robust, but hardly sumptuous in the manner of the late Ben Webster; still, his tone meshes nicely with LaBelle's wispy, unadorned singing. On the break, things heat up as LaBelle unleashes some of her gospel fervor, yelping with heartfelt emotion; Washington hangs on her every word, answering her every line with a supportive reply. The instrumental track on "Things Are Getting Better" provides a safety net for McFerrin's vocal acrobatics (check out the singer's debut for Elektra Musician), but his scat singing is effective here, bolstered by Washington's call-and-response on saxello.

Washington's versatility is evident throughout this LP as he switches easily from tenor and alto to soprano and saxello. His solos never lack for ideas, and his phrasing is pure jazz, but his accompaniment is pure pop and gentle funk. MacDonald, keyboardist Richard Tee and guitarist Eric Gale, among others, breeze through these mostly midtempo Caribbean-flavored numbers. And that's the rub: Unlike the best jazz, there is little tension-and-release; there is, in fact, no tension to be released. This music doesn't arouse, it lulls, and no soloists emerge to spur the leader's improvisatory imagination. Call-and-response is not necessarily synonymous with tension-and-release. Washington maintains an unwavering melodiousness, but he clearly needs strong soloists to prod him along. Sonny Rollins, himself stuck in a quagmire of contemporaneity, comes immediately to mind.

The production, like the leader's often limpid lyricism, retains its sound clarity even when the arrangements grow in density, as on "Mixty Motions," side two's opener. Consequently, the music may not swing, but it soothes. Washington gets over with charm. One hopes he replaces charm with daring the next time out.

GK

SCHON & HAMMER: *Here to Stay.* [Jan Hammer and Neal Schon, producers; "Self Defense" produced by Mike Stone and Kevin Elson; Hammer and Elson, engineers; recorded at Red Gate Studio, Kent, N.Y.; Bob Ludwig, mastering; art direction, design and photography, Bob Welch.] Columbia FC 38428.

Performance: **Excruciating**
Recording: **Bombastic**

Here we have arena rock in search of an arena. The music is loud and raucous, high on volume and low on musicianship. The lyrics are stupefying and the overall sound is more dulling than a 12-hour migraine. As for production "values" (if such a word can be used), this is a state-of-the-"art" album—state of the art, that is, if this were 1976. Those expecting mindless drivel won't be disappointed.

Had Jan Hammer, a founding member of the Mahavishnu Orchestra, and Neal Schon, a veteran of Santana and Journey, been weaned on Black Sabbath and Deep Purple, *Here to Stay* might be excusable. But no—Hammer and Schon grew up in the rarified environment of John Coltrane and Miles Davis and Jimi Hendrix and Sly Stone. To make matters worse—if that is possible—*Here to Stay* is the second collaboration by Hammer and Schon. They should know better.

But all of this is beside the point. According to the canons of corporate rock, musical considerations are secondary. Packaging is paramount. And, it must be said, these guys have a nifty logo, perfect for T-shirts, bumper stickers and posters.

Nonetheless, the music, such as it is, must speak for itself. From the opening power chords of Schon's guitar on "No More Lies," through insipid lyrics ("Won't you just turn around/Try to follow the sun") and shrieking guitars with the requisite reverb, delay and echo, *Here to Stay* is a plodding nightmare. What would have been innovative in 1976—the talk box "(You Think You're) So Hot" and various keyboard excrescences—is offensive in 1983. Schon does, however, contribute some guitar synthesizer. But it looks better on the credits than it sounds on the disk. The next time Schon comes east, he'd do well to drop in on Pat Metheny and pick up a few pointers on the instrument's capabilities. Schon sings all the tunes, mostly co-written with Hammer, but his voice, like his guitar playing, is virtually indistinguishable from that of his fellow executives in corporate headquarters.

Hammer, for his part, takes a back seat. Surprisingly, *Here to Stay* benefits most from the keyboardist's drumming, a workmanlike job but not without crispness and surge.

Several members of Journey make guest appearances on "Self Defense," but their contributions are wasted—assuming, of course, that they are worth anything to begin with. Side two offers a glimmer of hope, but it's too little, too late. The middle section of "Long Time" is a casual stroll down *Abbey Road*—no doubt bassist Glenn Burtnick, who played Paul in *Beatlemania*, had input here—while "Time Again" shows a trace of lyricism before veering off into Pink Floydsville. The albums's only redeeming cut, "Peace of Mind," is an instrumental, but it's woefully short by comparison with the other tunes. Schon's fluid guitar lines harmonize stunningly with Hammer's synthesized harmonica. No doubt the tune was recorded at another session.

Here to Stay is at best tolerable, at worst insufferable. If the Surgeon General's jurisdiction extended to the record industry, this record would surely carry a warning label. But, then again, there is that neat logo designed to resemble a box of baking soda.

There's no song called "Here to Stay" on the album, so the title is either a promise or a threat, depending on one's taste.

GK

Dewey Redman Quartet: *The Struggle Continues.* [Robert Hurwitz, producer; David Baker, engineer; recorded at Columbia Recording Studios, NYC, January 1982.] ECM-1-1225.

Recording: **Exemplary**
Performance: **Unobtrusively excellent**

After long, visible tenures recording and touring with Ornette Coleman and Keith Jarrett, and recently playing in the groups of Pat Metheny, Charlie Haden, and the cooperative Old and New Dreams, saxophonist Dewey Redman is poised for what will hopefully be a long and shining era of admiring recognition. In the seventeen years since the recording of his first album as a leader, Redman's studio activity has been mostly in the presence of the aforementioned leader-giants; leadership of his own domestic releases prior to this one has been woefully limited for a stylist of Redman's standing and power. Two records apiece for the Impulse and Galaxy labels and one issued on Arista, all beauts, have not prompted the well-deserved popularity befitting Redman—not even in jazz circles. But with the release of *The Struggle Continues*, the significance of which lies in Redman's having recorded for those champions of studio finesse (the folks at ECM), it can be hoped that Redman will be granted a more realistic stance among contemporary musicians. It's amazing what record dates have done for the performing livelihoods of jazz musicians, but the track record at ECM in this regard is perhaps unsurpassed. The label has launched Redman's newest group in a meticulous, reverent setting produced by U.S. branch chief Robert Hurwitz.

Yet another first here is that this is Redman's only album on which he plays the tenor saxophone exclusively. A compelling voice on both clarinet and musette, he nonetheless covers a varied enough repertoire here to keep any listener familiar with him from wishing for musings on other instruments. The tenor, after all, is Redman's principal

Conversations in Pure Jazz: Count Basie and Zoot Sims

By Nat Hentoff

In nearly empty jazz clubs—in the late afternoon or early morning, after hours—I have heard sounds so deeply satisfying that I wished with all my soul a tape recorder had been present. I mean the sounds of musical conversation. Not cutting sessions, but the warm play of imagination between musicians to whom improvising is as natural as breathing.

It is that ambience, that utterly relaxing and relaxed sharing of the pleasures of jazz, that pervades *For the Second Time* (Pablo) with Count Basie's Kansas City 3. These easeful, compelling swingers—Basie, bassist Ray Brown, drummer Louie Bellson—have worked with each other so often and are so compatible that they could be a near-miraculous dance team.

Basie, as always, is a master of space, making silence—the tacit beats between softly struck notes—an essential part of each story he tells. (Occasionally, by the way, he reminds us of his prowess as a stride pianist. Basie can cover the whole keyboard when he chooses. Louie Bellson, much underrated all these years (including by me) is a tasteful *listening* drummer, and thereby never interferes in the least with the directions of the other players. Ray Brown is so powerful a pulsing presence that he could make Lawrence Welk swing. The recorded sound is admirably clear and warm—just like the conversation.

Like the Kansas City 3, Zoot Sims is a natural swinger. And indeed, for a good many years, he was thought of more as a power hitter than as the lyrical, reflective romanticist he also is. Because, however, of a number of more intimate recordings by Zoot (notably, those with pianist Jimmy Rowles), the gentle side of Zoot is now quite well known.

Nonetheless, *Blues for 2* (Pablo) reveals new dimensions of Zoot's singing horns. It was the very bright idea of producer Norman Granz to pair Zoot with guitarist Joe Pass and no one else, because no rhythm section is necessary. With all that free space outside, and with his own internal Greenwich mean swinging time, Zoot lets his imagination unfold in marvelously unhurried, subtly colored ways.

As for Joe Pass, his occasional tendency to play too many notes—as nearly all virtuosos do—is curbed here by his role as accompanist to the one horn. In a duo setting, an accompanist simply cannot hide the horn player in swarms of notes. And so, Joe plays with an admirable economy that makes his usual good taste even more precise.

The tunes range from a blues to "Pennies from Heaven" to a lovely, venerable standard, "Poor Butterfly," that ought to be heard more often. The sound, as in most of Norman Granz's recordings, is clean, uncluttered, straightforward—like the kind of music he records. And, it is worth remembering that if it weren't for Granz, these kinds of sessions with these kinds of players probably would not be made at all these years.

COUNT BASIE/KANSAS CITY 3: *For the Second Time.* [Norman Granz, producer; Ed Green, engineer.] Pablo 2310-878.

ZOOT SIMS, JOE PASS: *Blues for 2*. [Norman Granz, producer; Mike Moran, engineer.] Pablo D2310879.

vehicle, and the corridors through which he guides it on *The Struggle Continues* are perhaps his most subtly provocative, and certainly his most telling and melodic jaunts recorded to date as a leader.

Engineering the session was David Baker, a New York veteran in great demand for his patience, skill and sensitivity with complex improvisational musics. Throughout each of the six tunes on the album, and there are different tempos and moods on virtually all of them, Baker's miking accurately conveys not only the democratic oneness of this quartet, but also the impassioned presence of each musician in it.

Taking the critique directly to a confrontation of the rhythm section, where most recording problems in these kinds of settings usually arise, the album is unassailable. Mark Helias' bass and Ed Blackwell's percussion are crisply reproduced to the extent that the movements of their bodies as they play are seemingly discernible. Just as important, the two are wonderful creators; Helias is one of the most versatile and excellent bassists in contemporary music, and Blackwell is a legendary innovator—who along with Elvin Jones must be considered one of the most influential artists ever to have held a drum stick.

Pianist Charles Eubanks proves a thoroughly tasty presence. His forward-paced solos, most notably on "Joie de Vivre," are both supportive of the music and brightly reproduced on the lp. Redman and Blackwell exhibit some wonderful trading antics on this tune, which is itself an excusably corny but ever-so-happy theme Eubanks is again luminous, and this time powerful, on "Combinations," a tune which recalls in its procession both the Jarrett and Coleman days of Dewey Redman. For Redman's part, the same track finds him wickedly articulate; his squealing, running solos are beautiful.

And speaking of beautiful, Charlie Parker's "Dewey Square" closes the album on an accessible, emotional upbeat which is purely infectious—awe-inspiring solos (Redman, Helias and especially Blackwell), trading and interplay, and material prompting a lasting impression for both the excellence of Dewey Redman's group and the continued integrity of the production mores at ECM. MF

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SONNY ROLLINS: *Reel Life.* [Sonny Rollins, producer; Richard Corsello, engineer; recorded at Fantasy Sound Studios, Berkeley, CA.] Milestone M-9108.

Performance: **Uninspired pop-jazz and funk**
Recording: **Clean and bright, with little bite**

For the final selection of *Reel Life*, "Solo Reprise (Sonny)," tenor saxophonist Sonny Rollins replays, without accompaniment, his solos from the preceding six tunes. It is uncertain whether Rollins intended this two-minute track as 11th-hour filler or as a second thought about his choice of instrumentation for the rest of the LP, but the fact remains: These are the most palatable two minutes of the 35-minute package.

Rollins—backed by Bob Cranshaw on electric bass, Jack DeJohnette on drums, and Bobby Broom and Yoshiaki Masuo on electric guitars—seems content to sound current rather than convincing. The saxophone colossus of the '60s, the tenorman second only to John Coltrane in renown, is now earning high marks as the saxophone sophist of the '80s. More's the pity. Sonny Rollins is well on his way to becoming the Chuck Mangione of the tenor.

The technical quality of *Reel Life*, recorded over five days last August, is top-notch, but the emotional spectrum of the music is narrow. The function of this music (absentminded diversion) seems to dictate the form (light pop-jazz, with more fluff than funk from electronic instruments). The instruments sound added-on, rather than in unison, and consequently, passion gives way to platitude in Rollins' playing.

He sounds fluid one moment, pinched the next; at one point he approximates the sound of a whistling teapot. Nifty. His sweet-and-sour, sometimes vinegary intonation has lost none of its resonance, nor are his solos devoid of new ideas; the problem is, those ideas aren't acted upon by his sidemen. Rollins reacts to *them*. The tenorist is given plenty of space and time (the album's worst two tracks, "Reel Life" and "Sonny Side Up," clock in at more than six minutes apiece), and while his solos occupy the foreground, he allows himself to be chauffeured when he should be steering.

Reel Life employs instrumentation similar to that of its predecessor, *No Problem*. Broom and Cranshaw return, DeJohnette replaces Tony Williams on the drums and, in effect, guitarist Masuo subs for vibraphonist Bobby Hutcherson. The latter change is more telling, for no soloist on *Reel Life* matches Hutcherson's caliber. As a result, the backing is tepid and insufficient to spur the tenorist's improvisatory potency.

"Reel Life," side one's opener and one of five Rollins originals, starts with the saxophone stating a simple thematic figure before DeJohnette's drums enter, followed by the guitars and bass. DeJohnette's drumming is smart and crisp, though far less intricate than it is with his own group, Directions. Moreover, his reliance on the bass drum undermines his stickwork. On this medium-tempo number, tension is created but never satisfactorily released, a problem that pervades much of the record.

Sonny Rollins

As we've come to expect on recent Rollins outings, there's a Latin groove ("Rosita's Best Friend") and a bluesy ballad (Billy Strayhorn's "My Little Brown Book"). If "Rosita" is too long at 6:19, then "Book" is far too short at 3:52. The latter is one of the highlights of *Reel Life* but, once again, the instrumentation fails to convey the proper mood. Another string instrument—the piano, as played perhaps by DeJohnette—would give the tune a sense of proportion that the guitar does not.

One is loath to deny that Sonny Rollins is in good spirits on this album. However, the proceedings come off as a multi-tracked Hallmark greeting card. There's no denying that Rollins has paid his dues, and certainly he deserves to reap the rewards after having enriched the jazz canon, but in this case we're witnessing a curious reversal: The listener must suffer for the musician's art. G K

HERBIE HANCOCK: ***Quartet.*** [Produced by David Rubinson & Friends, Inc. and Herbie Hancock; engineered by Tomoo Suzuki using the Sony PCM 1610 Digital Recording Process; recorded at CBS Sony Studios, Tokyo, Japan.] Columbia C2 38275.

Recording: **Lets the music breathe**
Performance: **A must-hear, memorable session**

Every once in awhile a record comes along that not only fosters critical ideas about musical and technical wherewithal, but also spurs thinking on a far broader plane—as in how a perfect digital recording of a bone-bare quartet can present more sheer musicality than any of our so-called production wizardry incorporating scores of electronic instruments.

Along same lines, consider this: pianist Herbie Hancock's new double set with Ron Carter on bass, Tony Williams on drums and Wynton Marsalis on trumpet is an edition which transcends classification as just a great album; it is a ready, demonstrative indication that the great artistic challenge of today's recording scene lies in capturing the unique styles of virtuoso instrumentalists through their interactions with others in small groups and their own formidable soloing powers.

Recorded in Japan, production priorities here are fitting in their representation of this group as *truly* a quartet, with bass and drums as prominently mixed as the piano and trumpet. Although, coincidentally, the pace-setting talents of Carter and Williams warrant this anyway, it is a practice which should be universally applied to rhythm players who are as much participants as accompanists. Certainly, some of the more traditional material here is covered in a trumpet plus rhythm section feel, but still, no one player is ever obscured in the shadow of another who is soloing or stating a theme.

Musically, this album is important

for two reasons. It marks Hancock's return, on vinyl in any case, to an acoustic format, and a challenging one at that. There has been a lot of talk lately about Hancock's supposed faltering pianistics, but all of that should be arrested with the circulation of this record. With patented comping and wonderful solos, Hancock indicates that he is still vibrant in the acoustic situation.

Then there is Wynton Marsalis, who has fairly well taken the musical community by storm. He is simply a *happening* figure, period, and any recording with his contributions is worth checking out. His talent, style and European classical background have contributed greatly to his rapid ascent, and his work with Blakey's Jazz Messengers, his 1982 debut lp *Wynton Marsalis*, and his role in this session mark him as a red-hot player and a future influence along the lines of a Freddie Hubbard or a Lee Morgan.

Herbie Hancock

There are standards here, including Thelonious Monk's "Well You Needn't" and "'Round Midnight," as well as Hancock, Carter and Williams originals which are all perfect vehicles for their long-tested triangular interactions. Mostly, there is a lot of fine music, recorded so that those who really wish to listen need not be burdened by extra-musical concerns.

MF

HENRY THREADGILL SEXTET: ***When Was That?*** [Ed Fishman, Alan Ringel, and Larry Shengold, producers; David Baker, engineer; recorded at Sound Ideas, New York, N.Y.] About Time AT 1004. Distributed by New Music Distribution Service.

Performance: **Impassioned improvisations by class brass**
Recording: **Who could ask for anything more?**

Henry Threadgill's name has attracted attention during the last decade among jazz critics and audiences as a result of his activities in an avant-garde trio, Air. Air combines Threadgill's adventurous sax playing with Fred Hopkin's singing bass and Steve McCall's light and subtle touches on trap drums. Their newest release on Antilles Records, *Eighty Degrees Below Eight Two*, is an ideal introduction to the richness of collective musical talent that *Air* embodies. And it showcases the composing as well as playing strengths of Threadgill. Three of the four long compositions comprising *Eighty Degrees Below Eighty Two* are by Threadgill; one is by Jelly Roll Morton. This program indicates an essential truth about Threadgill and friends: as uninhibited as they are in their experimentation, they are equally uninhibited in returning to the historic roots of jazz. Not that Threadgill's homage to Jelly Roll Morton's music sounds like a Smithsonian archival reissue. Quite the contrary. Threadgill's stylisms on sax fit squarely in the tradition of post modernist playing. There're tonal shiftings and angular improvisatory movements galore. But there's funk and punch and raunch—the electric touches that Jelly Roll Morton exemplified.

The Henry Threadgill Sextet exists contemporaneously with Air. The Sextet provides Threadgill with a broader palette of musical colors with which to weave his neo-traditional jazz tapestries. The Sextet is basically a brass oriented band with a ruthlessly tight rhythm section. Threadgill joins with Olu Dara, one of the hottest cornet players of the last decade, adds a powerhouse of a trombone player, Craig Harris, maintains Air's versatile bassist and has Brian Smith on piccolo bass. Two drummers, Pheeroan Aklaff and John Betsch, maintain the titanically churning rhythmic bottom.

This format challenges Threadgill. Provokes him into doing some of the most daring improvisatory playing on sax, flute, and clarinet I've ever heard from him. There's a rare quality of musical comaraderie going on with Threadgill, Harris, and Dara that's hard to describe. They compliment, foil, toy with one another's ideas. The ten minute title cut showcases their intricate and impassioned interplay. "When Was That?" is a romp in quick march time. New Orleans at the turn of the century meets Soho loft music of the Eighties. Dara's bursts of speed on cornet are ecstatic. Threadgill and Harris, miracle of miracles, actually keep pace with Dara, maintaining a rich stream of musical ideas. The two drummers and bassists propel the composition forward with slightly less energy than Dam #1 at the TVA. A sense of unstoppable and driving swing also informs "10 to 1," a piece which showcases Threadgill's Dolphy-like flute artistry. The remainder of the album features three dirges. The dirge might not seem the most dynamic type of composition. But Threadgill's dirges are bursting at the seams with sinewy horn riffs and sophisticated drum rolls. His slow numbers move my imagination toward conversations with the ghosts of voodoo queens in old New Orleans funeral plots.

This is highly complex music to capture in the studio. The dual drummers and bassists demand a precise engineer behind the control panel. The three man horn section presents its own set of sonic challenges for producer and engineer. These challenges were well met. The sound is lovingly recorded. The instruments are well separated, sensibly mixed. The sound rivals that of any number of digital discs I've received recently.

I look forward with considerable excitement for future releases from the Threadgill Sextet. And I trust that many of you will look for *When Was That?* at your nearest record store. If you do, don't be surprised to find it missing from the shelves. About Time Record is one of the thousand record companies committed to experimental music. These independently owned and operated record companies lack the distribution systems that the multicorporations like CBS and Warner Brothers possess. Which probably accounts for why these one thousand labels accounted for two per cent of all record sales last year. I certainly can't attribute the low sales figure to lack of quality. I will gladly match this release against anything the multicorporations have released this year. Anyway, you can get this record through the mail by writing New Music Distribution Service at 500 Broadway, New York, N.Y. 10012. Write for their free catalog listing hundreds of new jazz and experimental rock releases on obscure and hard to find labels.

And get ready for an outing with some class brass. N W

ROBBIE BASHO: *Art of the Acoustic Steel String Guitar 6 & 12.* [William Ackerman, producer; Denis Reed, engineer; recorded at The Annex, Menlo Park, Ca.] Windham Hill WHS C-1010.

ROBBIE BASHO: *Visions of the Country.* [William Ackerman, producer; Dennis Reed, engineer; recorded at Recording Etc. Productions, Berkeley, Ca.] Windham Hill WHS C-1005.

Performance: **Symphonic satur nalias for steel strings**
Recording: **Fine sense of presence and exquisite attention to detail**

Imagine "a 12th century French cathedral in the province with gardens, flowers, and buds. Now comes a rainstorm, and hear the organ sounding within the church. After the rainstorm, doves flying from the cathedral windows, and the immortal chimes." Does this description fit a Debussy tone poem? A canvas painted by an obscure French Impressionist? Strange to tell, you've just read the liner notes from Robbie Basho's *Art of the Acoustic Steel String Guitar 6 & 12.* And if you believe that such a lushly romantic description is a pretentious tag for a solo guitar album, I strongly suggest that you give these two records a serious listen.

Robbie Basho is a phenomenal master of the steel string guitar. He brings to his instrument the most eccentric and idiosyncratic blending of musical styles possible. Strains of flamenco guitar are synthesized with Indian sitar techniques. American folk guitar fingerpicking styles are flawlessly executed in unusual open tunings. Cross a Leo Kottke with a Ravi Shankar. Add a dash of Segovia. Then you can begin to appreciate Basho's unique musical signature.

But Basho is far more than an autochthonous and graceful guitar technician. He is an enthralling composer. *The Art of the Steel String Guitar 6 & 12* presents his "classical" side. There are variations on a piece by Grieg and upon "Clair de Lune." There are tributes to Japanese koto music and an Islamic hymn to the Goddess of the Rose. That hymn features one of Basho's forceful vocals. His full-bodied tenor is consistently inspiring.

Visions of the Country represents the "folk" side of Basho's playing. Basho's vocals appear throughout. His poetic lyrics about the spirit of the Western countryside make John Denver sound like a suburban simp. On "Rocky Mountain Raga" violinist Antoinette Marcus handsomely complements Basho's guitar and voice. It's worth noting that all of the other cuts on both albums feature *only* Basho on guitar or piano. No fancy overdubs, no additional studio musicians are utilized. The symphonic complexity he achieves with a single guitar is mindboggling.

So are you ready to add dozens of Robbie Basho recordings to your collection? Sorry. These two albums are IT. Over a dozen of his records are out of print. Windham Hill Records deserves our thanks for retaining Basho's music in the marketplace. It's a tiny company founded by guitarist William Ackerman. A firm known for quality pressings, fine studio work, and a challenging roster of young acoustic guitar and piano artists. The recording quality on these Basho releases is typical of Windham Hill: the sound is clean, dryly precise, finely detailed. Perhaps this excellent sound has something to do with the fact that this is a record company owned and operated by real live professional musicians. These records sound like the result of one masterful guitarist producing and recording another. Hope Windham Hill sets a standard that the rest of the record business heeds.

Suffice it to say that I think Basho is one of a handful of great American guitarists who has emerged in the last decade. Name another guitarist who can make the chimes of a medieval cathedral ring in your mind with a clarity matched only by being there? N W

ROBIN ARCHER: *Robin Archer Sings Brecht.* [John Mordler, producer; John Kurlander, engineer; recorded in 1981 at Abbey Road studios, London.] Angel S 37909.

Performance: **Archer misses the mark with anglicized sugar-coated Brecht**
Recording: **The Abbey Road Sound, which hasn't changed much since the Beatles**

It looks like we're in for a deluge of albums by female singers doing Weil (with or without Brecht) and Brecht (with or without Weil). First Nonesuch gave us the magnificent album of Teresa Stratas' definitions of Weil's material and now E. M. I. brings us Robyn Archer singing Brecht. Can Judy Collins & Martha Schlamme be far behind...maybe even Cleo Laine?

Comparing the two items at hand—Stratas/Weil and Archer/Brecht—is fairly unproductive. Teresa Stratas is an operatic soprano. She sings at the Metropolitan and the Paris Opera. She is a seasoned professional. Robyn Archer is a theatre singer little-known outside of her native Australia. Her genre has more in common with Edith Piaf and Lotte Lenya than it does with grand opera.

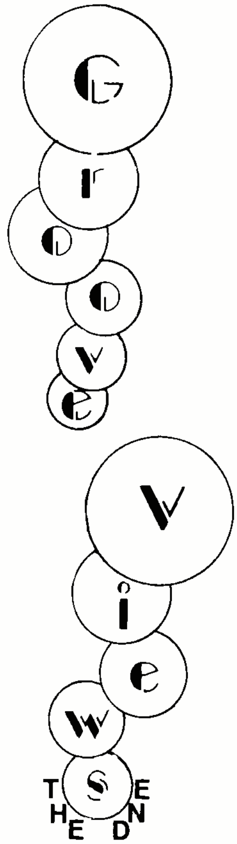
As a composer, Weil was enough of a genius to be able to overcome a poor lyric with a brilliant melody. He also showed remarkable taste in choosing lyricists after his association with Brecht. In the Stratas/Weil album we find poems by Jean Cocteau and lyrics by Howard Dietz in addition to the Weil/Brecht collaborations. In the Archer/Brecht album, we find Bertoltd Brecht in collaboration with—besides Weil—Paul Dessau and Hans Eissler (neither of whom quite made music magic in the way Weil did). It may be a comment on the wealth of great poets versus the dearth of great composers, but Weil seemed to have an easier time finding collaborators worthy of him than did Brecht.

These differences are merely surface. *Anything* of Weil or Brecht or Weil and Brecht together is worthwhile regardless of the worth of any other collaborators. Also, the individual comparison between Stratas and Archer can be minimized as Brecht's concepts had more to do with theatre than they did with the opera house. There is, however, one fundamental difference that works against Ms. Archer. With the exception of "Alabama Song," "Benares Song" and "The Swamp," which were written to English texts, the songs on *Archer Sings Brecht* were originally written with German lyrics. Yet all the songs are sung in English. Those which were not written in English have been translated by John Willett. Willet justifies—to his own way of thinking—this translation in the liner notes as follows:

> "These songs are distinguished and held together by the fact that they are concerned with meaning. And this is why Brecht, great German poet that he was, has to be performed in a language that the hearer will understand and by a singer who speaks direct to every member of her audience."

I find this both false reasoning and an insult to my intelligence. There are sharp corners and edges to the German language which no English translation will *ever* capture, and this goes for Willett's Brecht as much as it goes for Andrew Porter's recent English travesty of Wagner's Ring. There is much in the Germanic harshness and bitterness that complements Weil's (and Eisler's and Dessau's) music. For Willett to presume that I am incapable of following a libretto and translation is an insult to my intelligence. For Willett to annex his own poetic license to Brecht's genius is an assault on my sensibilities. Archer's diction is not good enough to be immediately understood by 'every member of her audience' anyway so a good libretto would still be called for. (But that would add to the printing cost of the album proper and we can't let that happen.)

What has resulted is the softening of the bittersweet quality that occurred when Lotte Lenya sang this material (in German or in English) into sort of a sugarcoating. I can't blame Willett for all of this. There is a certain prettiness to Robyn Archer's voice that takes her out of the Lenya/Piaff category. She is a good theatre singer and it would seem that she has more than average intelligence. Had she done the album in the original language it might well have been an altogether pleasant contrast to the Stratas album. As angelicized sugarcoated pap it is worth only what anglicized sugarcoated pap is worth. And while jelly beans may be doing well on the market these days, they will never take the place of the real steak and potatoes. J K



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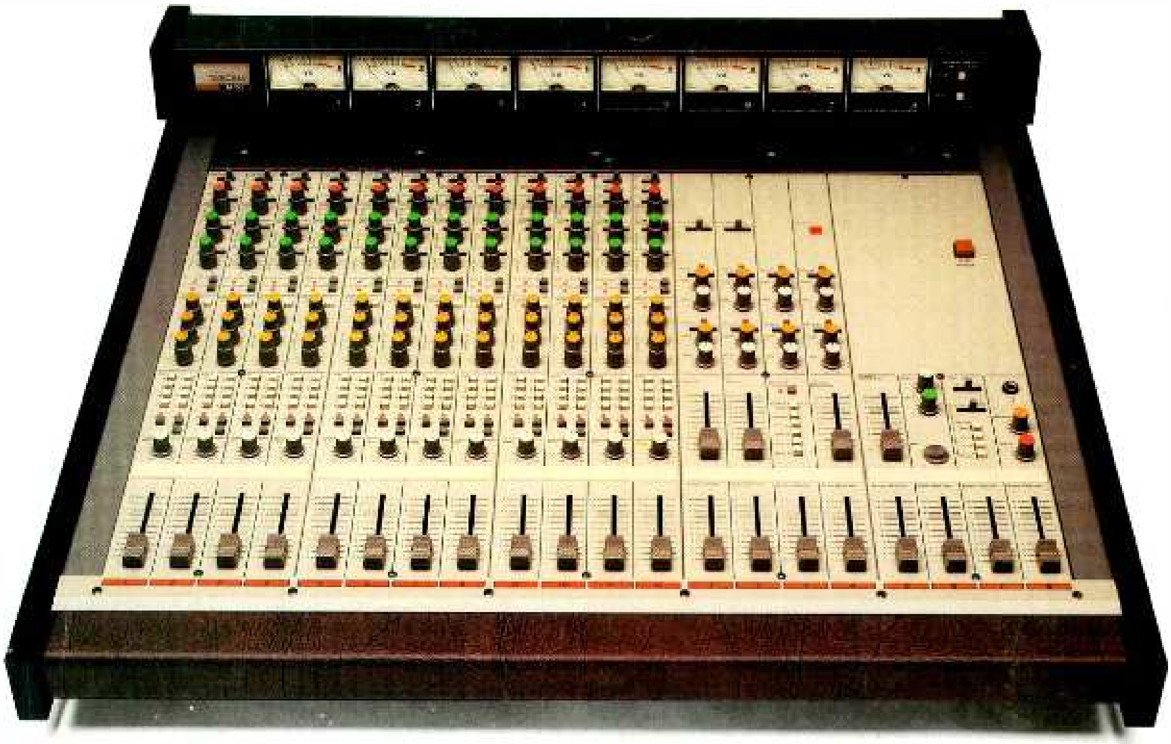

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