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MODERN RECORDING & MUSIC

NOVEMBER 1983 VOL. 9 NO. 10 $1.95

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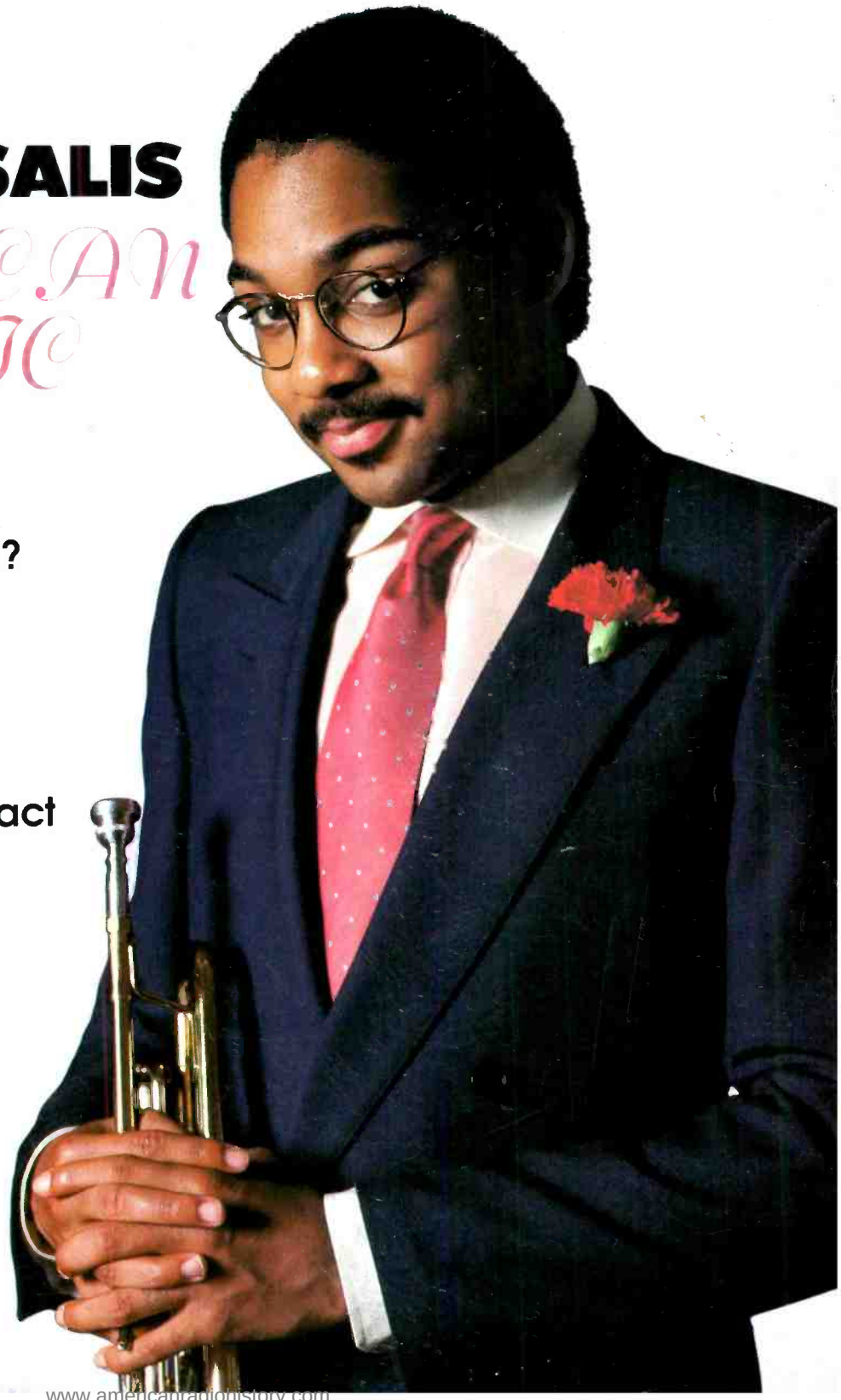

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NOVEMBER 1983
VOL. 9 NO. 10

MODERN RECORDING & MUSIC

FEATURES

SOUND IDEAS

DEPARTMENTS

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Letters

We Come Through

A long time ago I wrote requesting assistance with the confusion surrounding microphone padding. After awhile, I began wondering if the Pony Express had gotten lost along the way, and was thinking about writing a follow-up letter. My patience was more than rewarded when I saw Rick Chinn's extensive in-depth study in the April Talkback, and again in the Talkback discussion of humidity problems in the May issue.

I was overwhelmed by the thorough research and coverage of the problems involved, rather than offering just a simple explanation of the numbers involved. Judging by the number of letters appearing from those who do not have consoles, your report will surely benefit a great number of other readers.

Thanks for taking the time to present in detail the complexities behind what I thought was just a simple connection. You mentioned a "Tranz-Amp LZ" made by the Valley People of Nashville. Can you supply their address, which I have not seen in any of the magazines?

—Bernard E. Kirkwood
Reynoldsville, PA 15851

And thank you for the kind words about Rick's research efforts. And to answer your latest query, The Valley People are hiding out at 2820 Erica Place, Nashville, Tennessee 37204. Their phone number is (615) 383-4737.

And now, a word from The Department of Letter Answering. "Where do I find it?" inquiries are fairly easy to answer fast. Not always, though (see next letter). It's also not too difficult to help out with questions such as, "What happens if...", or, "Can I plug a whatchamacallit into a whosits?" In most cases, we can find the appropriate expert to deliver the answer you need.

Inquiries such as yours on microphone padding are always welcome, but they do take a bit more time to answer properly. By the way, we're always looking for new questions to (try to) answer. For one thing, it's a great way for us to find out what's on our readers' minds. However, you can help make those answers easier for us to find by being as specific as possible. Given the volume of letters we receive, and our limited people-power, we can't do much about the more subjective inquiries, such as, "What's the best used tape recorder?", or, "Will the Brand X microphone work out well in my system?"

It's not that these questions can't be answered. It's just that it takes someone a little closer to your own situation than we are. Going for the big cop-out, there is no official best tape recorder, or whatever. If there was, the others would have disappeared long ago. And then there wouldn't be any questions to answer.

Who Needs Marlin Perkins?

In your July/August Talkback, Mike Taylor wanted to know where to find Canary Mixing Desks, Ltd. The company was recently acquired by Keith Hand Music Supplies, 219 Walmerly Road, Bury, Manchester, England. In the USA, Canary exhibited their products at several NAMM shows, though not this year.

—Ursula Eastman, vice president
E and E Instruments
Laguna Hills, California

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Thanks to Ms. Eastman and several other bird-spotters for tracking down this canary. Several readers sent us the names of local representatives, but we haven't been able to track any of these down. Perhaps new reps will be appointed by the new owners. If we hear anything further, we'll pass it on.

Opinions and Suggestions

I just thought I'd write to convey my opinions about the change (or changes) you are about to or have already made. I have just received the September issue in the mail. Upon turning to the "Lab Report" pages, I began to feel dizzy. In school (Rock Valley College—Division of Technology) we have to turn in all of our lab work on quadrille grid. Personally, I find those particular kinds of pages extremely difficult to read. In addition, the grid on which you printed your reports ("Hands On" included) is much too dark for the type! A suggestion is to retain the idea of a grid, but find a different color.

As for the content of the magazine, I personally would like to see some more fusion folk in your "Profile" or "Session With" pages.

Sorry for all the trouble we put you through. We were all walking around the office a little cross-eyed ourselves. Since it was the first time we used this format, it was difficult to judge how dark to print the grid. As you can see, we've corrected the problem.

As for your other suggestion, we'll be happy to try and profile more fusion artists. We would, however, like to point out that past issues have featured such fusion luminaries as Herbie Hancock, Al DiMeola, Jean-Luc Ponty, and, most recently, Spyro Gyra and Oliver Lake.

Getting Your Hands On

Re: July/August 1983's TALKBACK. I'm not surprised at the number of people wanting to work in recording studios for "hands-on-training," and I have a suggestion for those people based on my past experience.

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Functions like stereo tape monitoring are switchable to become stereo effects sends for PA. And PA monitor sends can be switched to effects sends for recording or mixdown.

And whichever mode you're working in, you'll have a truly remarkable console at your finger tips. There's switchable high and low EQ with continuously variable midrange. Twelve-point bar graph meters let you monitor every signal that passes through the board. And the solo function lets you isolate and monitor any of the 18 input or four group signals.

But the only way to really appreciate what has gone into the Ramsa WR-8118, or its smaller 12 channel version, the WR-8112, is to see your nearest Ramsa dealer for a hands-on demonstration.

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I studied audio recording at Berklee College of Music in Boston. My major was Music Composition, because at the time I attended they didn't have an audio major in their degree programs. My main interest was recording, so I took as many audio courses as I could. The classroom, although very useful, didn't teach me everything I wanted to learn, and the labs allowed only minimal hands-on-training. I knew I could not get enough practical experience to make my learning worthwhile, so I took out a loan and bought some very basic 4-track semi-pro equipment to study on. I then bought a copy of *The Recording Studio Handbook* and was able to apply much of what I was learning, thus leading to a better understanding. Being a comp major helped because I had my own music to experiment with on the equipment. After weeks of experimenting, I placed an ad on the school bulletin board for demo recording at $5.00 an hour. Although my setup was very basic, I was still able to crank out some high quality demos after all of my experimenting. Now I was not only making enough money to keep up payments on my loan, but each time I would study a new piece of equipment that interested me, I had enough cash to go out and buy it. As you can plainly see, I was able to keep learning while expanding. The more I equipped my setup, the more I learned, the higher I was able to raise my rates, and the faster I was able to buy more equipment. (I must also mention that my subscription to your magazine helped both technically and business-wise.)

I have long since finished school and have an elaborate 4-track semi-pro setup that can compare in final quality to some 24-track studios, and am making high quality records.

My advice to fellow readers who want hands-on-training is: Don't wait for someone to give it to you. Go after it yourself. The money you would spend for college tuition or recording school could be spent on equipment and educational material such as *The Recording Studio Handbook* and *Modern Recording & Music* instead. Play with your equipment and know it well. Do a lot of experimenting and ask a lot of professionals a lot of questions. You'll get training, education, and experience this way.

—Taylor Sappe
Captain Blue Music

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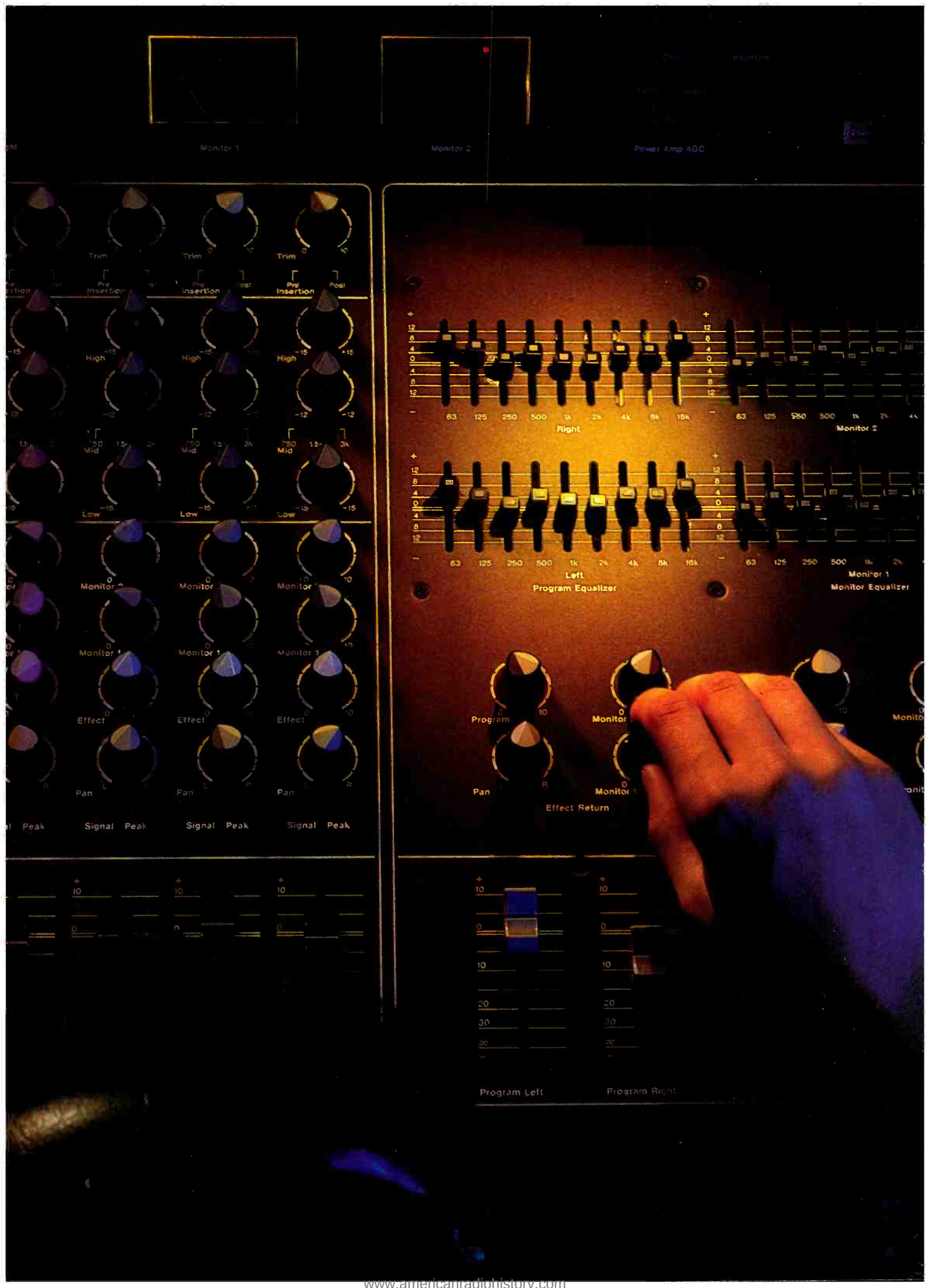

Announcing the mixers that the competition has been trying to build for ten years.

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All this sophisticated technology isn't worth much if you can't use it easily and comfortably, so we put an enormous amount of thought into how our mixers work in actual performance situations.

For example, our knobs are color-coded in different shades of blue, so you'll recognize them no matter what kind of light you're working under. And each one is a raised pointer that lets you feel its setting without looking down. All inputs and outputs are specially designed to eliminate interface problems with outboard accessories. And the meticulous circuit layout and thorough engineering practices eliminate the unpredictable grounding problems you get with other mixers.

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Talk Back

Pro Input

I must join the ranks of readers who love your (our?) magazine. However, I have noticed the decrease in "beginner" type questions in the "Letters" and "Talkback" columns. I think most readers are like me and look through back issues for answers.

This leads me to my question. The October 1979 issue contained an interview with John Storyk, a studio designer. His comments on the home studio were based on how much room was needed for x number of people. Your current "design a small studio" contest is more practically asking how x amount of room will stretch for however many people. How about including the pro designers opinions of the limitations we are faced with? I intend to build from scratch as soon as I get a little more peer input.

—Ronald Harless
Mt. Carmel, TN

We received the following reply from James Rupert.

That's a coincidence, Ron. I'm planning on building from peer input myself, but I'm waiting until I get a little more scratch.

I'm glad you asked the question anyhow. The pro designers and architects that will be participating in the contest judging will indeed be putting forth their opinions on the limitations that face the small studio builder today. (In fact, try and stop these guys from giving their opinions, for crying out loud!) Hopefully, between the professional designers and some of the excellent entries we've received, you'll be able to find just the information you need and want.

As far as the decrease in "beginner" letters, maybe *MR&M* has done its job so well that everybody who reads it regularly is asking smarter questions now! What d'ya think, huh?

Instrumental Information

I have some ½-in. Ampex 766 magnetic instrumentation tape I have been saving for when I go to 8 track. How will this tape work for recording?

—R. E. Lester
Harrisburg, PA

We received the following response from Cher Mladinich of Ampex's Magnetic Tape Division.

The ideal tape for this application would be an audio mastering tape, with thick oxide coating on 1.5 mil backcoated base film. Use of any other type of tape would result in performance degradation.

Specifically, the thick coating provides long wavelength (low frequency) performance. The thick 1.5 mil base film enhances print-through properties, and the backcoating provides better tape handling dynamics on the transport.

766 instrumentation tape has a thinner oxide coating on a non-backcoated 1.0 mil base film. Instrumentation oxide particles are selected for uniform output at short wavelengths (high frequency), and do not have the excellent distortion characteristics of quality audio mastering tapes.

Instrumentation tapes are low in abrasivity and will not harm audio heads, but they will not give the excellent, optimized full spectrum results of a quality audio mastering tape.

Shorting a Shure

I use a Shure SM10A Headset Microphone for live performance singing. With the mic always about a half-inch from my mouth, I'd like to use an off/switch to eliminate stage talk, tempo counts, etc. My dealer advises shorting the mic line to ground, to avoid picking up RF, since a long mic snake might act like an antenna. He also suggested adding a capacitor to eliminate switch clicks. What do you think?

I would also like to build or buy a Y cord for plugging two microphones into one input, especially when mic'ing drums. After talking to several people, I have heard that this could cause phase problems or microphone damage if one microphone activated the other one. Is this true?

—Steve Leonardo
Rochester, New York

Kerry Kendrix in Shure's Customer Service Department tells us that the recommended wiring configuration is to short mic lines 2 and 3 together. It is not necessary to short either conductor to ground, or to use capacitors. A Switchcraft T3F plug may be used for this application. The T3F is a female mic line plug, with a built-in on/off switch. If you would prefer a momentary-contact switch, specify T3FM instead.

As for the Y connectors, Kendrix recommends using no more than two identical microphones, and making sure that the pins in the male plug are wired to the corresponding-number pins in each female plug. Two microphones will drop the level about 6 dB, with even more loss if additional microphones are Y'ed to the same input.

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bruce bartlett

What Is Good Sound?

Seat an engineer behind a mixing console and ask him or her to do a mix. It sounds great. Then seat another engineer behind the *same* console and again ask for a mix. It sounds terrible. What happened?

The difference lies mainly in their *ears*—their critical listening ability. The first engineer has a clear idea of what he or she wants to hear and how to get it. The second engineer hasn't acquired the ability to recognize good sound.

That ability is essential. By knowing what to listen for, you can improve your artistic judgements during recording and mixdown. You'll be able to hear errors in microphone placement, equalization, etc., and correct them.

Training your hearing involves dissecting recorded sound into its components—such as frequency response, noise, reverberation—and concentrating on each one in turn. It's easier to hear sonic flaws if you focus on a single aspect of sound reproduction at a time. That's what we'll do in this article.

The $64 Question

How do you know when you've got it right? It's hard to give a definite answer, because what is considered "good sound" changes with the times. What was acceptable in the '50s and '60s won't do today. In past decades, many record productions were heavy on midrange, reverb, and echo. Since then, recordings have gradually become wider-range, smoother and clearer, with less reverb.

If the aim of a recording is realism or accurate reproduction, you know the sound is good when it matches the live performance heard in the best seat in the concert hall. The sound of musical instruments is the standard of quality.

But when the goal is to enhance the sound or produce special effects (as in most pop music recordings), the aesthetic is less well-defined. The live sound of a pop group could be a reference, but pop music recordings generally sound better than live performances. Recorded vocals are clearer and less harsh, the bass is

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cleaner and tighter, and so on. The sound of pop music reproduced over speakers has developed its own standards of quality apart from accurate reproduction.

Currently, a good-sounding pop recording can be described as follows: clean, wide-range, tonally balanced, and "tight," with a good mix. Quality recordings are also smooth and spacious, with sharp transients, wide and detailed stereo imaging, presence, and clarity. Dynamic range is wide but controlled, and special effects are creative and tasteful.

Let's explore each one of these qualities in detail so we'll know what to listen for. We're assuming the monitor system is accurate, so that any colorations heard are in the recording and not in the monitors.

Clean

"Clean" means free of noise and distortion. Tape hiss, hum, and distortion are inaudible in a good recording. "Distortion" in this case means distortion added by the recording process, not distortion already present in the sound of electric guitar amps or Leslie speakers. Clean sound results from proper recording levels, proper gain distribution in the console, well-calibrated tape machines, good tape, noise reduction, a quiet studio, and proper grounding and shielding techniques.

Low-frequency filtering removes rumble from trucks and air conditioning, as does a rolled-off low end in a microphone's frequency response. Breath pops in vocalists' microphones are prevented by using windscreens and by placing the microphones at eye level.

"Clean" also means "not muddy"—free of low-frequency overhang and leakage. You keep bass sounds under control by recording the bass direct and keeping the bass amp volume low. You reduce leakage by mic'ing close or by overdubbing.

A "clean mix" is one that is uncluttered, free of excess instrumentation. This is achieved by arrranging the music so that similar parts don't overlap, and not too many instruments play at once.

Wide-Range

"Wide-range" means extended low-frequency and high-frequency response. Cymbals should sound crisp and distinct, but not sizzly or harsh; kick drum and bass should sound deep, but not overwhelming or muddy. Wide-range sound results from using high-quality microphones and recorders, high speed tape, and clean tape heads.

Tonal Balance

The overall tonal balance of a recording should be neither brassy nor trebly. That is, the frequency response or spectrum should not emphasize low frequencies or high frequencies. Low bass, mid-bass, midrange, upper midrange, and highs should be heard in equal proportions. Emphasis of any one frequency band over the others causes listening fatigue after a while.

Good Mix

In a good mix, all the instruments and vocals are in a pleasing loudness balance with each other. Everything can be clearly heard, yet nothing is obtrusive. The most important instruments or voices are loudest; less important parts are in the background.

How do you know when the mix is right? When you don't notice it. That is, when you have all the tracks balanced correctly, nothing sticks out and nothing is hidden.

Sometimes you don't want everything to be clearly audible. In rare occasions you may want to mix in certain tracks very subtly for a subconscious effect.

The mix must be appropriate for the style of music. For example, a mix that's right for rock music usually won't work for country music. Rock mixes typically have the drums way up front and the vocals only slightly louder than the accompaniment. Country mixes, on the other hand, have the vocals loudest, with the drums used just as "seasoning" in the background. This distinction is lessening as country music is approaching a pop sound.

Level changes during the mix should be subtle, or else instruments will "jump out" for a solo and "fall back in" afterwards. Move faders slowly, or set them to preset positions during pauses in the music. Nothing sounds more amateurish than a solo that starts too quietly and then comes up as it plays. You can hear the engineer working the fader.

Smoothness

Now we get into some subtler aspects of sound. "Smooth" means easy on the ears, not harsh, uncolored. Sibilants or "s" sounds in vocals sound clear but not piercing. A smooth, effortless sound allows relaxation; a strained or irritating sound causes muscle tension in the ears or body.

Smoothness is a lack of sharp peaks or dips in the frequency response. This results from (1) using high-quality microphones, (2) using equalizers with a broad "Q," (3) preventing phase interference between microphones, and (4) avoiding excessive boost in the midrange or upper midrange.

Presence

This is the apparent sense of closeness of the instruments—a feeling that they are present in the listening room. Synonyms are clarity, detail, punch. Presence is achieved by close mic'ing, overdubbing, and using microphones with a "presence peak" around 5 kHz. Upper midrange boost helps, too. Most instruments have a frequency range which, if boosted, makes the instrument stand out more clearly or become better defined. This range is around 5 kHz for most instruments, 2 to 6 kHz for vocals, 1.5 kHz for bass, and 2.5 kHz for kick drum.

Note that "presence" sometimes conflicts with "smoothness." You'll have to find a tasteful compromise between the two.

Clarity

Instruments should not "crowd" each other's sound. They should be separate and distinct and should blend well. This quality is subtle to hear and difficult to achieve—a real art. Clarity arises when instruments occupy different areas of the frequency spectrum. For example, low frequencies are provided by the bass; mid-bass might be emphasized by the keyboards; upper midrange may be provided by lead guitar, and highs are filled in by the cymbals.

Rhythm guitar often occupies the same frequency range as piano, so they tend to mask each other's sound. You can increase clarity by boosting the guitar around, say, 2 kHz, and boosting the piano around 7 kHz.

Usually, the fewer the instruments, the clearer the sound. Too many overdubs can muddy the mix. Also, it helps to delay signals feeding a reverberation unit by about 20 milliseconds. This removes the onset

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of reverberation from the direct sound, helping to clarify the sound.

Spaciousness

"Spacious" or "airy" means "having a sense of air around the instruments." Without air or ambience, instruments sound like they're isolated in stuffed closets. Here are some ways to achieve spaciousness:

(1) Pick up some sound reflections from surfaces near the instruments. If leakage is not a problem, put microphones far enough from instruments to pick up surface reflections.

(2) Add artificial reverberation or echo to dry tracks.

(3) Allow a little leakage between microphones. Put microphones far enough from instruments to pick up off-axis sounds from other instruments. Don't overdo it, though, or the sound will become muddy and track separation will be poor.

(4) Allow a little crosstalk between the left and right channels. If tracks are totally isolated, it's hard to achieve the illusion that all the instruments are playing in the same room at the same time. You need some crosstalk or correlation between channels. Some right-channel information should leak into the left channel, and vice versa. Stereo reverberation can accomplish this crosstalk. So can a delayed signal panned to the opposite channel. Panning extreme left or right tracks slightly toward center may help, especially in recordings of duets.

Sharp Transients

The attack of cymbals, rim shots, drum hits, and bass guitar plucks should generally be sharp and clear. To record sharp transients, do the following:

(1) Use condenser microphones (rather than dynamic microphones) on acoustic instruments.

(2) Use direct boxes on electric bass. Have the bassist play percussively or use a pick. When compressing the bass, use a long attack time to allow the notes' attack to come through.

(3) Boost high frequencies on percussion around 5 to 10 kHz.

(4) Dampen the kick drum with a pillow or blanket, and mic it next to the center of the head near the beater.

(5) Avoid saturating the tape with excessive recording levels.

Tight Bass and Drums

The kick drum and bass guitar should "lock" together rhythmically so that they sound like a single instrument—a bass with a percussive attack. The drummer and bassist should work out their parts together to hit accents simultaneously.

Equalize the bass and kick differently to keep their sounds distinct. For example, boost the bass guitar a few dB around 60 to 100 Hz to give a sense of deep bass below the kick drum attack. Increase bass guitar definition by cutting around 200-400 Hz and/or boosting around 1.5 kHz. Add presence to the kick around 2.5 to 8 kHz. Damp the kick drum as described above and record the bass direct.

Good Stereo

Stereo means more than just "left" and "right." Usually, tracks should be panned to many points across the stereo stage between the playback loudspeakers. Some instruments should be hard left or hard right; some should be in the center; others should be half-left or half-right. Try to achieve a stereo stage that is well balanced between left and right. To separate instruments occupying the same frequency range, pan them to opposite sides of center.

You may want some tracks to be nonlocalized. Backup choruses and strings should be spread out rather than appearing as point sources. To give a lead guitar solo a fat, spacious sound, send its signal through a delay unit, pan the direct sound hard left and pan the delayed sound hard right.

You should also be able to hear some front-to-back depth. That is, some instruments should sound close or up-front, while others should sound farther away. You achieve depth by mic'ing instruments at different distances, and by using variable amounts of reverberation on each instrument. The higher the ratio of reverberant sound to direct sound, the more distant the track sounds. Set the "pre/post" switch on the reverb send to "pre" to move instruments closer and farther as you move the fader.

Wide But Controlled Dynamic Range

"Dynamic range" is the range of volume levels from softest to loudest. A recording with wide dynamic range gets noticeably louder and softer, adding excitement to the music. This is achieved by using noise reduction and by avoiding excessive compression. An overly compressed recording sounds "squashed"—crescendos and quiet interludes lose their impact.

Some compression or gain-riding is needed for vocals because their dynamic range exceeds that of the instrumental backup. A vocalist may sing too loudly and blast the listener, or sing too softly and become buried in the mix. A compressor can even out these extreme level variations, keeping the vocals at a constant loudness. Bass guitar also benefits from compression.

Interesting Sounds

The recorded sound may be too "flat" or neutral—lacking character or color. You can make your recordings more of a "production" by using special effects—equalization, echo and reverberation, doubling, chorus, flanging, or compression. Use and combine equipment in unusual ways. Overdub little vocal licks or synthesized sound effects.

Note that making sounds "interesting" or colorful can conflict with accuracy or fidelity, so use effects and equalization with discretion. It's possible to produce, say, a highly equalized snare sound that still is wide-range and well balanced.

A record that excels in *all* the attributes of "good sound" is "The Sheffield Track Record" (Sheffield Labs, Lab 20), engineered and produced by Bill Schnee. In effect, it's a course in state-of-the-art sound—required listening for any recording engineer or producer.

Another record with brilliant production is *The Nightfly* by Donald Fagen (Warner Brothers 23696-1), engineered by Roger Nichols, Daniel Lazarus, and Elliot Scheiner; produced by Gary Katz, and mastered by Bob Ludwig. The sound is razor sharp, elegant and tasteful; and the music just pops out of the speakers.

Classical Music Considerations

Like pop music, classical music should sound clean, wide-range, and tonally balanced. But since classical recordings are meant to sound realistic—like a live performance—they also require tonal accuracy, good acoustics, balance, perspective, and localization accuracy. Let's look at each of these requirements.

Tonal Accuracy

The reproduced timbre or tone quality should match that of live

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instruments. To ensure tonal accuracy

(1) Use flat-response microphones placed at a reasonable distance from the ensemble.

(2) Avoid microphone positions that pick up standing waves or room modes. Experiment with small changes in microphone positions.

(3) Avoid phase cancellations due to reflections from hard surfaces near the microphones. You may want to use PZMs or other surface-mounted microphones.

Good Acoustics

The acoustics of the concert hall or recital hall should be appropriate for the style of music to be performed. Specifically, the reverberation time should be neither too short (dry) nor too long (cavernous). Too short a reverb time sounds amateurish or cheap, lacking spaciousness and depth. Too long a reverb time blurs notes together, giving a muddy, washed-out effect. Ideal reverb times are around 1.2 seconds for chamber music or soloists, 1.5 seconds for symphonic works, and 2 seconds for organ recitals.

Balance

When a recording is well balanced, the relative loudness of instruments is similar to that heard in a good seat in the audience area. For example, violins are not too loud or soft compared to the rest of the orchestra; harmonizing or contrapuntal melody lines are in proportion. Usually, you can achieve a good balance by recording with two or three microphones placed several feet in front of and above the ensemble.

Note that the instruments are balanced acoustically by the conductor, composer, and musicians—rather than being mixed on a mixing board. There are exceptions to this rule, as some recording engineers use multiple mic'ing for classical music.

Perspective

This is the sense of distance of the performers from the listener; how far away the stage sounds. The closer the microphones are to the performers, the closer they sound in the recording. Find a microphone position that conveys an appropriate perspective for the piece of music. Incisive, rhythmically motivated works sound best with closer mic'ing; lush romantic pieces are best served by more distant mic'ing.

If the microphones are too close, the sound is edgy, overly detailed, dry. If the microphones are too far, the sound is blurred, over-reverberant, muddy. Do the performers sound like they're eight rows in front of you, or in your lap, or in another room?

Closely related to perspective is the amount of recorded ambience or reverberation. Settle on a microphone distance that yields a pleasing balance of direct sound from the orchestra and ambience from the constant hall.

Accurate Localization

Reproduced instruments should appear in the same relative locations as they were in the live performance. Instruments in the center of the ensemble should be heard in the center between the speakers; instruments at the left or right side of the ensemble should be heard from the left or right speaker. Instruments half way to one side should be heard half way off-center, and so on.

To achieve accurate localization, monitor the sound and adjust the angle or spacing of the stereo pair of microphones until localization is accurate. Or, use a stereo microphone with adjustable stereo spread. Angling or spacing the microphone pair closer together reduces stereo spread, and vice versa.

The reproduced size of an instrument or instrumental section should match its size in real life. A guitar should be a point source; a piano or string section should have some spread.

Reproduced reverberation (concert hall ambience) should either surround the listener, or at least it should spread evenly between the speakers. An even reverb spread occurs when the instrumental localization is accurate.

On playback, you can make the ambience surround the listener by using the Madsen *ambience extraction* playback method: connect two additional speakers to the power amp outputs, and place them on either side of the listener, 3 to 12 feet farther from the listener than the front pair of speakers. Adjust the level of the side speakers with an L-pad or rheostat, so that the side speakers are as loud as possible without becoming noticeable as separate sound sources. This method immerses the listener in concert hall ambience and greatly increases depth and realism. Try it!

Suitable Sound

A general rule of aesthetics that applies to recording is, "The medium is the message." The way a recording sounds should imply the same message as the musical style or lyrics. In other words, the sound should be appropriate for the particular tune being recorded.

For example, some rock music is rough and raw. The sound should be, too. A clean, polished production doesn't always work for high-energy rock and roll. There might even be a lot of leakage and ambience to suggest a garage or nightclub environment. The role of the drums is important, so they should be loud in the mix. Toms should ring.

Disco,r&b, or MOR music is slickly produced. The sound should be tight, smooth, and yet spacious, to suggest luxury and taste. Drums should be damped.

Country music is about feelings, so the bass and vocals are emphasized for warmth and emotion. Acoustic guitars and drums are mic'ed at a respectful distance, giving an airy, natural effect.

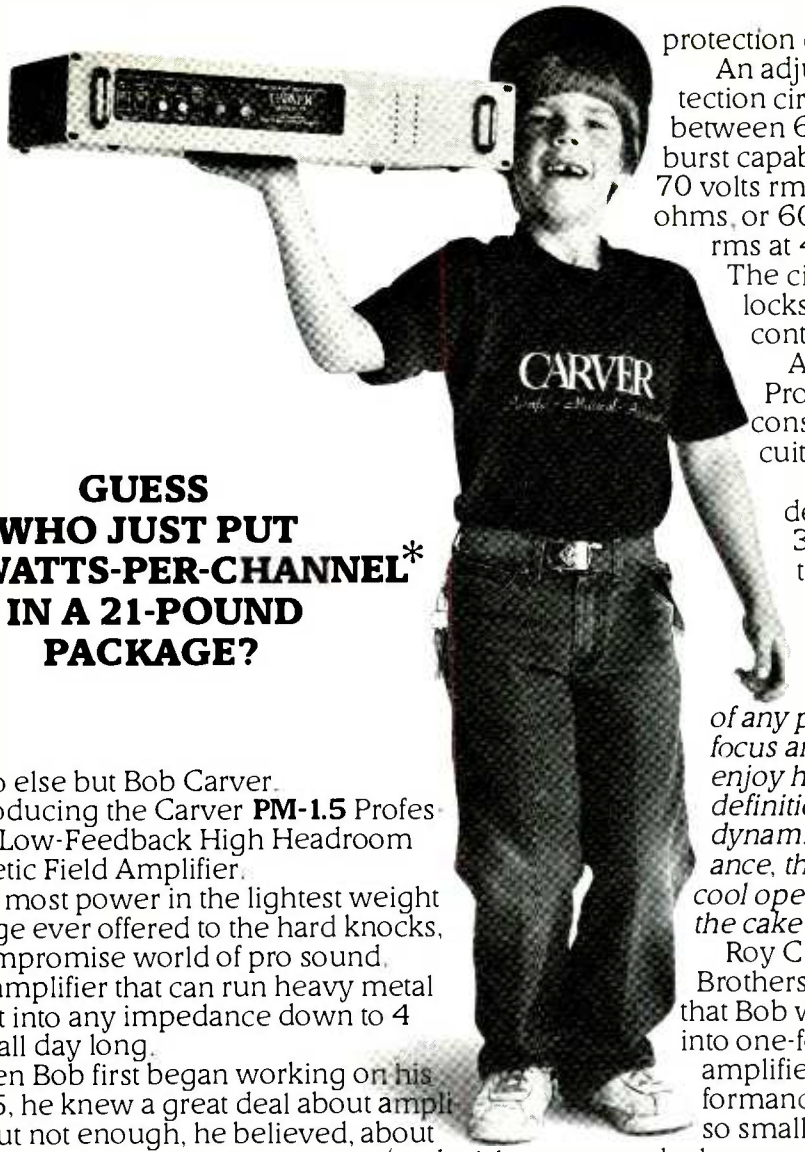
Actually, each style of music is not locked into a particular style of production. You tailor the sound to complement the music of each individual tune.

When you don't know what kind of sound to go for, keep the tonal balance natural—or a little brighter—and you'll be in the ballpark. For example, vocals should sound like someone actually singing in front of you, not like someone singing through a P.A. system. An acoustic guitar should sound like it's playing out in front of you, not like the sound hole is stuck next to your ear (unless that's the desired effect).

Developing an analytical ear is a continuing learning process. Train your hearing by listening carefully to recordings—both good *and* bad. Listen for all the qualities mentioned in this article. Compare your own recordings to live instruments, and to commercial recordings, to see what you're doing right or wrong.

Once you're making recordings that are technically competent—clean, natural, and well mixed—the next stage is to produce imaginative sounds. You're in command; you can tailor the mix to sound any way that pleases you or the band you're recording.

Finally, the supreme achievement is to produce beautiful-sounding recordings.



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When Bob first began working on his PM-1.5, he knew a great deal about amplifiers but not enough, he believed, about the night-in night-out requirements (and wish dreams) of the pro sound world. And so he spent much time consulting with a large number of sound reinforcement professionals, including the "pros' pro," Clair Brothers.

Clair Brothers asked for lower input ac line current, greater transformer thermal capacity, dual modes of precision balanced inputs with 1% resistors, back-to-front cooling with a fully proportional fan system that can just tick over at idle or blast 1000 ft/min. to keep output transistor temperatures constant. They wanted greater noise immunity and unbelievable long-term, high-power operation, as well as a 3/16" front panel with deep-recessed controls.

Others wanted rear rack-mounts, adjustable protection circuit thresholds, front panel selectable clipping eliminator, and even a sequential, soft-start power-up mode.

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The PM-1.5 is designed to run balls-to-the-wall, even into 4 ohms day and night without compromising itself or your drivers. Because the PM-1.5 has three special adjustable speaker protection circuits.

An adjustable Short-Term Speaker Protection circuit adjusts dynamic headroom between 60 and 77 volts with a short term burst capability of 70 volts rms at 8 ohms, or 60 volts rms at 4 ohms. The circuit locks-in to the control setting after 0.5 seconds.

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Roy Clair and Ron Borthwick of Clair Brothers said it this way, "We are amazed that Bob was able to put the same wattage into one-fourth the volume of conventional amplifiers without sacrificing audio performance. It's hard to believe that an amp so small and lightweight can put out so much clean power. But it does!"

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*Power: 8 ohms. 450 watts/chan. 20 Hz-20 kHz both channels driven with less than 0.1% THD. 4 ohms. 600 watts/chan. rms 20 Hz-20 kHz both channels driven with less than 0.1% THD. 16 ohms. 300 watts/chan. 20 Hz-20 kHz both channels driven with less than 0.1% THD. 2 ohms. 525 watts/ chan. at clipping. 1kHz. with less than 0.2% THD. Note: 2-ohm specification for information purposes only. Operation at 2 ohms is permissible but not recommended. **IM Distortion:** Less than 0.1% SMPTE. **Frequency Response:** −3 dB at 3 Hz, −3 dB at 80 kHz. **Damping:** 200 at 1 kHz. **Gain:** 26 dB. **Noise:** Better than 115 dB below 450W A-weighted. **Input:** Balanced to ground. XLR or phone. **Impedance:** 15k-ohm each leg, balanced to ground. **Bridging:** 1200W into 8 ohms. 1000W into 16 ohms, accessed through rear-panel recessed switch. **Dimensions:** 19 in wide. 3½ in high. 10 13/16 in deep. Weight: 21 lbs.

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Studio Notebook

james rupert

Editor's Note:

The following manuscript arrived at the Modern Recording & Music offices inside of an empty "Burpsie Cola" bottle, having been delivered by a Boy Scout who saw it being thrown from an upper story window of one of the seamier midtown hotels on his way to school. The Scout reported that the balding and bearded man who threw the manuscript from the building motioned for him to pick it up and take it to the address that was scotchtaped to the outside of the bottle. As the scout fled the scene he looked back to see the man also throwing 400 cigarette butts and several empty "Lickin-Chicken" boxes from the same window. After careful examination of the evidence (particularly the testimony about the cigarette butts), we have been forced to the conclusion that the figure in question was indeed our boy Rupert. MR. & M will be keeping you advised of Rupert's situation as soon as more details become available.

Whoever finds this manuscript is hereby instructed to deliver it ASAP to the editorial offices of ***Modern Recording & Music*** *magazine in Long Island, New York. Deliver this manuscript personally to Ms. Ricki Zide, who will be so relieved to hear that I'm alright she'll probably give you a big reward! (Ed. Note: We slipped the kid a buck.)*

I'm being held captive by all of the entries in the 'Design-A-Studio' Contest and am unsure of my present location. The entries seem to have taken on a collective personality all of their own and a demanding one at that. They want all of my time and attention. The situation has grown so bad that they have drugged me while I slept (as if they had to, the way I sleep,) and locked me in this cheap room in a low-rent dive somewhere with continuous entertainment downstairs. They are insisting that I devote my entire life to them with no time for anything else. It's like dealing with monsters. It's like dealing with dictators. It's like dealing with an editor. What a nightmare!

Consequently, I'm having to sneak this month's installment of the Studio Notebook series out while they're not watching. I've been assembling it on the sly while here in the bathroom (sorry about the paper, I couldn't find anything else to write on in here) because it's the only place they let me be alone.

While we're waiting for the results in the design contest, I think it's time we considered a few other particulars in regard to opening our studio businesses. A grand opening is more than just hanging out a sign and licking your collective chops in anticipation of the money that is about to come sliding in. There are a few boring administrative details that you as a prospective studio owner have to be sure are taken off.

If your studio is not in your home, have you made any arrangements with your local utility companies to have the power, gas, and water turned on? Sometimes this means a deposit has to be paid with each utility office to make sure that in the future you don't fold up your tent and steal away to leave them weeping over your unpaid bill. Perhaps a parent or business partner has a good enough past credit history with the utilities so that their signature may be good enough to replace a money deposit. In some cases you may be able to pay a flat fee per month that can save you some dollars in the long run. The catch to this might be that you have to pay your flat monthly fee in advance to qualify for the program. Yet think about how much electricity you'll be chomping each month and the advantages of such a plan will become more obvious.

There may be additional special programs and options that your local utility companies can offer, but you've got to work out the details that will work the best for you. Don't put this off until the last minute. You could discover that the soonest they can hook up power or water or gas or oil is two weeks after you've got to have them. (Try running a studio without a toilet for a month and tell me how much fun it was.) Opening a business is no time to catch sleeping sickness.

If your studio is in your home, sit down with your partner(s) and decide what percentage of your home utility bills are going to be paid from studio funds. (This is an exceptionally bad time to wait until tempers flare into a problem before you handle it.) Unless you're planning on eating the higher bills personally, get an agreement with your peers in writing so everybody is singing out of the same hymn book.

Next, call for bids from area printing firms to have your letterhead stationary, envelopes and business cards made up for you. Remember that printing

companies have a huge outlay of mazuma for their equipment just like you do. Think of all of the times you've gritted your teeth at starchbrains who thought your studio prices were much too high before you walk into a printer's shop and turn into Mr. Potato-Head yourself. By calling around for samples and bids, you can get the best price without having to insult another businessperson by insinuating that their prices are so high they stink up the joint. Also keep in mind that for good quality printing, you do indeed get what you pay for.

When you've found the printer that's right for you, don't forget those specially printed sales receipts for your company. You'll probably need at least a two part (carboned or carbonless) form so that you can give the customer a legitimate receipt for any monies received or services rendered and keep another copy for yourself and your records. (Receipts and records are a pain, but you gotta have them.) A good printing firm will have several formats to choose from. If nobody can please you locally, there are excellent mail-order business-printing companies that stand ready to serve your needs. Shop around and see what you think.

While you're out shopping, stop in at the bank that finally ponied up the bucks to get your studio flying and open a business checking account. Since the bank was good enough to trust you with a pile of their dough, the least you can do is to use their fine and necessary business services. Once you've ordered your checks to be printed, do whatever is required to set up a line of credit with the bank. This is a pre-arranged amount of money that you can borrow automatically and write checks against. (Pretty neat, huh?) Before you can make such an arrangement, the bank will have to review your credit history, the extent of your assets as a studio owner, and your own personal character to see how much faith they are willing to place in your business venture. If this line of credit cannot be worked out, be assured that it is no shame on you or your studio. As money grows tighter and tighter, the fewer and fewer of such credit arrangements will be seen.

This is also the time that an opening inventory will have to be ordered. You've got to have tape to have a recording studio, right? How about reels? Splicing tape? Reel flanges and hubs? Cassette shells? These are inventory items directly related to the production of product. What about those non-product related but still necessary items like head-cleaning solution? Coffee? Toilet tissue? Vacuum cleaner bags? Trash bags? Don't forget the sundry printed materials like track sheets and work orders. See a pattern forming? Some of these things can be lived without, but sooner or later you're going to at least wish you had them around—if not absolutely require them.

How about garbage service? (Are you planning on making a trip or two to the city dump each week?) Cable TV would be nice to have in your lounge, wouldn't it? Got that insurance bought on your equipment and building yet? (More on insurance in the months to come.) Has the furnace and air conditioning been checked out and safety inspected yet? Fire extinguishers mounted throughout the building? When is the phone company coming in to hook up your phone(s)? Do you know where you want the phones in your building? Does the phone company want a money deposit, too?

You may also want to look into obtaining a post office box. This is especially true if you are trying to run a business out of your home. The post office box will technically become your legal business address. If you are looking for a more permanent free-standing location for your studio, the P.O. box can make for a nice sense of continuity while the transition and move is being made. If any aspect of mail-order business is being contemplated, then a P.O. Box is a must.

Have you looked into placing your Yellow Pages ad? My own experience tells me that you cannot have any more vital means of advertising than a listing in those holy pages in the back of the phone book. Most customers, even if they have had dealings with a recording studio previously, will do a little "finger-walking" when the time comes to find a recording service again. I'm not advocating grabbing the largest ad you can get your paws on, but it is important that you be listed. (You can be sure that the competition will be in there!) Perhaps even a multiple listing under different categories such as "Recording Studios," "Audio/Visual Production," "Tape Duplication," "Recording Tape and Supplies," or any of a half dozen others might be in order. Your local Yellow Pages Representative can give you all the aid you'll need in finding the right sized ad and number of listings to best serve you. Costs of the ads are not just based upon size, but upon how many people live in the area that the phone book is being published to cover. A phone book that covers an area of 100,000 people will charge considerably less for the same ad than a directory that includes a half a million. It all depends on how many readers will potentially be able to see your ad.

The charges for a Yellow Pages ad are split up into twelve monthly installments with each installment payable with your monthly phone bill. In essence, you receive twelve months of interest-free credit on an ad that appears from the first day the directory is distributed. Where else can you purchase an ad that is higher in effectiveness at such a reasonable cost that gives you a year to pay it off? You've still got to carefully plan the information that you present in the ad to be as attractive, professional and enticing as possible, but even on a very limited budget there should be an ad size and format that will suit you down to the ground. ***Please*** *don't overlook the Yellow Pages when planning your advertising budget for the next year. It is possible to succeed without it, but it is a lot easier to prosper with it. Check with your local representative on deadlines for copy and reservation of space.*

I can't make any bones about the fact that to take care of all of the above and everything else that you might have thought of is going to be a real time-consuming task. You'll be filling out enough forms to make your head swim, but it's got to be done. Nobody ever said that being a studio owner would be all beer and skittles. (I never said it anyhow... Hell, I don't even know what it means!) When you open a well-planned, smooth running recording operation, hopefully it will all be worth it.

That's all for this time. The 'Design-A-Studio' entries are getting ugly outside the door here and I've got to run. With any luck somebody will find this toilet paper manuscript and get it to ***MR&M*** *safely. If they don't, this whole article could be wiped out!*

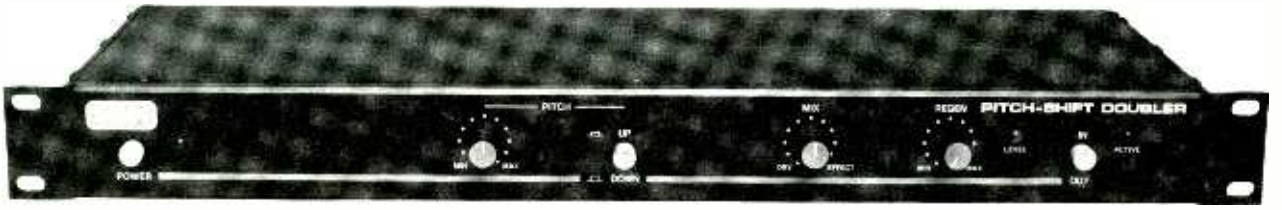
Somebody send me a cheeseburger.

See you next time (I hope).

Musicians Notebook

craig anderton

MXR Pitch Shift/Doubler



I'm sure that most of you have heard the story of the ugly duckling and how, once it found out its true identity, realized it was actually a beautiful swan. That story parallels my experience with MXR's Pitch Shift/Doubler (PS/D for short). Upon first trying it, I thought I had an ugly duckling on my hands. But once I figured out what the device was *really* good for, in certain applications it turned into the signal processing equivalent of a swan. And just what are those applications? Read on....

What is It? First, a little background. Pitch transposing devices, such as the Eventide Harmonizer™ and MXR Pitch Transposer (reviewed in a previous issue of *MR&M*), have been around for quite some time. However, it's interesting to note that many musicians seldom use these devices' full transposition range (usually plus or minus one octave). Instead, they most often set the unit for a slight amount of detuning to create a richer, fuller stereo spread (see *Figure 1*). While you might think that this technique is no different from using a delay line to split a signal into stereo, pitch transposing creates a true detuning effect instead of a simple delay (which does not affect tuning). There is a noticeable difference between these different effects and, to my ears, the stereo spread caused by slight detuning is superior to the spread produced by using a short delay.

MXR recognized that a simplified pitch shifting device would be perfectly adequate for many applications, and proceeded to build a scaled-down, low-cost ($450 list), single rack-space pitch transposer that would transpose over a ±50 cents range (50 cents = one quarter tone). Note that even this limited transposition range is more than enough to create most spreading and doubling effects.

The PS/D has very few jacks and controls, namely:

- High impedance (500k) input jack with associated hi/lo level switch. In the hi position, the PS/D accommodates signals up to +18 dBm, while the lo position handles signals up to +6 dBm. In addition to the rear panel jack and switch, there is also a front panel level-setting two-color LED (green for active, red for peaks).
- "Synthesized stereo" output jacks. Output jack 1 has an associated front panel pot which sets the mix of the effect and straight signal. Output jack 2 consists of the effect signal mixed equally with the straight signal; the effect signal is 180 degrees out of phase with the effect signal present at Output 1. This phase difference synthesizes a spacious sound when the two outputs are sent to opposite channels of the stereo field. However, I recommend using Output 1 most of the time, for reasons we'll get into later.
- Bypassing. There are two ways to bypass the unit: a front panel switch, or a rear panel jack which accepts a remote footswitch.
- Pitch control and switch. The pitch control varies the tuning from just off concert pitch to a quarter tone sharp (with the pitch switch in the up position) or a quarter tone flat (with the pitch switch in the down position).
- Regeneration control. This sends some of the pitch-shifted signal back to the input, which imparts a metallic-sounding "edge" to the PS/D's flanging effects.

You might think that a device with this small a roster of controls would be easy to apply. In one respect that's true; the PS/D is easy to hook up and get working. But making it work right musically is something else altogether.

Evaluating the Pitch-Shifted Sound. All but the most expensive pitch shifters are, by their very nature, plagued with sub-standard sound. This is because the process of electronic pitch transposi-

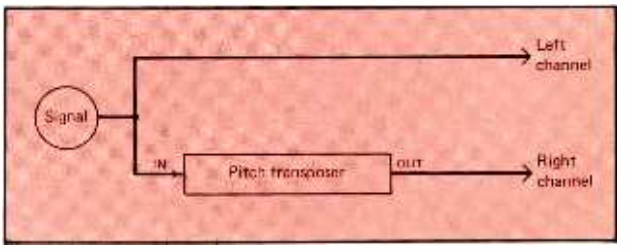


Figure 1: A slight amount of detuning creates a fuller stereo spread.

tion involves tearing a signal apart electronically and splicing it back together again, which produces "splicing glitches" at the splice points. The greater the degree of transposition, the more obvious the glitching. These glitches take the form of discontinuities in the signal, which result in degraded sound quality. Note, however, that in other respects the PS/D's pitch-shifted sound is excellent (frequency response, 50 to 20 kHz ±3 dB; noise, -88 dB).

Because of the splicing glitch problem, it is important to use the PS/D with the right kinds of signals. Processing more-or-less continuous sounds (such as organ, voice, or even guitar in some cases) will accentuate the splicing glitches. However—and this is the important part—it is very difficult to notice the splicing glitches when processing percussive sounds. So, depending on what kind of signals you put through the PS/D, the sound quality can be anything from glitchy to excellent.

Applying the PS/D. The manual lists the following applications: doubling, chorusing, infinite ("barber pole") flanging, feedback suppression, pitch correction, vibrato, and rotating speaker simulation. However, when it comes to traditional doubling, chorusing, vibrato, and rotating speaker simulation, a good digital or even analog delay can do better. Sure, the PS/D does an okay job, but if you try to think of it as a conventional signal processing device, you will probably be disappointed. And although feedback suppression and pitch correction are possible, the splicing glitches negatively affect the overall sound.

What the PS/D *will* do is work wonders with percussive sounds—whether you're interested in stereo spreading, infinite flanging, or the creation of a "Big Drum Sound." Lest you think that sounds rather limited, think again. How many times during mixdown would you have liked to augment a drum sound? Or get a stereo tambourine sound out of a mono tambourine? Or spread a high-hat in stereo so that it's not off to one side of the stereo field? Or flanged a guitar with something other than the old whoosh-

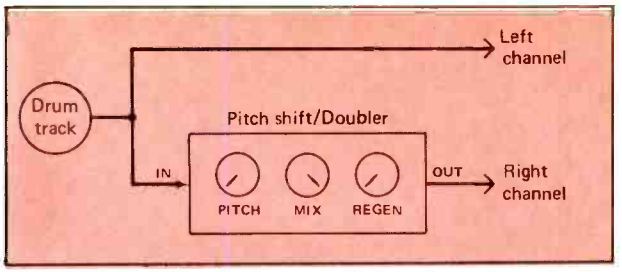


Figure 2: To improve the sound of a mono drum track, patch the pitch shift/doubler as shown.

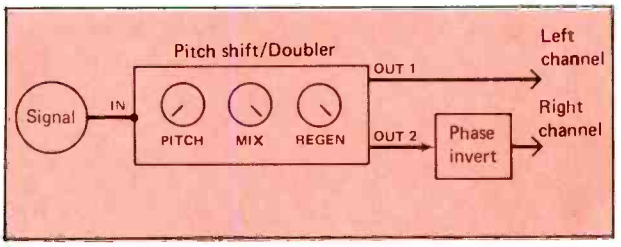


Figure 3: Adding a phase inverter between output 2 and the mixer keeps the flanging effect intact in stereo or mono.

whoosh-whoosh of a standard flanger? Or flanged drums so that the flanging effect gives the illusion of constantly rising or falling? Or, in fact, combined these functions so that you have infinite flanging along with stereo spreading? All of these applications are extremely well suited to the PS/D.

For example, suppose you cannot allot two tracks for drums on your multi-track master and you are stuck with a mono drum part. To improve the sound, patch the PS/D during mixdown as shown in *Figure 2*; then set the pitch knob for minimum pitch shift, the mix control for effect only, and no regeneration. Spreading the drums in stereo during mixdown produces a richer, fuller sound that is far superior to mono drums. In fact, with electronic drum units I often prefer to take a mono track and spread it in stereo using this technique rather than take individual drum outputs, patch them into a mixer, and go for a more traditional stereo spread. Overall, the sound seems somehow larger and prints better on tape. (Note that this does *not* involve using the synthesized stereo outputs; the patch works just fine with Ouput 1 in parallel with a straight signal.)

Turning up the regeneration control introduces flanging effects, with the rate of the flange dependent on the pitch control (the greater the amount of transposition, the faster the flanging sweep). The direction of the flange depends on the pitch switch. With the pitch switch in the up position, the PS/D acts like a flanger being modulated with a positive-going sawtooth wave. With the pitch switch down, the PS/D acts like a flanger being modulated with a negative-going sawtooth wave. When processing percussive signals, the flanging effect gives the illusion of always sweeping up or down, unlike standard flangers which produce a more cyclic effect. While this sounds like it might produce a gimmicky sound, when properly applied the results can be most dramatic.

The synthesized stereo outputs come in handy for flanging effects, but be careful: if you play back the synthesized stereo over a mono system, any flanging effects disappear. I've had good results by setting the pitch control to minimum, the pitch switch to up, mix to effect only, and regeneration on max. Then, I add a phase inverter between Output 2 and the mixer (*Figure 3*). While this cancels some of the non-flanged sound, the advantage is that the flanging effect remains intact whether in stereo or mono. Creating good flanging effects in stereo will require some experimentation—sometimes it's better to use the synthesized stereo outputs, sometimes it's preferable to run the PS/D in parallel with a straight signal (as shown in *Figure 2*).

I don't mean to imply that the PS/D is good only for stereo spreading and infinite flanging. However, the splicing glitches can really be problematic with chorus and rotating speaker effects. If you're playing percussively this will minimize any problems, but it's easier to simply reach for a standard chorus unit instead.

Overall Evaluation. I absolutely love using the PS/D with drums (especially electronic drums) and for infinite flanging effects. Although that may sound like a limited repertoire of tricks, the PS/D does them very well, and besides, when did you last cut a track that *didn't* have something percussive on it? Also, using the PS/D for these percussive applications frees up an expensive pitch shifter you might need for more critical applications, such as vocals.

I should also add that aside from the splicing glitches, the sound quality of the PS/D is quite good. When you spread a track in stereo, the PS/D channel will not sound duller or more distorted than the straight channel. I've used the PS/D on some really demanding percussive program material, yet it has always sounded just fine.

The PS/D is not for everybody, especially studios on an extremely tight budget. After all, something like a good general purpose delay line or parametric equalizer would be more immediately useful. But the PS/D is really quite cost-effective, and if you've ever wanted a pitch shifter but couldn't swing the bucks, the PS/D will give a pitch shifter's "greatest hits" along with some great flanging sounds. And if you're into extensive mixdown processing (many smaller, electronically-oriented studios are), the PS/D can really fatten up thin sounds and turn drums into DRUMS.

Should you decide to try out the PS/D at your local music store, remember that you're not dealing with a conventional signal processor. The PS/D is deceptive; try to use it in conventional ways, and you'll be dealing with an ugly duckling. But if you take the time and effort to find the optimum settings and applications for the PS/D, you might feel—as I do—that the PS/D is really a swan in disguise.

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rick chinn

Compressors and Limiters

This month's column will deal with the ins and outs of compressors and limiters. These extremely useful audio tools can perform as a preventative tool, a repair tool, or as a creative effect. We'll look at each aspect of these qualities in the paragraphs to come.

What is a Compressor and What Does It Do?

Very simply stated, a compressor is an automatic gain-control device. What this means is the output level is nearly constant in spite of variations in the input signal level. One of my favorite analogies is that

of a person with his/her hand on the volume control knob. This person has been told: "As the sound pressure level goes up, turn the volume down so the sound level is the same as before."

A limiter is a special type of compressor. It differs because its action is generally much more severe. The modus operandi is: "Don't let the volume exceed this level...or else!" This very general explanation will suffice—for the time being.

Throughout this month's column, I will refer to compressors without mention of limiters. Don't despair, I'm not opposed to limiters, just lazy (like the rest of us). Everything we're about to discuss applies to limiters, too. If there are exceptions, I'll spell them out. OK...?

What Can I Do With One?

Compressors and limiters are used in many places in the audio chain. Almost always, they are used to protect something from overload. This something might be a loudspeaker, amplifier, broadcast transmitter, or a tape recorder.

Another usage is dynamic range reduction. Instead of preventing something from overloading, the compressor is used to reduce the overall dynamic range of the signal as opposed to putting an absolute ceiling on the signal's level.

A third usage is as a creative tool. They are often used on an electric bass guitar. Here a gentle amount of compression is used to improve the apparent consistency of the bass player's technique. This same technique, used to extreme, increases the amount of sustain. Deliberate misadjustments of the attack and release controls can produce interesting and musically useful effects by deliberately causing envelope distortion and harmonic distortion.

Later in this column, we'll look at some of the ways you can use a compressor or limiter, both as a protective measure and as a creative tool. First, some explanations are in order.

How Does a Compressor Do Its Magic?

Any compressor (or limiter) consists of the following building blocks:

1. A Voltage-Controlled Amplifier or VCA
2. A Control Voltage Processor or CVP
3. Time-Constant Modification circuits

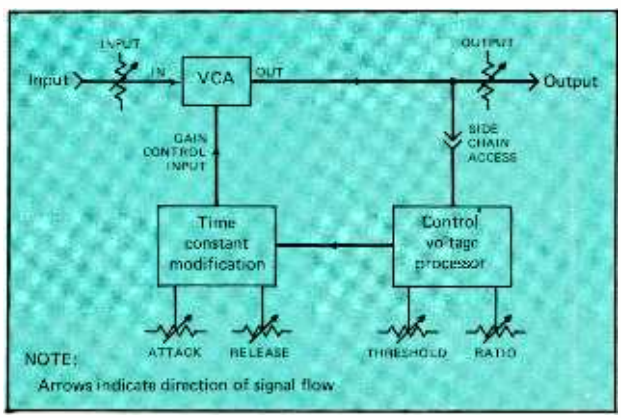

Figure 1: A Feedback Compressor/Limiter.

A Voltage-controlled amplifier is an electronic volume control. Its gain (or amplification factor) is determined by the value of a control voltage applied to its gain-control input. Some VCAs are capable of amplification as well as attenuation, others are capable of attenuation only.

A VCA's circuitry may take the form of a specially designed amplifier circuit, a field-effect transistor and its associated circuitry, a light source and a photocell, or some other circuit element whose impedance can be made to vary by the application of an external voltage.

A Control-voltage processor (CVP) converts an audio signal to a direct-current signal whose value is proportional to the amplitude (voltage) of the audio

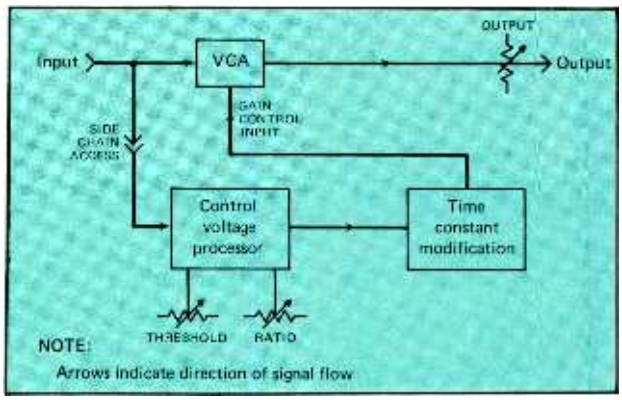

Figure 2: A Feedforward Compressor/Limiter.

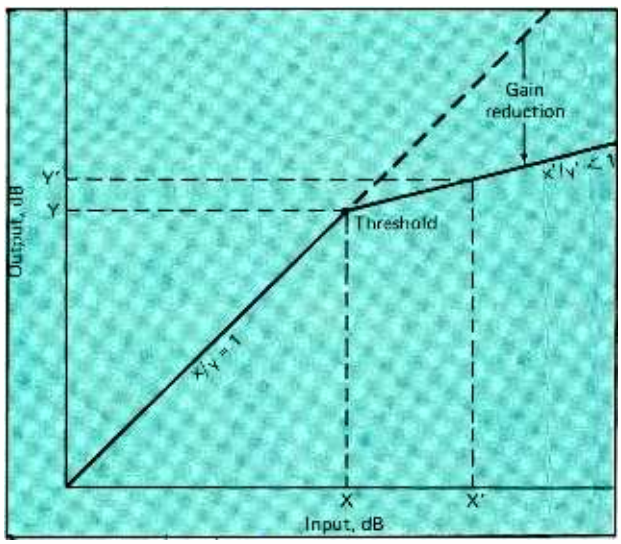

Figure 3: An example of linear amplification (below threshold), followed by compression.

signal. Most synthesizers have a CVP in the form of an envelope follower. In its crudest form, you can think of it as a rectifier. In the CVP's most exotic form, it can actually compute a voltage related to the True-RMS value of the input signal.

The time-constant modification circuit changes the attack and decay characteristics of the DC output of the CVP. In the simplest units, the attack and decay characteristics are fixed (non-adjustable). In more

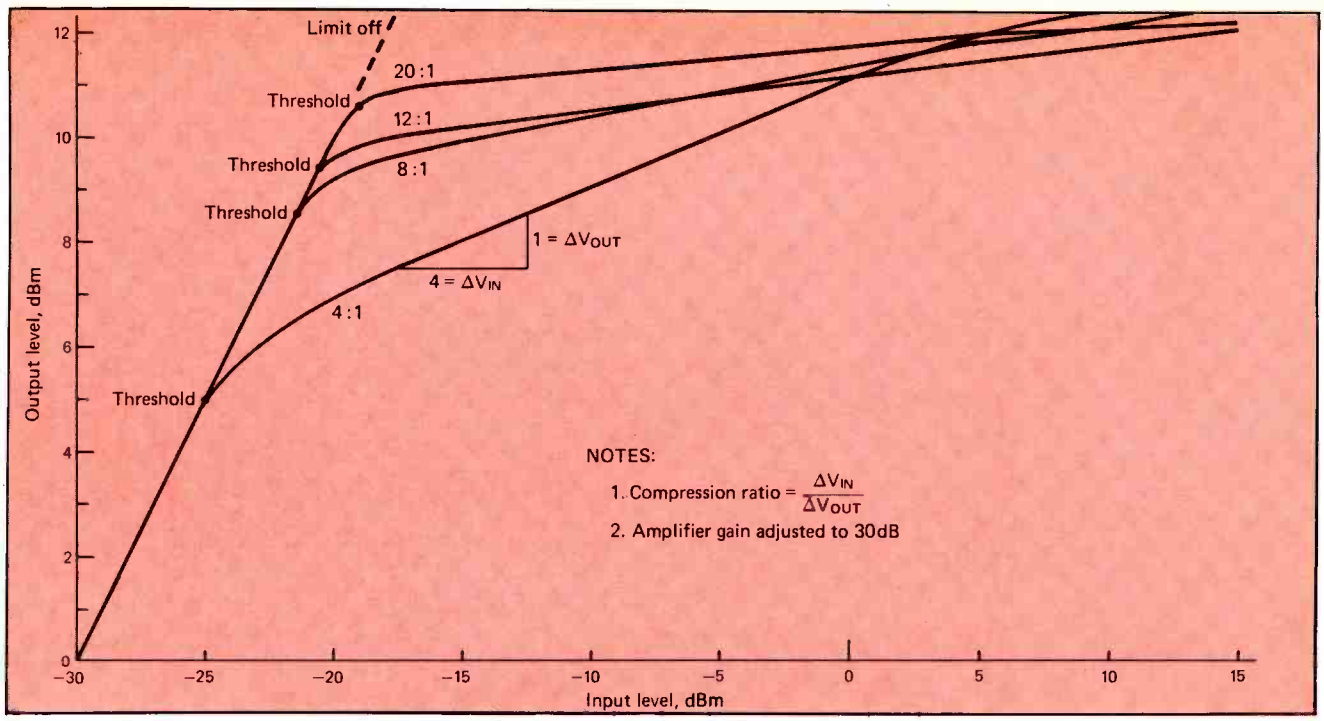

Figure 4: Compression ratios and thresholds for a UREI model 1176 LN Limiting Amplifier.

sophisticated units, the attack and/or decay time may be varied or made program-dependent. The synthesizer analogy to this circuit is the ADSR (Attack-Decay-Sustain-Release) circuit.

Now that the building blocks have been defined, let's build a compressor out of them. There are two basic circuit topologies used today: Feedback (the oldest) and Feedforward (the newest).

In a feedback compressor (*Figure 1*), the audio input signal drives the input of the VCA. The VCA output drives the output jack and the input to the CVP. The DC output of the CVP feeds the time-constant modification circuit which drives the gain-control input of the VCA. As the output signal from the VCA varies, so does the DC output of the CVP, which is applied to the VCA so as to cause its gain to decrease, as the output of the CVP increases. In this manner, the circuit tries to find equilibrium, or a balance between its own output level and the output level of the CVP. This tends to keep the output signal level constant, in spite of variations on the input.

In a feedforward compressor (*Figure 2*), the audio input signal drives both the audio input of the VCA and the audio input of the CVP. The CVP output again drives the time-constant modification circuit, which again drives the gain-control input of the VCA. Since there is no feedback mechanism (as in the feedback compressor), the CVP must produce an output voltage that is directly related to the decibel value of the input voltage, referred to some reference (usually the threshold setting). This voltage is then applied to the VCA, which reduces the gain by precisely that amount.

A Few Terms

When discussing compressors and limiters, a few unique terms always crop up. Let's take a look at them.

Compression. The absolute definition of this term is subject to some debate. For the purposes of this discussion, I will refer to compression as an input-to-output level ratio of less than 10:1.

Limiting. Limiting is nothing more than an extreme case of compression. For this discussion, limiting is defined as an input-to-output level ratio of 10:1 or greater. Many people also associate limiters with fast attack and release times, although a compressor could also have these characteristics. Peak clipping is an extreme form of limiting.

Compression Ratio. It doesn't matter whether or not you are referring to compression or limiting, it's still compression ratio. In the two preceeding definitions, we spoke of the input-to-output level ratio. This is the ratio of the change in the level of the input signal to the change in the level of the output signal or the slope of the input-vs-output curve. These levels are measured in dB. Thus, a compression ratio of 20:1 means that a 20 dB change in the input produces a one dB change in the output.

Threshold. The threshold level is the signal level where the amplification deviates from linear (1:1 input/output ratio) and begins compression or limiting. *Figure 3* shows the input/output relationship of a linear amplifier (before clipping), and then the transition from linear amplification to compression. *Figure 4* shows the effects of various compression ratios, as seen in a typical professional-quality compressor. Also note the differing threshold points.

Attack. The attack time is that required to bring an over-threshold signal under control. Typically, it is measured in micro- or milli-seconds (one millionth or one-thousandth of a second). Although it may seem like the faster the better, this is not always the case... as we shall see.

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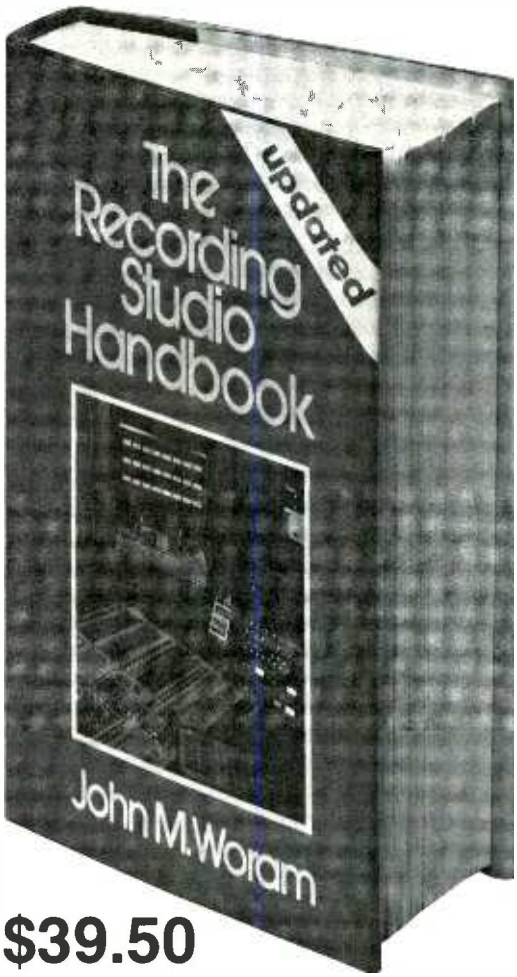

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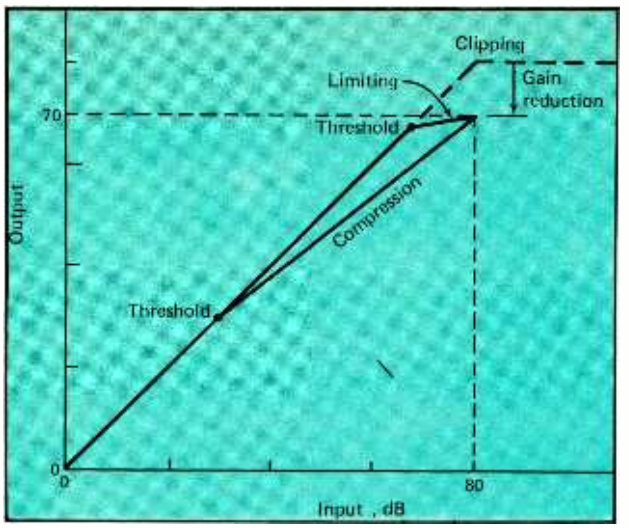

Figure 5: The limiting or the compression curve seen here will compress an 80 dB dynamic range into a 70 dB space.

Release. The release time is that required for the VCA to return to a specified amount of its below-threshold gain. This percentage is usually 63 percent, which is related to the RC time constant at the time-constant modification circuits. Some compressors or limiters have a program-controlled release circuit. This causes the release time to be a function of the peak that caused it. Short duration peaks are released almost instantaneously while longer duration peaks are released more slowly. This is a useful and worthwhile feature to have.

Peak. This refers to the characteristics of the CVP. A peak-responding CVP will deliver an output voltage that is related to the peak, or highest value of the input signal. For absolute overload protection, a peak-responding CVP is the way to fly.

Average. Again, referring to the characteristics of the CVP. An average-responding CVP delivers an output voltage related to the average level of the input signal. This means that short-term peaks are ignored, unless there are many of them. An average-responding CVP does well at handling signals with high peak-to-average ratios (the ratio of the peak level to the average level) without mutilating the signal too badly.

RMS. Once more, this refers to the characteristics of the CVP. The term RMS stands for the mathematical concept of "root-mean-square." The RMS value of a complex signal (read this as speech or music) tends to be more closely related to the average energy content in that signal than the peak or average value. Since the ear responds to the RMS value of its input signal, the proponents of RMS-responding CVPs say their units are less audible, since they work on the same basis as human hearing. On the other hand, an RMS-responding CVP can have slower response time and may not be as well suited to protecting something from overloading on fast peaks.

Gain Reduction. Gain reduction is the amount of compression or limiting taking place at any instant—that is, the difference between what the output is now (with compression) and what the output would have been (without compression). Most compressors or limiters have a meter or other indicator to monitor the amount of gain reduction taking place.

Side Chain. The side chain of a compressor or limiter is the control-voltage processor. Some compressors or limiters allow access to the audio portion of the CVP's signal path. Some very interesting and useful effects may be created by patching a parametric equalizer into the side chain access jacks.

The effect of the outboard equalizer on the compressor or limiter is to make the threshold setting frequency-dependent. If a high-frequency boost is applied, this has the effect of lowering the threshold setting for high-frequency signals. This is a common method for "de-essing"—removing sibilance from vocals.

If the equalization attenuates certain frequencies from the side-chain signal, these frequencies will be *emphasized* in the final output. This technique has come to be known as "vocal stressing."

Where Do I Hook a Compressor or Limiter Up?

Where depends on what you intend to do with it. A few examples should suffice.

There are several possibilities if you are connecting a compressor into a mixing console. The interstage patching point in the input channel is one place. The patching point in the submasters (if any) is another. Of course, there's always the mixer output.

Optimally, the best point to patch is a pre-fader, pre-equalizer patch point. In many mixers, this is just after the microphone preamp. Here, the signal level is unaffected by the channel fader and any channel equalization. It's important that the signal be pre-equalizer because the action of the compressor tends to negate any equalization. Therefore, it's usually more effective to equalize the output of the compressor.

One place this doesn't work is the effects loop. Since this is a parallel connection, the signal that travels through the mixer's normal, direct path will probably mask anything that the compressor is doing.

In an instrument system, it is important that signal levels are matched to avoid blaming a noise problem on the poor compressor. Of course, if your unit was designed for this application, the preceeding caution does not apply. If you are trying to patch the compressor in-between the instrument and its amplifier, a preamp should be used to bring the instrument's signal level up to some semblance of line level (-15 to +4 dBv) to match the input signal *level* requirements of the compressor. Then, an attenuator (which can take the form of a voltage divider) should be used at the output to match signal levels to the input of the instrument amplifier.

Some instrument amplifiers have patching jacks between their preamps and power amps. These jacks are a natural for connecting a compressor or limiter into an instrument system.

In a sound system (PA), limiters are very often used for peak overload protection or to prevent amplifier clipping. If you only have one unit to devote to this function, the only sensible place is at the crossover input (assuming an electronic crossover), otherwise a separate limiter can be used for each crossover output. If you are still using passive, high-level crossovers, then the power amplifier input is the place. The limiter should be the last processor in the signal chain

to the power amplifiers (with the exception of the electronic crossover, of course).

In a broadcast station, compressors and limiters are used to reduce the peak-to-average ratio of the audio signal and to prevent overmodulation. Reducing the peak-to-average ratio helps to make the station sound "loud." Preventing overmodulation helps keep the station from interfering with other stations on adjacent frequencies and keeps the FCC happy. At any rate, the compressor or limiter is connected at the audio input to the transmitter.

Setting the Controls

There are three general types of compressors or limiters on the market today. These vary according to the controls offered to the user. They are:

1. Simple-minded units, usually guitar level effects boxes with a minimum of controls, usually level and sensitivity or something similar.
2. The "standard" collection of controls: input and output level, compression ratio, attack and release time.
3. Many newer designs now offer this collection of controls: threshold, output level, compression ratio, attack and release time.

With the exception of the first type, the controls can be set as follows:

1. Connect the unit into the signal path. Be sure you match levels.
2. Set the ratio control based on what you want to do (limit or compress). Here are some suggestions:
   Bass Guitar—4:1 to 12:1
   Lead Vocal—4:1 to 20:1
   Spoken Voice—12:1 to 20:1
   Anti-Clipping—20:1 or higher
   Other—around 5:1
3. With a typical input signal, adjust the input control until the gain reduction indicator indicates the onset of compression. If your unit has a threshold control in lieu of an input control, start with it at its highest setting (remember that most threshold controls are marked in negative numbers), adjusting it downward until the threshold indicator begins to indicate, or when the gain reduction indicator indicates the onset of compression.
4. Continue increasing the setting of the input control or decreasing the setting of the threshold control until the desired amount of gain reduction is indicated.
5. Adjust the output level control for the desired amount of output.
6. Adjust the attack control for the desired results. Normally, the attack time is set as fast as possible, without causing a popping sound on the initial attack.
7. Adjust the release control for the desired results. Normally, the release time is set fast enough to follow the program material, but not fast enough to cause distortion or pumping.

Applications

As mentioned previously, the compressor or limiter finds its way into many places in the audio signal processing chain. Here are a few.

Making Something Fit. Sure sounds odd, doesn't it? What this means is: I've got a singer who is capable of 80 dB-plus dynamic range, but the tape machine I'm trying to record with is only capable of 70 dB. What do I do with the other 10 dB?

There are two ways to attack this problem. One way is to use a limiter on the last 10 dB of the singer's dynamic range. Another is to use a compressor to reduce the dynamic range of the output, by compressing over a larger input range. The diagram in *Figure 5* should help make this clear.

To take the first approach, pick a high setting for the ratio control (greater than 20:1). Then set the threshold control or input control so that limiting only occurs on the highest peaks. If the tape machine's VU meter still indicates potential overload, either reset the input or threshold control for more limiting and/or use a higher setting of the ratio control.

To take the second approach, use a gentler ratio or slope setting, say 2:1 or 4:1, then set the input or threshold control for about 10 dB of gain reduction on the highest peaks of the input signal.

Pick an attack time that is just fast enough to keep the compressor/limiter from making a sucking sound at phrase beginnings. The release time should be fast enough to allow the compressor/limiter to recover almost completely between phrases. In sound reinforcement applications, excessive compression can cause feedback when the gain reduction recovers to its steady-state value.

Speech Applications. Speech, as opposed to singing, can tolerate much higher amounts of compression or limiting. Here, compression or limiting is actually used more as an effect than as a cure. Thus the actual amount of compression or limiting used will depend upon artistic license. Start at about 5:1 and work upwards. Use as much gain reduction as is necessary to achieve your goals. Here, the manual attack and release control may work better than the AUTO selection. Use the fastest attack time that allows the desired amount of limiting, without the loss of attack at the onset of limiting being audible (a sucking sound). Pick a release time that allows almost complete recovery (one to three dB gain reduction) between words.

Tightening Up a Bass Guitar. Another common use for a compressor/limiter is on a bass guitar. Some of the reasons usually given are:

1. Controlling transients produced by slapping the strings.
2. Improving a player's consistency in playing at a constant volume. This helps the bass and kick drum to work together to provide a solid rhythmic foundation.
3. Modifying the tonal characteristics by shaping the attack and decay characteristics.

The first situation calls for a fast attack, fast decay and a high compression ratio. The attack control should be set so that enough of the transient gets through for punch, but not enough for overload. The decay should be set so that the limiter recovers completely between notes. Pick a ratio setting of 10:1 or higher.

The second situation calls for a less radical approach.

Here, because we are interested in controlling the average level, a moderate compression ratio of maybe 4:1 to 8:1 with 3 to 6 dB of gain reduction on the highest peaks is called for. A compressor with automatic or program-controlled release may work well here. If your compressor has this feature, try it. If not, use a fairly long attack time and a moderate release time. The attack control will affect the initial attack of a note (percussiveness), while the release time governs the sustain of the note. If the release time is too quick, distortion will result as the output of the CVP follows the actual waveshape rather than the envelope of the signal.

The third situation "misuses" the compressor. Here, almost any settings will do. You are using the compressor as a creative tool, so there are no hard or fast rules.

As the release time of a compressor approaches the period of the input signal, the output of the CVP tends to try to follow the actual signal waveform rather than its envelope. This effect is more pronounced at low frequencies than at high frequencies. The audible consequence of a too-short release time is harmonic distortion.

If you are dealing with an instrument that doesn't have much "tone" (harmonic content) in its output, you can use a compressor (with the release time deliberately misadjusted) to introduce some controlled distortion. Now you can equalize the added harmonics to (hopefully) restore some semblance of tonal color to the instrument.

You can use the same trick to increase the harmonic content (an enhancement rather than a remedy). Pre-equalize the instrument's output by rolling off the low frequencies and boosting the mid and high frequencies. Set the preamp gain for as much gain as possible without clipping and set the input or threshold control for maximum gain reduction. Use a very high ratio setting and experiment with the attack and release settings. These settings should give an incredible amount of sustain. Lower ratio settings will decrease the amount of sustain and allow more dynamics.

Squashing a Piano. Pianos are capable of incredible dynamic range coupled with a high peak-to-average ratio. Sometimes this excess range gets in the way and the only solution is to compress it. Attack and release time settings are crucial to preserving the piano's characteristic sound. RMS responding compressors tend to work better here because the CVP isn't fooled by the piano's high peak-to-average ratio.

When using a compressor/filter to control the dynamic range of a voice or piano, the settings for the attack and release controls must be carefully chosen. If a very fast attack time is chosen, the high peak-to-average ratio of the piano's sound may cause excessive amounts of gain reduction. This could make the piano sound "squashed." The piano's characteristic "edge" or brightness would be destroyed because the compressor/limiter is compressing on the peaks that give the piano its characteristic sound. A slower attack time will let some of these transients pass somewhat unaffected while still allowing the compressor to maintain control over the total dynamic range of the instrument. On the other hand, a squashed piano sound may be just what the doctor ordered. This example holds true for many other sound sources.

Stereo Applications. When compressing or limiting a stereo signal, it is customary to link the CVPs of each compressor or limiter together. This avoids image shifts caused by unequal amounts of gain reduction in the stereo channels.

Most high-quality compressor/limiters have stereo interconnect jacks. A few require an interconnection accessory other than a patch cord. Usually all that is required is to patch the two jacks together.

A Few Experiments

With all this under your belt, here are a few experiments to try in your spare time.

Connect the compressor/limiter to your console channel insert jacks. Plug a bass guitar in via a direct box or a high impedance microphone input. Experiment with various settings of the compressor/limiter. Try setting the ratio control at maximum, the threshold control to -40 dBm, and experimenting with the attack and release controls and various playing styles. Now repeat the experiment with the ratio control set at a less extreme setting.

If you have a bass guitar amplifier that has preamp output and power amp input jacks, repeat the above experiment.

Try the bass guitar experiment with a lead guitar. Extremely high compression ratios, coupled with lots of gain reduction, should increase the amount of sustain markedly.

To de-ess a vocal, patch an equalizer into the side chain insert jacks. Set the equalizer for an overall high-frequency boost; six to ten dB should be sufficient. If you have a way of listening to the output of the equalizer, what you want to do is to emphasize the sibilance frequencies in the voice. Trim the amount of equalization for the desired amount of de-essing; leave the equalizer patched in as above, and repeat the bass and lead guitar experiments. Experiment with various settings on the equalizer.

Patch the compressor/limiter into the output of your mixer. Use the AUTO setting of the attack/release switch. Mix one of your favorite tapes, setting the compressor/limiter for three to six dB of average gain reduction. See what happens to the mix when you push one particular source way up in the mix. If you do this to something that is destined for a broadcast station, you can pre-compress the signal in such a way that the station's compressor can't alter the balance of your mix.

Repeat the last experiment with an announcer over a music background. Try the MANUAL setting of the attack/release switch. Experiment with the setting of the RELEASE control.

Conclusion

A compressor or limiter is a useful and versatile addition to any audio engineer's toolbox. Properly used, they can literally pull the bunny out of the hat. Used with an equalizer patched into the side chain circuitry, de-essing or level sensitive spectral shaping can be done. As a creative tool, a compressor or limiter can introduce deliberate harmonic distortion or modify the envelope of a signal in unusual and musically useful ways. If you have one already in your toolbox, try the experiments given above and see if you don't add a new trick or two to your own arsenal.

So You Wanna Be a Rock'n'Roll Star

Part I: Producing and Marketing a Demo

denny anderson

Everyone knows the scenario: the hard-working band finishes its set and takes a break, dispiritedly stepping off the seedy beer-soaked bandstand to join the six bored drunks that constitute the audience. Suddenly, from out of nowhere, a well-dressed big-label talent scout appears; he tells the startled musicians he loves their sound and whips out a million-dollar recording contract for them to sign on the spot. Discovered at last! A star is born! No time to lose, boys—the limousine is waiting outside.

I suppose everyone also knows that such fairy-tale scenarios are just that—fairy tales; sheer fantasy, about as realistic as the average Disney cartoon. Yet new artists *do* get recording contracts all the time. How do they do it? Is it just a matter of luck? Being in the right place at the right time? Politics? Voodoo? Some quirk of nature or fate? Although scoring a big-league recording contract may seem like a mysterious hat trick to some, it is really a matter of hard work, talent—and knowing how to produce and market an effective demo tape.

Going after a recording contract with a major record company is serious business. The odds against success are astronomical. With thousands upon thousands of aspiring hopefuls all trying to pick the same plum, the competition is...well, *intense*, to say the least. If you're seriously thinking about going for it, you'd better know what you're doing. You can't afford to make too many mistakes. The more you can learn about the way the process works, the better your chances will be.

This series of articles is intended to help you learn the ropes, so to speak. In this first installment we'll look at how the game is played; we'll learn a few of the fundamentals and find out what makes a demo effective. As the series progresses, we'll get right into the nuts and bolts—how to work with inexperienced musicians in the studio environment; how to produce a competitive demo tape with—if you must—basic consumer-audio gear; how to go about getting radio airplay for your tape; and finally, how to most effectively present your demo tape to the record companies. I can't give you a simple formula that will guarantee your success (if I could do that I'd be in the Bahamas right now, counting Kruggerrands), but I can give you some useful information and suggestions that will enable you to avoid costly mistakes and improve your odds for success. The rest is up to you. It *can* be done.

Getting Discovered

These days the artist who sits around waiting to be "discovered" is in for a long wait. It just doesn't work like that. Record companies are always on the lookout for new talent, but not with the roving talent scouts of our opening fairy-tale scenario. With so many artists trying to get recording contracts, the companies don't have to roam far and wide searching clubs and garages for undiscovered talent; the talent comes to them—in droves. They won't come looking for you; if you want them to hear you, you'll have to make the first move. You'll have to actively sell yourself to the record company, and to do that you'll need a demo tape.

The Demo Process

A demo tape is a sampling of your recorded material, intended to show the record company what you can do. It's like an audition recording. You're trying to convince a company to gamble a lot of money on your ability to make recordings for them, and your demo tape is your sample product. You should prepare it very carefully. It helps to understand how the record company operates before you begin putting your tape together.

All the major labels and many independents maintain a full-time staff of A&R (artists and repertoire) people. A large company will have a whole A&R department; a smaller label may have only one person who performs this function. These are the "talent scouts" in charge of finding new acts for the label. It's part of their job to listen to all the demos that come in, screening them for promising material. Yes, they actually do listen to every tape; if you submit a demo to a record company, you can be sure that someone will hear it. This sounds encouraging, and it is—but only if you know what you're doing. The typical A&R

person in a major record company is generally young, extremely sharp, and very much wired into the current commercial market. A good A&R person can generally tell immediately whether a song has commercial potential or not. That's not to say they're infallible, of course, but they usually hit more often than they miss, or they wouldn't be there. The A&R field is intensely competitive and volatile. Things move fast; decisions are made quickly. Piles of demo tapes come in every day. They don't waste time savoring each note and nuance; they pop 'em on and kick 'em out ruthlessly. You can be sure they will hear your tape, but if you haven't grabbed their attention with something they like in the first twenty or thirty seconds, you may be in trouble. They may not wait around to hear the rest.

This may sound pretty cutthroat, but remember, you're trying out for the big leagues. They play hardball, and they don't mess around. They know what they're looking for, and it's up to you to show it to them. You have to have your act together and make your demo tape as effective as possible.

What Makes a Demo Effective?

The main thing the A&R people are looking for is strong commercial material—songs that have the potential to become hits. After all, record companies are in business to sell records. This may seem obvious, but there are an amazing number of aspiring artists and would-be producers out there who fail to fully grasp the significance of this fact. It is a common misconception that the recording industry exists for the sole purpose of providing entertainment; that record companies are in the business of supplying people with musical enjoyment. They do these things, of course, but only incidentally. To look at it in bluntly cynical terms, the recording industry exists to *make money;* record companies are in the business of selling vinyl for profit. Now before I get a lot of nasty letters, let me freely admit that this is a grossly unfair oversimplification. Nevertheless, it's a hopeful perspective to keep in mind as you're producing your demo. The A&R person who listens to your tape will be looking at it from a business perspective. If he or she feels that your material is something a lot of people will want to buy, you stand a good chance of being offered a contract. If, on the other hand, your tape lacks this commercial appeal, it will almost certainly be rejected —no matter how slickly you've produced it or how musically sophisticated it may be.

For your demo to be successful, your material must have strong commercial appeal.

Okay, so how can you insure that your material will have commercial appeal? No one has the magic formula for guaranteeing that a record will be a hit, but there are steps you can take to increase your chances for success.

Doing Your Homework

Follow the record charts. The more you can familiarize yourself with what's selling successfully in the current market, the better you'll be able to judge the commercial strength of your own material. Keep abreast of current trends. You don't have to rush to embrace every little fad and fancy that comes along, but you should make it a habit to know what's going on in the commercial end of the music business. *Billboard* and *Cashbox* magazines are good sources for this kind of information.

Listen to the radio. Don't just tune in your favorite station and space out—*pay attention* to what's being played on all the available stations in your area. Familiarize yourself with the major radio programming formats: Hot 100, AOR (album-oriented rock), adult contemporary, urban/soul, country, and Top 40. Does your material fit readily into one of these major formats? If your music is off the beaten paths of the programming formats, radio stations won't play your material. No airplay means no sales. A&R people are extremely aware of current trends in radio programming; they won't offer you a contract if they don't think your material can get widespread airplay.

Study the current hits; pay particular attention to the way they're constructed. Listen for the hooks (the catchy repetitive musical phrases and lyrics that make the song popular). Learn to recognize a good hook when you hear one. A hot-selling record usually has a short, strong introduction that instantly grabs your attention and sets you up for the first lyric, which in turn leads you right into the hook. Experiment with your own material. Be sure your songs have very definite hooks, and don't waste any time getting to them. Remember if you haven't hooked them in the first twenty seconds, they may never hear the rest of the song. Keep your material short and tightly arranged. Avoid rambling introductions and extended solos; get in, say what you have to say, and get right back out. Keep hitting those hooks!

In your listening research, keep abreast of new sounds and production trends. Take time to analyze the production in songs you really like. For example, how does the snare drum sound? Is it placed up front in the mix, or further back? What are the kick drum and bass doing? How do they work with the guitar parts? How are strings and background vocals used to build texture? How does the song build and develop, and how does the production complement and enhance the musical development? In a demo tape, content is more important than fancy production. The sound quality of your tape should be as good as you can make it, but don't let hair-splitting technical details cause you to lose your perspective. Remember, the record company is looking at your material first and foremost. Your production should help put the song across, not overshadow it.

As you analyze hit records over a period of time you'll develop a greater awareness of the techniques successful artists and producers use to create that all-important commercial appeal in their recordings. The more effectively you can employ these techniques in your own work, the more effective your demo tapes will be.

To summarize, then: Producing a demo tape is the necessary first step toward getting a recording contract. An effective demo will contain material which is fresh and original, yet fits easily into existing radio programming formats; it demonstrates an awareness of the current market; and it skillfully blends songwriting and arrangement techniques with appropriate production to create strong commercial appeal.

In Part II of this series we'll help you lay out your strategy and prepare for your demo session. Until then, keep your ears to the grindstone and do your homework. Your limousine is waiting!

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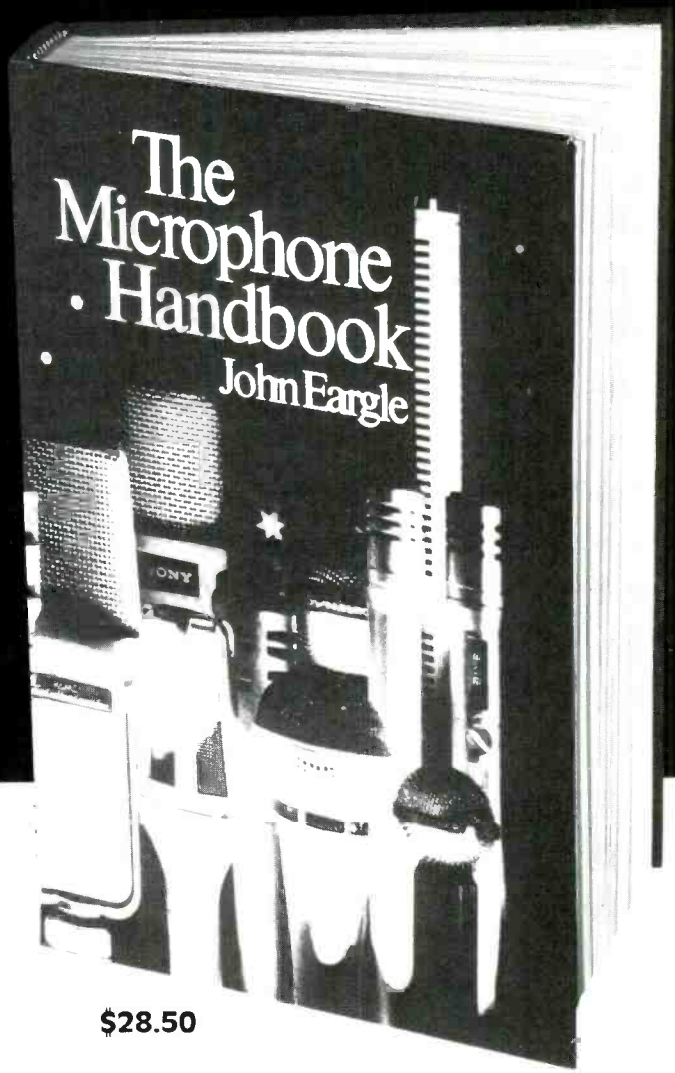

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Wynton MARSALIS

gene kalbacher

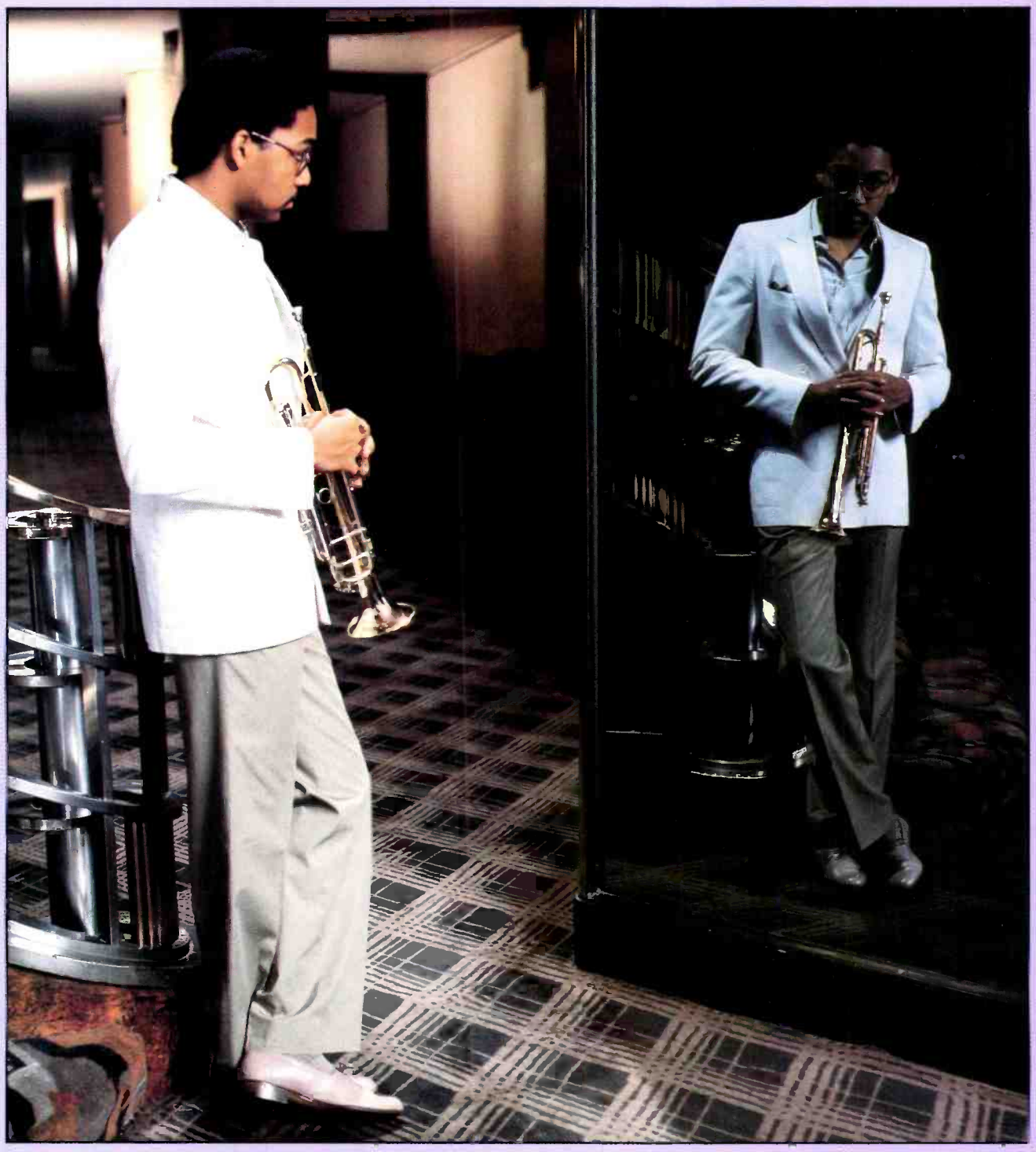
Success has yet to spoil Wynton Marsalis. He seems to thrive on success, perhaps even to expect it, but he isn't beholden to it. That the 21-year-old trumpeter has attained so much of it so soon appears remarkable, unless one has followed his meteoric career since he joined Art Blakey and the Jazz Messengers in 1979. Those who heard the New Orleans native's accomplished playing in Blakey's band—a musical dynasty that has incubated such trumpet marvels as Donald Byrd, Lee Morgan, Freddie Hubbard and Woody Shaw—knew he was something special.

By 1981, when Columbia released his first album as a leader, *Wynton Marsalis*, this nattily attired and bespectacled young man was astounding audiences the world over. His debut album, produced by Herbie Hancock and featuring the keyboardist and his former companions in Miles Davis's mid-'60s quintet (drummer Tony Williams and bassist Ron Carter), sold 100,000 units, an incredible figure in an acoustic genre that considers 10,000 a major success. By 1982, Marsalis's stronghold had evolved into outright supremacy. The results of the 1982 *Down Beat* readers poll attest to his dominance: Marsalis earned honors for album of the year, best trumpeter, and jazz musician of the year—easily outdistancing his nearest competitor, Miles Davis, in every category.

Wynton, son of the noted pianist-educator Ellis Marsalis and brother of the saxophonist Branford (whose debut LP for Columbia is expected soon), has been even more active this year. Besides touring and recording with the Herbie Hancock Quartet, a veritable '80s VSOP band, Marsalis further distinguished himself by being the first American artist to issue jazz and classical albums simultaneously. *Think of One*, the full-scale recording debut of his working band (Branford Marsalis, saxophones; Kenny Kirkland, piano; Jeff Watts, drums, and Phil Bowler, bass), was recorded at Mediasound in New York City and produced by the trumpeter. *Haydn/Hummel/L. Mozart: Trumpet Concertos*, showcasing Wynton with the National Philharmonic Orchestra under the direction of Raymond Leppard, was digitally recorded at EMI's Abbey Road Studios in London and produced by Thomas Mowrey.

The trumpeter, who received his first horn at the age of 6 from none other than Al Hirt, has proved convincingly that the field of classical music and jazz are not mutually exclusive. But his love for classical music is hardly a publicity seeking flirtation. At 14 he played the "Haydn Trumpet Concerto"; two years later he tackled the "Brandenburg Concerto No. 2 in F Major." Upon arriving in New York, he attended Juilliard and performed as a pit musician on Broadway in *Sweeney Todd*.

Being first has become second nature to Wynton Marsalis. A graceful improviser with a voluminous vocabulary, he is an engaging, brash interviewee; he is outspoken—some would say opinionated—but not conceited. In the following interview, conducted at CBS Records in Manhattan, Marsalis speaks in a clear, forthright, articulate manner.



Modern Recording & Music: On *Think of One*, the time and rhythm, particularly behind the soloists, is so elastic, so open. From a purely production standpoint, did this pose any problems in terms of mic'ing or placement of musicians?

Wynton Marsalis: Yes. We made a major mistake on this record in that we didn't record it in a big enough room. The trumpet sound doesn't have a chance to travel. We had a lot of problems with the drums because Jeff [Watts] plays so loudly. But I like that. We also had problems in the dynamics; they change. We're playing really soft in some spots and really loud in others. Sometimes we didn't get the sound like we really wanted it on the saxophone solos.

[Mediasound] is a nice room but it's just too small for what we're trying to do. We need a really high ceiling. But I like the studio.

MR&M: Which studios are more appropriate for your concept?

WM: CBS' Studio B was good, but they sold that. And I've heard from all the musicians that CBS' old studio was really good. Rudy Van Gelder's studio is a really big room. The sound can travel a lot and you get a fuller sound. You can back away from the microphone, or maybe use two mics.

MR&M: How were you able to circumvent the studio's lack of a high ceiling?

WM: I backed away farther from the microphone. On my first album, I made the mistake of playing too close to the microphone. Everybody kept saying, "Back away from the microphone," but I would still come closer. And, we used two mics for the trumpet. We always use two mics for the saxophone. That compensated for it to some degree.

MR&M: *Think of One* marks your debut as a producer. Did you want to produce your own albums all along?

WM: No. It doesn't make any difference to me. I'd rather find a producer, personally, but it's really difficult to find a producer for the type of stuff I'm trying to do. We had this '70s drought where trumpet sounds weren't really recorded, so the trumpet sounds I hear on records now aren't good.

MR&M: To what do you attribute this? Might the engineers and producers be rusty?

WM: That's what I think. I'm not saying that toward Tim [Geelan, chief engineer on *Think of One*], because I love Tim. It's just the whole scene in general of the '70s, the fusion era. No trumpets were really being recorded.

MR&M: And the ones that were recorded were often modified with effects.

WM: The trumpets were used mainly for rhythmic effects, not really for the beauty of the sound quality. The records I did with Art Blakey—I hate the sound on those records. *Hate* the sound. Not good. And that takes away from a record. If you have a really good sound on a record, that makes the music sound a lot better. A record is supposed to enhance the music, not take away from it.

MR&M: Black American classical music is....

WM: Don't call it that! I hate that name.

MR&M: What should I call it?

WM: Call it *jazz*.

MR&M: Wait a minute! Just a few months ago in *Down Beat*, you said, "...jazz—I hate that word."

WM: I hate it, too, but it's better. Just call it *American music*. That's the best term. When you say *black American classical music*, that makes it sound like you're trying to define it in terms of classical music. It will always lose that battle. When you say it's black American music, that means it's solely restricted to black Americans.

MR&M: Which, of course, it's not. My question is this: How did the setup of musicians in the studio for *Think of One* spark improvisation, a group dynamic and, at the same time, the trumpet sound you desire?

WM: We set up so we can all see each other when we're playing.

MR&M: Since the album was recorded live in the studio, overdubbing seems to have been minimal.

WM: We overdubbed one time. It's not an overdub of a part; it's an overdub of a solo behind a melody on "Fuchsia," a Kenny Kirkland tune (hums melody). My brother and I are playing the melody in the background, and the trumpet [solo] is overdubbed. I couldn't do both of those things at the same time.

MR&M: You mentioned a moment ago, Wynton, that you didn't set out to produce *Think of One*. What made you accept the responsibility?

WM: Because I wanted to see it done the right way, the best I could make it. I don't mind getting somebody who knows more about it than me—which, I'm sure, there are a lot of. But it's hard to find someone. So to all the producers, step forward.

MR&M: How much did the experience of working with producers Herbie Hancock [on *Wynton Marsalis*] and *Stanley Crouch* [on *Fathers & Sons*] assist you in producing yourself?

WM: (Raises eyebrows and rolls his eyeballs.)

MR&M: That's a nonverbal answer (laughs).

WM: They were both good, but I do it a different way.

MR&M: As producers, did they use certain approaches that you felt enhanced certain tunes or your trumpet sound in general?

WM: We won't deal with that (chuckles).

MR&M: Did you find that producing was easier or harder than you had anticipated?

WM: It was about what I had expected. I didn't know enough. I wish I had known more stuff. I'm learning now, you can tell. The bass drum is too loud on the record.

I like the pure—like Duke Ellington, Louis Armstrong, Trane, Miles in the '60s and '50s, the real hard grit [Thelonious], Monk. That's what I like a lot.

MR&M: Did you do much advance preparation with Tim Geelan, the engineer, before recording *Think of One.*

WM: A little bit. He knew what to do. We did the first one [*Wynton Marsalis*] together.

MR&M: How many recording sessions were conducted?

WM: Four or five. A week of recording and a week of mixing.

MR&M: How active a role did you take in the mixing stage?

WM: Very active. I'd done some mixing before, but never editing.

MR&M: Having worked in a variety of recording studios—Mediasound, CBS Studios, CBS-Sony Studios in Japan and EMI Abbey Road, where you cut the trumpet concerto record—what characteristics do you look for in a studio? If you had the inclination and wherewithal to build your own studio, Wynton, what type would you construct?

WM: You need a large room so the sound can move around a lot. Live but not *really* live. I can't say it in technical terms like echo of how many seconds. A room that's not dead—you know, it has some echo but not an echoey room. Not a bright room; toward the dark side. That's the most important thing, but I haven't found that room for a jazz record yet.

MR&M: By *dark* do you mean a room that brings out the lyrical qualities of the music?

WM: It makes the music sound warmer. So many people get the trumpet to sound really bright and nasal-sounding. I can't stand that, I hate that. That makes me sick.

MR&M: Has that bright trumpet sound become a stereotype?

WM: It has, and it's terrible. Plus, then you can't hear the attacks the guys are making. It takes away from the music.

MR&M: You mentioned when we first spoke, about two years ago, that you consider yourself a jazz musician who can play classical, not a classical musician who can play jazz. Are you pleased with the way the trumpet concerto record turned out?

WM: It's all right. I hear flaws in it. But when you listen to your own things, you always hear flaws. I really liked working with Tom [Mowrey], the producer.

MR&M: You recorded with the National Philharmonic Orchestra, under the direction of Raymond Leppard. How many sessions were needed?

WM: Three sessions, one concerto per session.

MR&M: How were the microphones arranged?

WM: We just had one microphone that was far away, suspended. In jazz, the mic is closer. In classical music, the microphone is farther away, so you get more room sound.

MR&M: Did the distant mic force you to make any alterations in your customary methods?

WM: It didn't make any difference.

MR&M: You didn't have to alter your attack?

WM: I had to alter my attack automatically, but not due to the microphone placement. It was due to the different nature of the music. With classical music, you're trying for a purer type thing. In jazz, you can give any range of attacks. You try to get as many different attacks as you can get. In classical music, you're trying to maintain a certain consistency of attack.

MR&M: Now that the classical album has been released, what do you believe you've accomplished?

WM: I had fun doing it (pause), after I got over being nervous. I'm going to keep trying to play it for a couple more years.

MR&M: Live performances?

WM: Yes (raises eyebrows).

MR&M: Trumpet concertos?

WM: Yeah (rolling eyeballs).

MR&M: In this country?

WM: Yeah (softly).

MR&M: I'm getting this one word at a time. I feel like a prosecutor (laughs). I'm told that you began playing classical music almost on a dare, that teachers and friends told you, as a youngster, that blacks couldn't do it.

WM: Kinda. That's gone now, man. I don't care. When you're young, you react in different ways. The older you get, the more you learn.

MR&M: Did Thomas Mowrey do anything as producer of *Haydn/Hummel/L. Mozart: Trumpet Concertos* that assisted you in producing your jazz quintet album?

WM: He came in and helped us on [*Think of One*]. He was there in the studio a couple days.

MR&M: What input did he have?

WM: A lot of different technical things....He tried to help us get as natural a sound as we could. But we needed a bigger room, definitely.

MR&M: For live gigs, do you travel with a sound man?

WM: No. We depend on the sound of the band to carry whatever [room acoustics] it is.

MR&M: You credit Louis Armstrong, Dizzy Gillespie, Miles Davis, Lee Morgan, Clifford Brown and Woody Shaw as being important trumpet influences. What about classical influences?

WM: Maurice Andre, Thomas Stevens, Gerard Schwartz, Adolph Versa.

MR&M: Have you had any personal contact with these musicians?

WM: Yeah. I've met all of them.

MR&M: Are you finding, Wynton, that orchestras these days are becoming more open to different kinds of nonclassical music?

WM: I hope so, but I don't know. People have different prejudices, based on what they can and cannot do.

MR&M: Are orchestras becoming less of a white man's domain?

WM: It is. Black people really don't want to play classical music, by and large. We don't relate to that. It's another sound.

WM: Why is it, then, that you play it?

MR&M: I like it. If I hadn't started playing it [as a youngster] or listening to Maurice Andre, I wouldn't be interested in hearing it. But I like to listen to jazz more, honestly. I love classical music, too, and respect it, but jazz is more today, man, 20th century music. It's more about *today.* Which is not to cut classical music down. It's great music...and I respect the musicians who play it. But it's yesteryear and it's European; jazz is today and it's American. And I'm American.

MR&M: Yet a lot of jazz, and I think a lot of good jazz, draws from both the European classical tradition and from the American jazz tradition.

WM: I don't think so.

MR&M: You don't think the juxtaposition has produced some interesting hybrids?

WM: No. Come up with some examples? Tell me what it is.

MR&M: A lot of ECM records come to mind.

WM: I don't like that. Not swinging. It just sounds like what it is. I like the pure—like Duke Ellington, Louis Armstrong, Trane, Miles in the '60s and '50s, the real hard grit [Thelonious], Monk. That's what I like *a lot.* I can like some ECM stuff and say that it's nice. That's light kind of stuff. I'm talkin' about really heavy.

European music is great music.

Jazz automatically fuses some portions of European music, because jazz is the result of a fusion of cultures, which is the only way that a real fusion can occur. A fusion of cultures. The people who make the [jazz] music see it from the experience of black people who were slaves. And since they were slaves, they combine what they brought over here, which is more an African-type thing—more of that oral relationship and more spontaneous—with European-type music, which is more literal. And it was fused in America, which is the place where everything new was happening....The element of harmony is European and the bass line is a European concept, but the music [jazz] is American music. The Europeans have elements in their music that come from somewhere else; it's not all European. The music becomes assimilated from a cultural standpoint because music is a reflection of culture and life. So it's not a matter of somebody sitting down and saying, "I will use these influences and my music will come out sounding this way." It's a natural combination.

MR&M: Is understanding jazz essential to appreciating it?

WM: Not essential at all.

MR&M: You've done an about-face on this question since we spoke last. Two years ago, you stated, "...It's not enough for people to say, 'I just *feel* jazz.' That's crap! You have to know what it is."

WM: I know. You change your opinion. People shouldn't understand music—understanding on a *technical* level. I might've meant [two years ago] that they should understand the goal of the music. It's impossible for a person who has other things on his mind to figure out what we're doing. There's no way. Forms and themes—there's just too much [for the average listener] to understand. You should sit down and enjoy the *sound* and the *feeling* that the sound emits. Music has a *texture* to it, and people should relate to the texture. The reason that pop music is so popular is because it has a pleasing texture. It's very easy to understand, and I think that's good. Classical music and jazz should learn from that—but not by trying to sound like pop music. They should try to achieve that texture while keeping the music pure.

MR&M: On the subject of texture in pop music, let me ask you this. Much of today's so-called urban contemporary music sounds like it's made more by machines than by musicians. Synthesizers and drum machines are replacing pure, acoustic instruments. What are your thoughts about this scene?

Now all the black musicians are trying to make their music sound *white* so they can make money: Earth, Wind & Fire, Michael Jackson, Lionel Richie, Prince. All those people—their music sounds white.

WM: That's cool, because that's the contemporary thing—computers. That catches the 20th century-type feel. Everybody has a computer. We're fascinated with machines now. That's why the music is like that.

MR&M: I'm rather surprised to hear that answer from you, an acoustic jazz and classical musician. You don't feel any unease about much of this numbing, repetitious, mindless, "A-E-I-O-U, I Love You" music?

WM: That's what the music is for, though. You go into a disco, get high, dance, have a good time, score, split. That's what that's about. And that's what the deal is. It's not about you sittin' down and listenin', saying "This is that" and trying to explain what it is. It's cool.

MR&M: You're not bothered that it puts musicians out of work?

WM: That's life. So do computers. Progress, man...I'm not bothered by it at all. A synthesizer will never put a musician out of work; a synthesizer can't do what people can do, period.

MR&M: Granted, a synthesizer can't do what a musician does, but some producers will settle for a cheap, synthetic sound over the sound from a musician.

WM: But if that's what people [listeners] want, that's good then.... It took 100 years for them to discover Bach.

MR&M: On the surface, I do see one positive development in urban contemporary music: There seems to be more cross-pollination than ever between black and white dance music. If this music reduces racial discrimination, then its cultural effect is worthwhile.

WM: The thing I *don't* like about it is that what is lost is the *blackness* of the music. What you have to understand is that white music automatically started out imitating black music....The business people separated the music so they could make money. They knew that the mass number of white people, back then, would not support a black person, period. Whereas a mass number of black people will support a white person, like Elvis [Presley] or something. So they coined *black* and *white*. So *white* became the white cats who could imitate the black people as best they could. Over the years of imitating us, they're getting better and better at it. The thing that we [blacks] had—we're losing that by trying to appeal to the white market, now that a mass number of white people will listen to black people.

Now all the black musicians are trying to make their music sound *white* so they can make money: Earth, Wind & Fire, Michael Jackson, Lionel Richie, Prince. All those people—their music sounds white. I'm not saying that's bad or good. I'm saying that a certain element of the music is being lost, definitely. Stuff like Marvin Gaye and Donnie Hathaway [with] that kind of edge on it. That's definitely not present in that music [now]. And a lot of black people are noticing that. Another thing, it's very hard for black musicians to get their shit played on white radio; but white musicians can get their stuff played on black radio. Hall and Oates [for example].

MR&M: Black musicians are up in arms that MTV programs very few black videos.

WM: In TV, people are trying to present a white image of black people. Also, a clown image....

MR&M: Did you record any tunes for *Think of One* that don't appear on the album?

WM: No. That was it.

MR&M: Were they essentially first takes?

WM: Second, third. Sometimes it takes us a long time to get the concept together on a tune.

MR&M: Were the concepts developed in the studio?

WM: Sometimes. We had played some of these tunes before in gigs, though. We do things on this album, conceptually, that I haven't heard done. But (softly) nobody noticed that.

MR&M: The rhythm section-soloist relationship?

WM: We're not playing all chord changes, man. The chords will be changing all through the solo. And it's not on a mode, either.

MR&M: It's not scalar?

WM: Not on scales, no. No scales or modes. Just the chords are changing as you play. (Laughs) I'm tellin' ya. On "Knozz-Moe-King," there's no mode on that.

MR&M: One of the most distinctive things about *Think of One* is the *band* sound.

WM: Good cats, man. My brother—great ears. Kenny Kirkland—great ears. Jeffrey Watts—great ears. They play together. It's more thematic. We play more as a *group*, everybody feeding off each other's theme. People don't notice that, man.

MR&M: With all of our discussion about you, I've given short shrift to your working quintet.

WM: They [critics] never talk about them cats. In reviews they'll try to use the fact that my brother is good to cut me down.... You can use the whole band to cut me down if you want. All the cats are good. Don't snipe on everybody in the band.

MR&M: I haven't heard a rhythm section-soloist dynamic so strong since some of Miles Davis's mid-'50s music with John Coltrane, Red Garland, Paul Chambers and Philly Joe Jones.

WM: What you have to understand is that [as soloists] Davis and Coltrane were playing *on top* of the rhythm section. We don't play on top of the rhythm section—we play *with* the rhythm section. Not saying that we're doing it as good as what they did. We're playing *inside* the rhythm section. Check it out. Get the records out and listen to them.

MR&M: The approach, then, is different, but to my ears the effect is similar.

WM: It's just the same music. The effect of what they're doing is the same as the effect of what Monk's band did, the same as the effect of what [Charles] Mingus's band did. Just like the effect of what Beethoven did is the same as what Haydn did. It's different but it's the same, because it's the same music. It's going to be a certain way because it's the same type of music. But it's the subtle differences that make the differences in the music. Everything is supposed to sound like something else. If it don't sound like something else, it ain't shit. Everything sounds like something else because all of life is cause-and-effect and change. Everything. Everything comes from something. Everything. And whoever can figure something that *didn't* come from something [else], they know something we all need to know.

MR&M: Were your originals on *Think of One* written specifically with that album in mind?

WM: They'd been written before.... I just write when I feel like writing. Different people have different approaches. Sometimes Wayne [Shorter, the saxophonist] will take solos that he played on [previous] records and make a tune out of them. On one Miles record, *Nefertiti*, Wayne plays a solo on "Madness," on the second side of the album (hums solo). On *Water Babies* there's a tune that goes (hums new melody drawn from solo).

MR&M: I wonder whether he purposely went back and listened to the record or whether the solo reemerged from his subconscious in a transmuted form.

WM: Oh, he definitely did [go back and hear the album]. I think he did that. He does that a lot of times.

MR&M: I'm reminded of a tune on Miles's double-live *We Want Miles*. He does a sing-songy, nursery rhyme-like tune called "Jean Pierre" that goes like *"rain, rain, go away...."*

WM: Yeah, but that's "Filles de Kilimanjaro."

MR&M: I could swear I also heard that phrase during a Miles solo on a Columbia disc recorded live at Philharmonic Hall in '64.

WM: Yeah, but they did a good arrangement of that on "Filles de Kilimanjaro" (scats excitedly). You have to know the history, you have to listen to everything. We listen to Miles, man. We listen to Trane. And we listen with the same reverence.

MR&M: Wynton, although your recording career is still so embryonic, do you, like Wayne Shorter, go back and listen to the albums you've made?

WM: I know what's on there. No, I know what mistakes I made. I listen to all kinds of other people's stuff. That's the only way you can learn.

MR&M: What are some of the principal mistakes you made on the *Wynton Marsalis* album, for instance?

WM: I wasn't playing as good as I could play there. I didn't really know what I was doing. I didn't have a group concept. Just the typical mistakes you make when you're young. I didn't like the stuff that I personally played on it. Some good moments in there. The cats in the band sound real good. We didn't get a good enough sound on the record.

MR&M: Have you become progressively more pleased with the sound on each of your subsequent records?

WM: Yes, I think so.

MR&M: Do you see a logical progression from one record to the next, or is that anathema to what jazz is all about?

WM: You have to get better. That's why I do it—to get better. I'm not putting out records to make hits. I'm putting out records to make a statement.

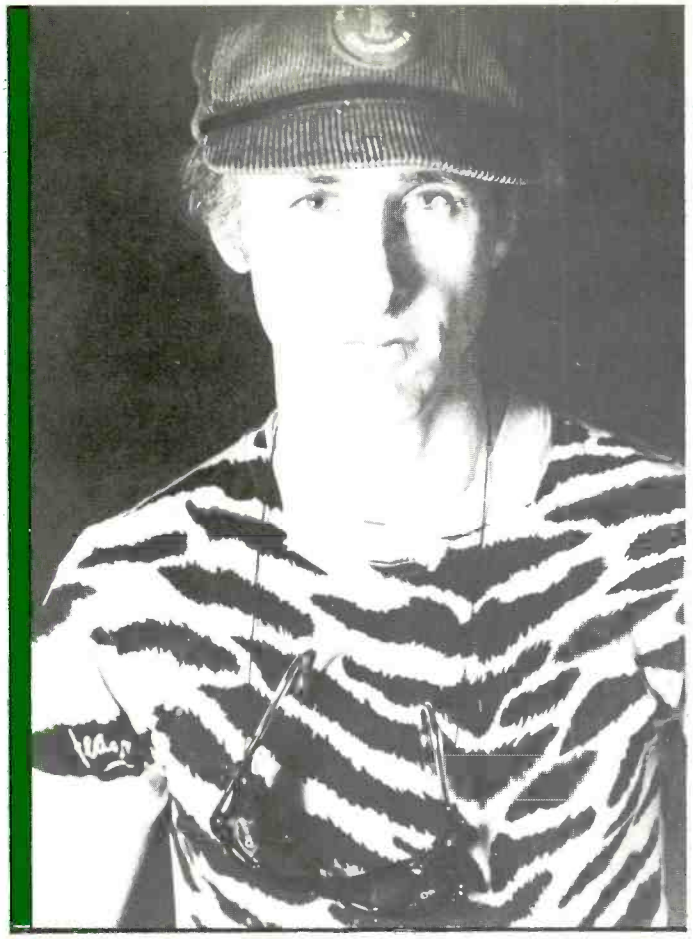
MR&M: How are you pursuing your goal to *get better*?

WM: Practice, listening, and work.

Wynton Marsalis Discography

As a Leader

Haydn/Hummel/L. Mozart: Trumpet Concertos (Columbia)
Think Of One (Columbia)
Fathers & Sons (Columbia)
Wynton Marsalis (Columbia)



jeff tamarkin

Any listener familiar with rock radio is well aware of the fact that the Doors are at least as popular today as they were during their prime in the late 1960s. Doors records, notably such hits as "Light My Fire," "Love Me Two Times," "Touch Me," and "Love Her Madly," receive frequent radio airplay, acquainting a young generation—some of them in diapers when the group was recording these hits—with the music of one of the most legendary '60s bands.

This resurgence is due in large part to the myth and legend of Jim Morrison, the late lead singer of the Doors who wrote poetic lyrics of doom and decadence and forged an image as one of rock's most rebellious outlaws. Morrison's exploits—getting arrested for lewd behavior (allegedly exposing himself on concert stages), drunkenness, etc.—are almost as big a part of the Doors story as the music.

But there was more to the Doors than Jim Morrison and his reputation as the daredevil sex symbol. There was also a great body of timeless music. And there were other members of the Doors, lest we forget: Ray Manzarek, keyboardist; John Densmore, drummer; and Robby (formerly Robbie) Krieger, guitarist. What most people don't know to this day about the Doors is that Jim Morrison wasn't responsible for all of the lyrics that have helped earn the group's songs their place in the rock lexicon. While Morrison might have been responsible for

ROBBY KRIEGER Sets Sail...

the dark, often frightening poetic images of such excursions as "The End," "When The Music's Over," "Riders On The Storm," "People Are Strange" and the jumpy blues boogie "Roadhouse Blues," the aforementioned hits—"Light My Fire," etc.—came from the pen of Robby Krieger, the quiet, virtuoso guitarist who provided the stinging slide guitar work and crystal clear lead on all of the Doors' recorded works.

After leaving the Doors in 1972 (following two post-Morrison LPs that went nowhere), Krieger, born in 1946, formed a band with drummer Densmore called the Butts Band, whose achievements included recording some of the first reggae covers outside of Jamaica (Krieger remains a big reggae fan today), but not much in the way of commercial acceptance. After the band's demise, Krieger took five years off to work on his guitar technique. Although many consider him one of rock's most distinctive guitarists, Krieger believed he was known primarily as a songwriter and wanted to brush up on his playing. In 1977, he released a solo album, *Robbie Krieger And Friends*, and in 1978, he emerged in an L.A. band called Red Shift, featuring a Zappa sideman, synthesist Arthur Barrow, with whom Krieger worked the L.A. club circuit.

In 1982, Krieger landed back in the spotlight in a big way. He recorded a solo album, *Versions* (Passport), using Barrow as well as former Knack drummer Bruce Gary and several other session musicians (including the other remaining ex-Doors on two cuts, one a remake of the Doors' "Crystal Ship"). The album was an all-instrumental effort designed to showcase Krieger's guitar work, and that he did with impressive results, working in a number of styles from jazz to new wave to reggae. Following the album's release, Krieger and his band took to the road—Krieger for the first time since the Doors' breakup—and when they stopped in New York to play the Peppermint Lounge, *Modern Recording & Music*'s Jeff Tamarkin spoke to the guitarist/songwriter about his album and his past work with the perennial West Coast favorite, the Doors.

Modern Recording & Music: Robby, most people know you from your work with the Doors, but you recently released a solo album, *Versions* (Passport Records). Do you find it a problem that you are so strongly associated with something from the past? Is it hard selling yourself as a contemporary artist?

Robby Krieger: Not really, because one is just an extension of the other. I've kept playing, so I don't consider myself someone out of the past.

MR&M: What about in the eyes of the audience, though? When you play live with your band, do people still call out requests for Doors' songs?

RK: Some of them do, but the majority of them appreciate the new stuff.

MR&M: You didn't record for five years before making this album. What were you doing?

RK: I was developing my guitar playing. I wanted to have it really down before I recorded anything. Also, I've been doing a lot of production for other groups, mostly young L.A. groups.

MR&M: Why did you choose to make the album all-instrumental?

RK: I wanted to establish myself as a guitar player; I've been known mostly as a songwriter.

MR&M: This isn't the first record you've made since the Doors. You also worked with a group called the Butts Band in the early '70s. How does this sound compare to what you did with them?

RK: There's really no comparison at all. This is totally my thing whereas the Butts Band was a group, with everybody writing and contributing.

MR&M: Who are some of the main personnel on the album?

RK: Arthur Barrow, who plays bass and keyboards; I've known him for some time. I have Bruce Gary from the Knack playing drums. They're in the touring band, too, and so is Berton Averre from the Knack, who plays guitar.

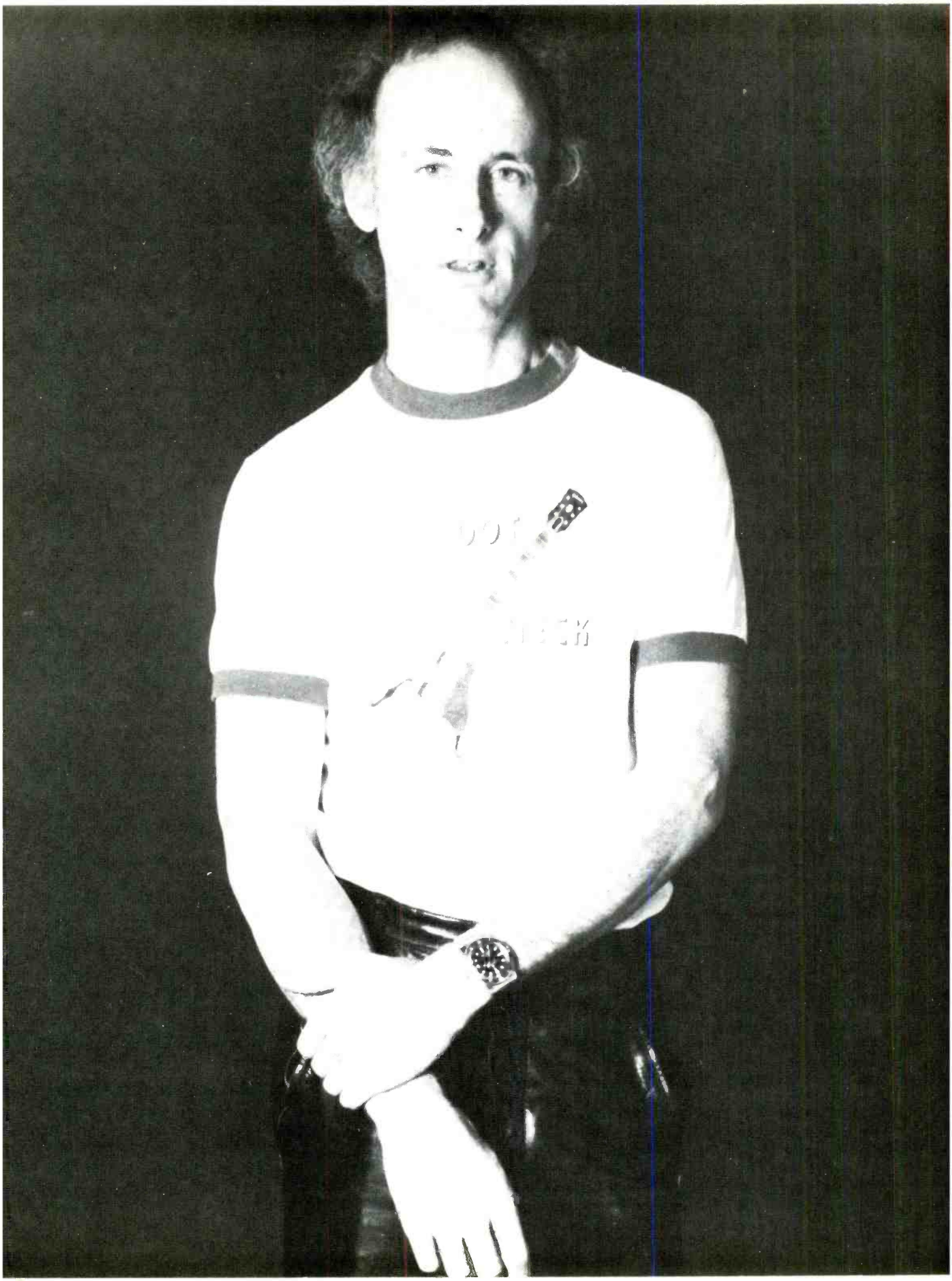
MR&M: Some of the songs on the album are covers, some very interesting choices I might add. Let's run through them one at a time. The first track is "Tattooed Love Boys," a Pretenders song.

RK: That's one of my favorite songs by them. I really like the time changes.

MR&M: The next track is called "Her Majesty."

RK: That's one that I wrote. Ray Manzarek and John Densmore (of the Doors) played on that one. That's about all there is to say about that one.

MR&M: "East End, West End"?



RK: That one was by Eric Gale, who is a great guitarist. He has a few albums, one of them called *Negril*, which was done in Jamaica.

MR&M: The next one is "Crystal Ship," which is, of course, a Doors song. The arrangement must've shocked a lot of people. Was that your idea to alter it so much from the original?

RK: Yeah, I came up with the idea for that and Arthur Barrow put the great bass line on it. John Densmore and Ray Manzarek are on that, too. I just wanted to make it very different from the original.

MR&M: By redoing a Doors song, isn't there a danger of having people think you're trying to hang onto that association?

RK: I'm sure that a lot of people think that it's sacrilege for me to do that. I've found that most people like this version, that it's a lot of people's favorite song on the album.

MR&M: Side one ends with the Stones' "Street Fighting Man."

RK: That's one of my all-time favorite Stones songs. I wanted to recreate the raw sound on that and I thought it would also be a perfect song for me to play slide guitar on.

MR&M: The first one on side two is also a '60s cover, the Four Tops' "Reach Out I'll Be There."

RK: That's one of my favorite Motown songs.

MR&M: Great production on that.

RK: Thanks. Bruce Gary came up with that big drum sound; it's kind of a cross between Gene Krupa and Bow Wow Wow.

MR&M: Next is one called "Gavin Leggit." What does the title refer to?

RK: That's a guy's name. I had a group called Red Shift a couple of years ago and a singer/songwriter wrote that song with me and Arthur Barrow. It used to have words but I can't remember what it was about; I think it was about a bank clerk who got drunk all the time and couldn't keep it down.

MR&M: "Underwater Fall"?

RK: I wrote that a couple of years ago. The title came from one of Jim Morrison's poems.

MR&M: "I'm Gonna Tell On You"?

MR&M: I wrote that one and my wife came up with the title. It had a nice rhymey sound to it that fit the melody.

MR&M: And finally, "Harlem Nocturne," another old song.

RK: We were just trying to come up with some songs to do on the album and my manager, Richard Linnell, and I heard that and thought it would be a good one to close the album.

MR&M: Why did you choose to do so many covers, especially since, as you say, you are a songwriter of renown?

RK: I think if you're doing an instrumental album you have to do songs that people are familiar with.

MR&M: You also produced the album yourself. Did you find it hard to get a perspective on your work in the studio while you were recording?

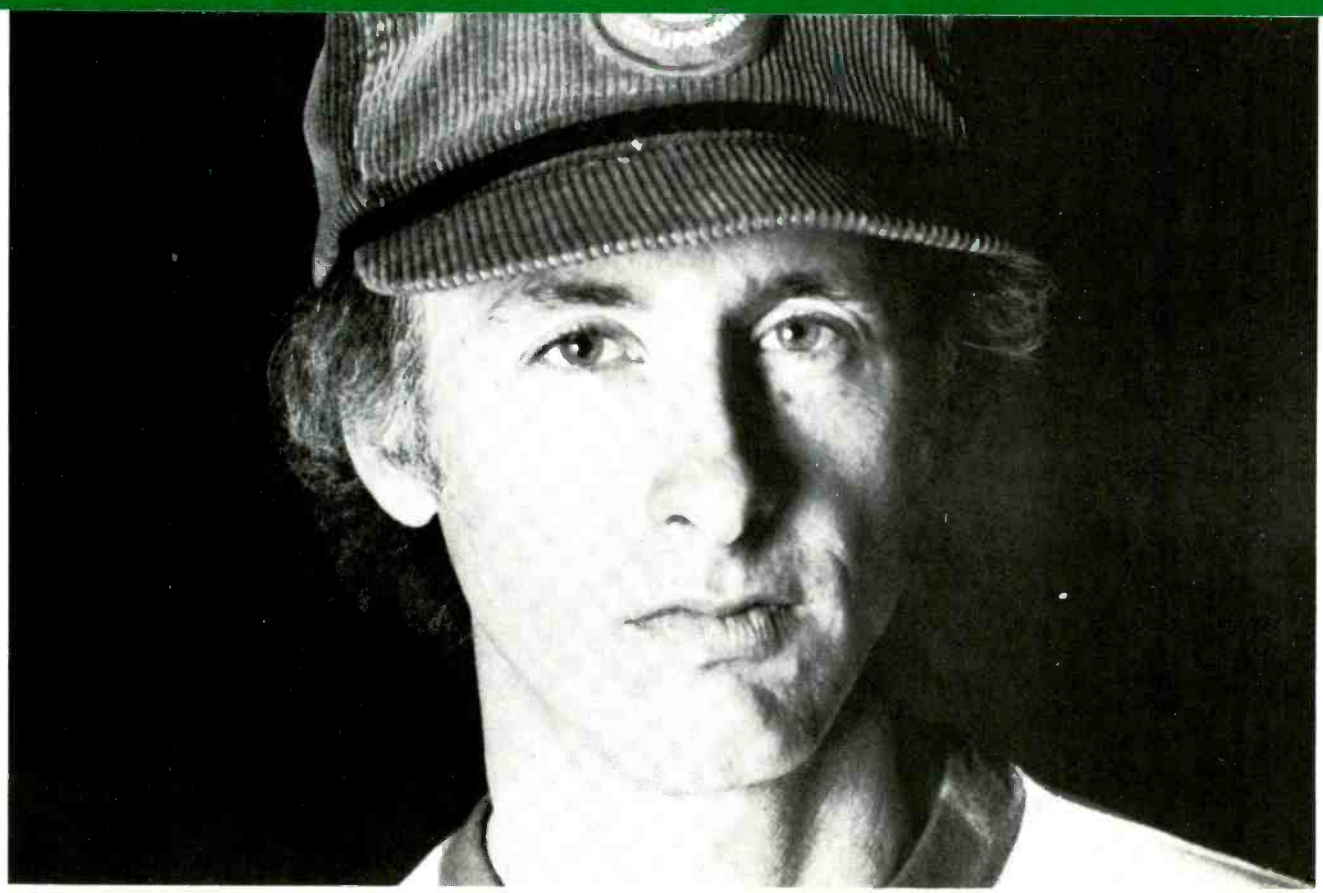
RK: Well, I would rather have had a producer, actually, if I could've found someone really great. But nobody seemed to be available. It is hard to get a perspective, but I'm happy with the way it turned out.

MR&M: What guitars do you use?

RK: I used a '55 Mono Gibson and a Twin Reverb amp. For slide I used a '54 Les Paul.

MR&M: There are quite a few different guitar styles on the record. Were you influenced by anyone in particular?

RK: I try not to copy any other guitar player. But I do listen to others.



I like Alan Holdsworth, Pat Martino, a lot of jazz guys.

MR&M: How has your own playing changed over the years?

RK: I've gotten more technically advanced. And hopefully I've been able to retain some feel. It's sort of a struggle because once you start conconcentrating on the technical, you forego some of the feel.

MR&M: Did the musicians on the album have a lot of input or did you take complete control?

RK: Sometimes they had input in the arrangements, but in some cases I did it all myself. On "Reach Out," for instance, we all worked on the arrangement.

MR&M: What is the stage show like? Are you doing the album? Any Doors songs?

RK: Most of the album. We're doing some Doors songs, though, because people do want to hear that stuff.

MR&M: Robby, I don't think most Doors fans are aware that you wrote some of the biggest hits for them, including "Light My Fire," "Touch Me," "Love Her Madly" and "Love Me Two Times." Does it bother you that people just assume Morrison wrote those songs?

RK: A little bit, but I don't know if people care. Anyway, now it's coming out in these interviews. As long as they don't think Jose Feliciano wrote "Light My Fire." (Author's note: the classical/pop guitarist had a big hit with a cover of that song.)

MR&M: I'm sure you've been asked this a million times, but why do you think the past few years have seen such a resurgence in popularity for the music of the Doors?

RK: A lot of people think it has to do with the book about Jim [*No One Here Gets Out Alive* by Jerry Hopkins and Daniel Sugerman, Warner Books]. But I think it's just natural that each new generation of kids will be able to get into Doors music. They still hear it on the radio all the time.

MR&M: But do you think a 17-year-old who listens to the old records can really understand the context in which all of that was happening?

RK: Probably not. In fact, not at all. But they don't have to to appreciate the music for itself.

MR&M: Looking back now on the whole phenomenon, why do you think the Doors were singled out for so much persecution? Jim was doing some pretty outrageous stuff, true, but so were a lot of others, and they didn't have police harassing them all the time.

RK: I think it's because we sang about the darker side of life. They feel threatened by that. It got to the point where the vice squad would start coming to every show waiting for something to happen.

MR&M: What kind of mark does that leave on a person? Did it really wear you down?

RK: It didn't bother me that much but it did Jim, certainly. I was outside of it pretty much. I didn't get busted, it was always Jim.

MR&M: You're working now with young L.A. bands. Do you see any correlation between the L.A. punk and new wave scene, which is one of the most active in the country, and the scene on Sunset Strip in the '60s that the Doors came out of?

RK: Yeah, I do, but not in a musical sense, because the punk groups are not necessarily good musicians. The similarity is that they're doing something they believe in, that they're telling the truth about what they see around them.

MR&M: How would you compare recording with the Doors in the '60s to recording now?

RK: It's not that different, really. The equipment is just better today. I recorded [*Versions*] on 16-track instead of 24-track. I didn't have any vocals, so why waste the extra tracks? You actually get a better sound on 16-track. The first Doors album was done on 4-track.

MR&M: When the Doors made a record, did the group have the songs all worked out beforehand or was a lot of it improvised in the studio?

RK: It was pretty much down before we went in. Most of the improvised stuff was down on the page first. A couple of things were made up in the studio.

MR&M: What about your guitar solos? Were those worked out note for note?

RK: No. I'd go in and take a lot of tries at it until I got one I liked.

MR&M: When the Doors were first starting out, did you ever have any idea the group would become what it did?

RK: I think we always knew we had the potential to. Jim was always disappointed that it took so long, even though we had a smash the first time out. We envisioned ourselves as a shooting star, up real fast and whoosh!

MR&M: Do you have any particularly strong memories that stick out about the Fillmores and playing rock festivals, that whole era?

RK: Oh, just that it was strange (laughs). The first time we went to San Francisco we went to the first Human Be-in and it was incredible. I really thought we were going to change the world. We played that night and it was great.

MR&M: Do you think that the combined effect of the Doors and all the other bands from that era changed the world somehow?

RK: Well, I hope so.

MR&M: Do you think the Doors progressed musically over the years or were you somewhat held back because of all of the trouble that surrounded the band?

RK: Well, I think all of that was stifling. We didn't have a lot of time to grow musically because we were always doing shows. That's pretty much why I didn't want to have another group after the Butts Band and just wanted to develop my own style.

MR&M: Were you happy with the later Doors albums? Do they hold up?

RK: Oh yeah, we never put out anything we didn't like. I think they still hold up; I listen to them. Some of it is embarrassing now. I wasn't crazy about some of the stuff on the *Soft Parade* album.

MR&M: What ultimately led to the end of the Doors? You did carry on for a couple of years after Jim died.

RK: Yeah, but that really broke up the band. I think we would've kept going on if Jim hadn't died.

MR&M: What effect did Jim's death have on you personally?

RK: I wasn't surprised by it. In fact, I was surprised he lasted as long as he did. I guess I prepared myself for it a long time before it happened. I still think about him. Of all the people I know who have died, I have a Morrison dream at least once a month.

MR&M: If he had lived, what direction do you think the Doors would have taken?

RK: I think it would've been more blues.

MR&M: Were you happy with the two Doors albums made after Jim's death?

RK: Yeah, I think there was some good stuff on both of them.

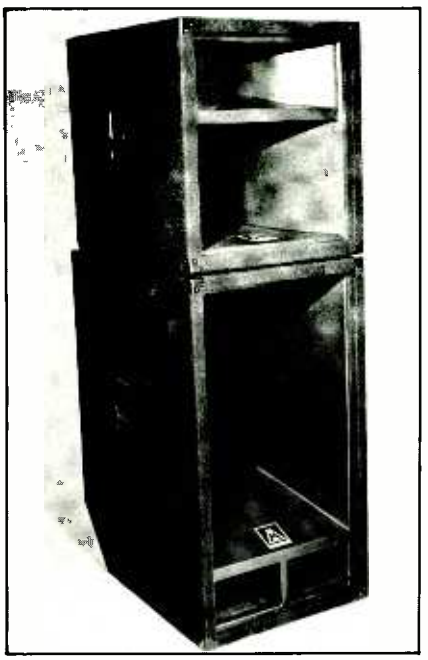
MR&M: Where are you going from here? Will you be doing another album?

RK: We'll do some touring and then maybe another album after that.

The Market Place

what's new in sound and music

HARBINGER AUDIO 828 TRI-AMP P.A. SYSTEM



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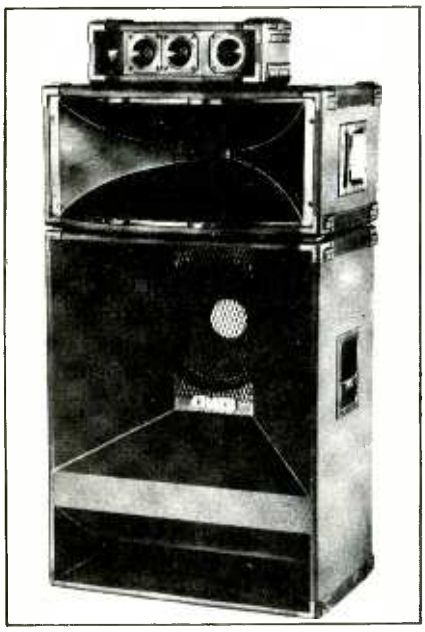
Circle 32 on Reader Service Card

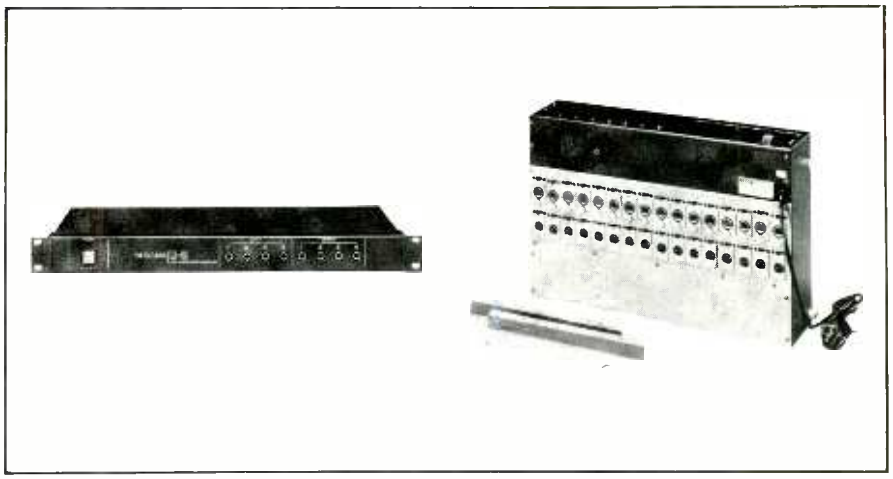
CRATE SYSTEM 80 STACK



The Crate System 80 Stack is now available from SLM Manufacturing, the electronics division of St. Louis Music Supply Co., Inc. A horn-loaded bass bin (PS1580), a separate radial horn assembly (PS80R), and the new Crate Tweeter Bank (PS80TB) have been combined to complement each other in reproducing and projecting a big sound. The PS1580 horn-loaded bass bin has a smooth low frequency response extending down to 40 Hz. The PS80R radial horn assembly has an extremely wide range, extending to 12,000 Hz. Radial design aims the sound where desired, projecting into the audience. The PS80TB Crate Tweeter Bank contains four Foster Bullet tweeters to add crispness to music or voice. The compact cabinet can handle a full 200 watts (RMS).

Circle 33 on Reader Service Card

TASCAM LINE AMPLIFIERS



Tascam's newest line amplifiers include the LA-85 and the LA-40. The LA-85 converts the 85-16 and 85-16B recorder/reproducers to accept and control a +4 dBm balanced signal from three-wire XLR-type connectors. Both input and output circuits of the LA-85 are transformer-less and fully balanced. The output stage delivers 19.5 volts into a 600-ohm load and can drive long cable lines without suffering from signal loss. The output operating level of the LA-85 is switch-selectable between +4 dBm or -8 dBm. Output signal-to-noise measures 100 dB referenced to +4 with the D.I.N. 20 Hz to 20 kHz filter, and crosstalk between sections is 80 dB. The compact, four-channel LA-40 matches balanced and un-balanced circuits and serves as an adapter between mismatched line levels or impedances. It permits interconnection between +4 dBm, -20 dBm, and -10 dBV inputs and outputs, and also establishes compatibility between Tascam or similar -10 dBV unbalanced inputs and outputs. Each of the LA-40's four channels are actually two circuits back-to-back. By plugging a jumper across the RCA in and out jacks, the LA-40 isolates a line or converts from unbalanced to balanced operation when both the input and output are at the same +4 dBm or -20 dBm nominal level. The LA-40 utilizes either XLR, phono, or phone connectors. The phone jacks are located on the front panel where they can be used in conjunction with a standard TRS jack patch bay. All other connections are located on the rear panel. This system is able to drive or be driven by any line-level equipment.

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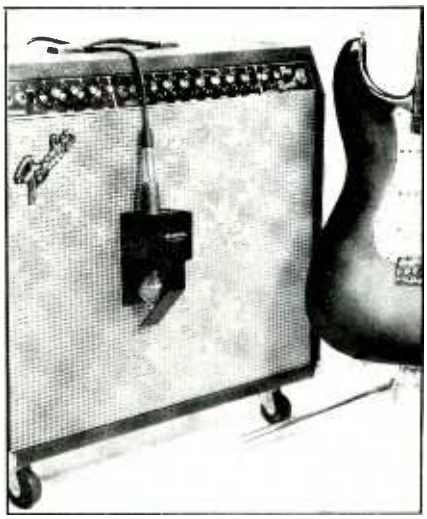
HARDY TECHNOLOGIES REFLECTOR™



Hardy Technologies' new Reflector™ Microphone Sound Reflector is designed for sound reinforcement purposes where direct mic'ing of an instrument amplifier is desired. It features a brushed black or gold anodized aluminum body and reflector plate (secured together with velcro) and an internal foam isolator. The Reflector weighs 10 oz. and accepts most conventional microphones.

Circle 34 on Reader Service Card

C-TAPE DEVELOPMENTS C-DUCER

C-Tape Developments' C-Ducer Saxman employs an easily mounted reed transducer requiring no modifications to the instrument or change of playing technique. The system, which also features a battery-powered preamplifier, provides reed players the independence from microphones and feedback that electric instruments have long enjoyed. The Saxman comes in a sturdy carrying case with several accessories that give the option of mounting the preamp in three positions: on the instrument, on the player's belt, or on a music stand. The last two options enable quick changeover when the musician is doubling up on instruments.

Circle 36 on Reader Service Card

BIAMP SYSTEMS CROSSOVERS

Biamp Systems' new family of crossovers consists of the SX/35 five-way mono, three-way stereo; the SX/23 three-way mono, two-way stereo, and the MX/2, a mono version of the SX/23. All three crossovers have 18 dB slopes and are switchable (X1 to X10) to expand and select the crossover points with greater accuracy. All inputs and outputs are floating and balanced in order to maintain low noise, high slew rate, high-level output, and complete isolation without transformers. Other features include phase switching on high-frequency outputs, input level control with +12 dB gain, subsonic filter (20 Hz), and signal, power, and peak LED indicators. Specifications: frequency response, ±0.1 dB (20 Hz to 20 kHz); THD, less than .01 percent (20 Hz to 20 kHz); hum and noise, less than -87 dBm. The dimensions of the new crossovers are 1¾-in. high by 19-in. wide by 6-in. deep.

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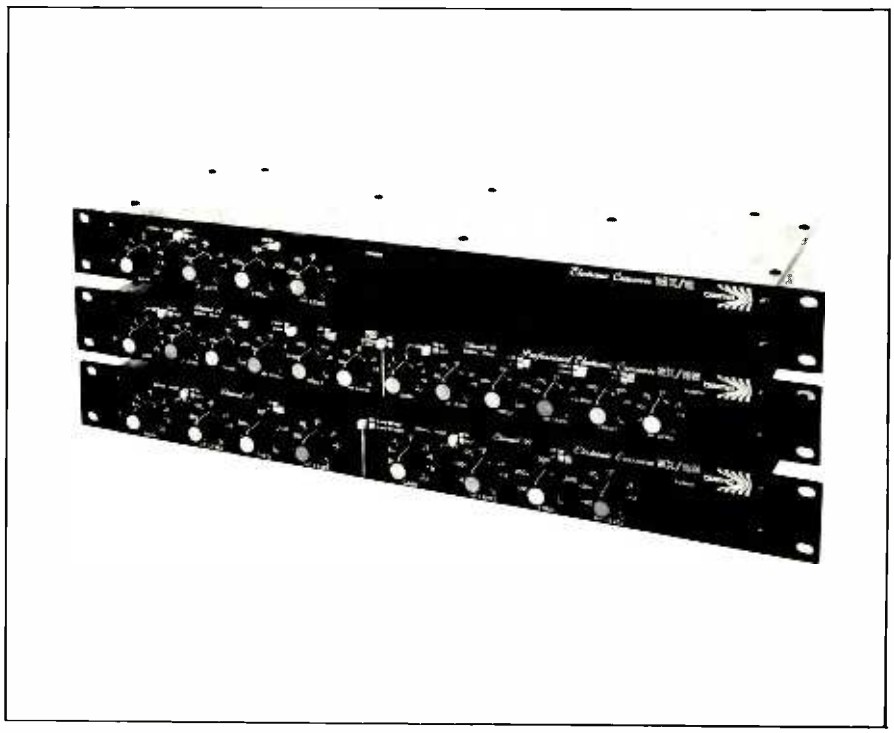
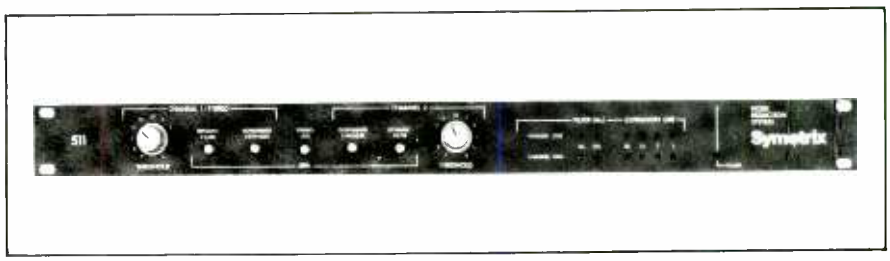


SYMETRIX STEREO NOISE REDUCTION SYSTEM



Symetrix Inc.'s new Model 511 Stereo Noise Reduction System is a non-complementary or what is sometimes referred to as a "single-ended" type system. This means that it does not rely on an encode/decode process to accomplish the noise-reduction function. The obvious advantage to the user is that the Symetrix unit may be used to remove existing noise from pre-recorded tapes or from virtually any audio source including noisy mixing boards effects and processing devices, etc. The 511 is suited for recording and broadcast studios, video post-production, video and audio tape duplication facilities, and professional sound re-reinforcement companies. In operation, it works to eliminate noise by incorporating a "soft knee" downward expander. The user may choose to use either circuit element independently or both at once. The amount of actual noise reduction is controlled by a single variable threshold control, and visual indicators are provided for both filter and expander circuit operation. The two channels of the 511 may be used either as independent mono channels or in the stereo link mode, where equal amounts of noise reduction will occur in both channels. The 511 is packaged in a convenient, single rack-space chassis. Electronically balanced inputs and outputs are provided in addition to standard unbalanced inputs and outputs. The manufacturer's suggested retail price for the unit is $595.00.

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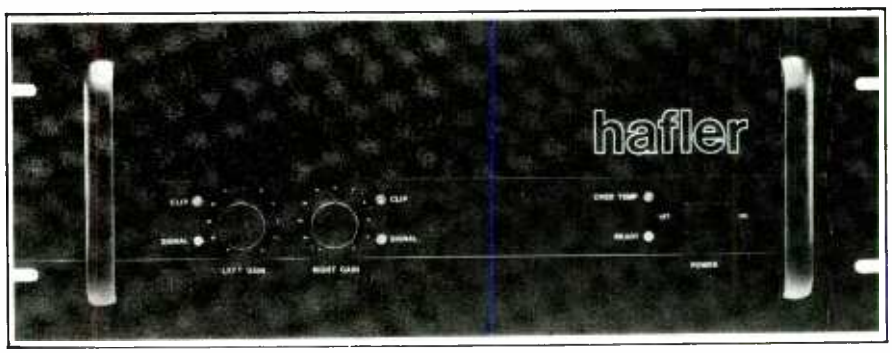
HAFLER POWER AMPLIFIER



The David Hafler Company's P500 Power Amplifier is a full-featured version of the DH-500 Audiophile Amplifier designed to withstand the rigors of professional use. The P500 delivers 400 watts-per-channel into 4 ohms at less than .04 percent THD from 20 Hz to 20 kHz with both channels driven. In the bridged mono mode it is capable of delivering over 800 watts into an 8-ohm load. The amplifier uses a push-pull complementary symmetry circuit design which employs MOSFET output devices. Because the amplifier uses MOSFETs, which are inherently self-protecting,the P500 has no need for extra protective circuitry that can be sonically degrading. It does, however, contain an AC line current-inrush limiter and a protection relay for DC offset with turn-on delay to protect loudspeakers from switching transients. The front panel contains a power switch with a 20 ampere rating, a two-color ready light that initially glows red and turns to green when the load is connected, an over-temperature indicator, two 31-position laser-trimmed input level controls, plus separate signal present and clipping indicators for each channel. On the rear panel are five-way heavy duty binding-post output connections, input connections via barrier strip, XLR connectors and two sets of ¼-in. phone jacks in tandem, both circuit and chassis grounds, and provisions for balanced and unbalanced inputs and a mono/stereo switch. For cooling purposes, the amplifier is equipped with an automatic three-speed fan and specially designed cooling tunnel. The P500 employs thermal circuit breakers for each channel that will automatically shut down the amplifier in the case of overheating. As the amplifier cools, the thermal breakers reset. The P500 is capable of operating from a 100, 120, 200, 220, or 240 volt, 50 or 60 Hz AC line, and is equipped with a three-wire grounded AC power cord. Dimensions are 7-in. high by 19-in. wide by 14-in. deep, including rack mount handles. A four-point suspension system ensures mounting integrity for road use. The unit is finished in black and will be available in fully or partially assembled form. The Hafler P500 retails for $949.95 fully assembled and $799.95 partially assembled.

Circle 39 on Reader Service Card

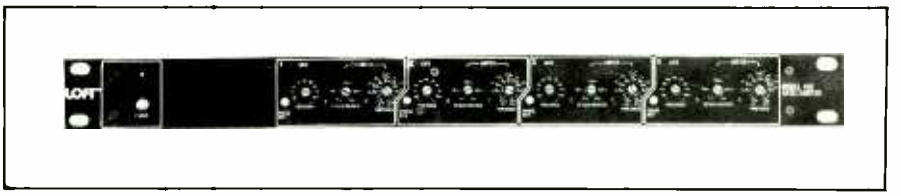
PAIA PERSONAL AMP

PAIA Electronics' Gemini is a low power personal amp intended for studio work and private practicing. In the studio, the compact size and dual six-inch speakers allow for creative mic'ing (for example a mic or pair of mics may be moved closer or further away from one or both speakers), while battery operation minimizes hum. Built-in distortion, three-band equalizer, and master volume control simulate sounds that come from much larger amps. There's also a second "clean" input (switchable pre- or post-EQ) with associated attenuator. In addition, the Gemini includes a pre-EQ effects loop for adding external signal processors, as well as stereo line inputs for mixing records or tapes into the Gemini for practicing or jamming (these inputs may also serve as post-EQ effects receive jacks). There is also a headphone jack and speaker on/off switch. The Gemini is available in kit form for $79.00 (plus $5.55 postage/handling).

Circle 40 on Reader Service Card

LOFT MODEL 400 QUAD LIMITER AND NOISE GATE

The Loft Model 400 is a four-channel feedforward limiter and noise gate (i.e. four limiters with noise gates in a single unit). It is an ideal outboard signal processor for recording, broadcast or sound reinforcement applications. The model 400 can also be used for sound reinforcement systems to control the maximum level and reduce noise. The front panel controls include: noise gate threshold ("-" infinity to 0 dBv), limiter threshold (-12 to +12 dBv), and limiter attack/release time (1 ms - 1 sec). Front panel LEDs help make adjusting the controls simple. A phase reversal switch is also included on each channel. The model 400 is a rack-mount unit that occupies one 1¾-in. rack space and comes standard with balanced inputs and ¼-in. phone connectors. Options include XLR connectors with electronically balanced inputs and outputs and/or recessed panel controls. Suggested retail price is $649.00.



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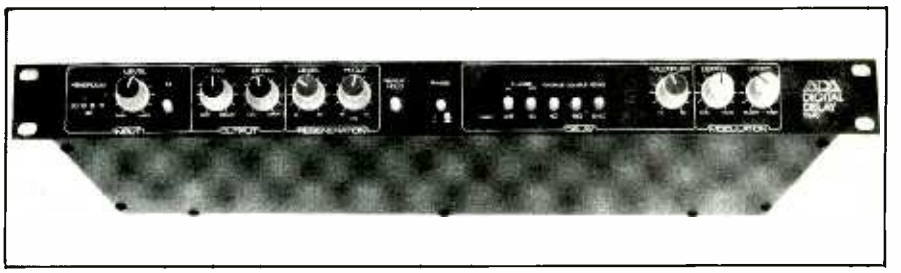
ADA D640 DIGITAL DELAY

ADA's new D640 Digital Delay is a full function delay processor designed for both live performances and recording use. The D640 produces delay times from 0.25 ms to 640 ms at a full 15 kHz bandwidth. For producing special effects, the D640 has Regeneration, Modulation and Repeat Hold features. The Regeneration Hi-Cut control reduces the high frequency content in the delayed audio signal as it recirculates for more natural sounding repeat echoes and reverb. The Modulation section has a Depth control that sweeps the delay time up to a 10:1 range for flanging that sweeps over three and one-third octaves. The Speed control varies the sweep rate from 25 seconds to 0.1 second for a complete sweep cycle. The Repeat Hold feature allows the entertainer to lock in up to a 640 ms musical segment and repeat it indefinitely for background rhythm effects. The D640 fits one standard rack space and has a 4-step LED Headroom indicator. Options include the FS-2 Dual Footswitch for Effect Bypass and Repeat Hold In/Out, and 240VAC power supply. The suggested list price is $499.95.



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SUNN MIXING CONSOLES

The new Sunn SPL 3416 and 3424 mixing consoles are full-featured 16 and 24 channel mixers with four independent submasters into two into one. The assignment systems, patching capabilities and internal jumpers make these boards ideal for live P.A. main, monitor or multi-track studio applications. All internal buses are attached via a quick disconnect multipin cable. All PC boards are made of fiberglass/epoxy laminate.

Circle 43 on Reader Service Card

len feldman

On Tape Speeds

I recently received a provocative letter from a long-time regular subscriber to *MR&M* which raised questions that are worth exploring in some detail. Let me quote parts of his letter first, and then I'll comment on the points raised by Mr. Phil Giargiari of Ashland, MA.

"I am a...pro-audio enthusiast/disc jockey, etc. In the February, 1979, issue of the magazine, you and Mr. Eisenberg did a lab report on the first cassette deck capable of 3¾ ips operation, the B.I.C. T-3. Up to then I had been unimpressed with 'normal' cassette equipment regardless of price or sophistication, and did most of my serious recording on reel-to-reel. After reading the report, however, I decided to do a little research.

"I immediately ran out and purchased the B.I.C. unit....*At last*—headroom, wow/flutter, and frequency response on a par with open reel! The deck sold me on 3¾ operation." He goes on, "...I traded up to a TEAC C-3X (high speed version of the C-3). This machine also checked out well and has run reliably for three years.

"...Now, here we are in 1983, and while standard decks have improved somewhat (especially with Dolby-C), I'd still rather make high speed cassettes on medium priced, single-capstan machines. Perhaps you can shed some light:

"(1) Why has 3¾ operation been suppressed in today's market? Except for the Tascam 122 and Marantz's SD-9000 flagship, it seems to have vanished. (Is B.I.C. still in business?)

"(2) If Philips is the problem, when are they due to release their grip on the deck makers?"

To begin with, Mr. Giargiari, as far as I or anyone else has been able to determine, Philips is not the problem. If Philips had wanted to prevent manufacturers from coming out with 3¾ ips machines, B.I.C. would never have gotten as far as they did with several models of two-speed cassette decks, nor would Marantz be selling such cassette decks to this day. As it happens, at the time that B.I.C. was planning to introduce its then-revolutionary two-speed cassette decks, I was a part-time consultant to that company and I recall that extensive investigation and research was conducted by the legal department of Avnet, Inc. (B.I.C.'s parent company) to make certain they would not be inviting legal difficulties if the project went forward. By now, of course, the patents involved in the compact cassette standard have expired in any event. (Patent protection in the U.S. lasts only 17 years from the date that a patent is granted.)

As I'm sure you are aware (or will be, after you read this issue of *MR&M*), 3¾ ips speed for cassettes is anything but dead on a professional level. Our test report covering Tascam's 4-track, 4-channel model 234 Syncaset unit (in last month's *MR&M*) more than confirms that. As for 2-speed cassette decks at lower prices and intended for consumer use, I suspect that the flurry of 2-speed cassette development that we witnessed in the late 1970s has died down for several other good and valid reasons.

Metal Tape

Perhaps the most important of these reasons was the development of metal particle cassette tape which occurred at just about the same time that B.I.C. introduced its two-speed series of decks. Metal tape offered most of the electrical advantages formerly found with oxide-formulation tape and high-speed operation. Specifically, high-frequency headroom was dramatically improved with no sacrifice in signal-to-noise ratio. In fact, when properly biased, metal tape actually delivered slightly better overall dynamic range than the best chromium dioxide or cobalt-doped (high bias) tapes of the time. When metal-particle tape was first introduced, it cost approximately twice as much as premium high-bias tape. In terms of cost-per-minute of recording, that made it an even trade-off (all other things being equal) with using conventional high-grade at double-speed. Since then, however, the cost of metal tape has come down substantially so that it is actually less expensive to use metal tape at 1⅞ ips than to use premium high-bias tape at twice that speed.

One of the most important reasons for the world-wide success of compact cassettes was the standardization of operating parameters such as speed, track configuration, and playback equalization. The initial strict control and licensing of the cassette format by Philips contributed to that world-wide standardization. By the time individual companies decided to introduce 3¾ ips cassette decks, there was no longer central control of operating parameters, and each of the introducing companies was free to use its own optimized parameters for the new machines. Thus, total compatibility—a feature of cassettes that had been the main objective of Philips—could no longer be guaranteed for those machines that operated at speeds other than 1⅞ ips. And lest we forget, B.I.C. and Marantz were not the only ones to introduce 2-speed machines. At one point, Fisher and a few leading Japanese companies also introduced two-speed models. It is also interesting to recall that Nakamichi, one of the most respected names in cassette deck technology, was against the 3¾ ips speed for consumer use even then. To show their disdain for the

higher speed in a cassette deck, they deliberately introduced a few models of decks that, in addition to operating at standard 1⅞ ips, also could record and play back tapes at 15/16 ips!

To be sure, Nakamichi's 15/16 ips models were more of an exercise in clever engineering than a crusade for the slower speed. The units were actually able to record with reasonably flat response out to 15,000 Hz at the slower speed. For many casual recording applications, that was plenty good enough, and, of course, the cost of tape was reduced by half—unless you resorted to metal particle tape to compensate for the reduced headroom at high frequencies and the poorer signal-to-noise ratios. After a year or two, Nakamichi abandoned this two-speed approach much as the others abandoned their 1⅞, 3¾ two-speed approach.

Wow-and-Flutter

Mr. Giargiari cites another advantage of the 3¾ ips cassette speed: reduced wow-and-flutter. There is a bit of a misconception here, which is perhaps attributable to the advertising and promotion employed by B.I.C. and others at the time their two-speed decks were introduced. The fact of the matter is that if you double the speed of a tape system that has a given amount of wow-and-flutter in its tape transport system, you are not going to reduce the absolute values of wow-and-flutter. What you will do is double the *frequency or repetition rate of the various wow-and-flutter components.*

Now, here comes the tricky part. Suppose a deck operating a 1⅞ ips had its predominant wow component situated at 8 Hz (a not uncommon frequency for a wow component). That wavering in speed might be caused by any one of several rotating parts in the tape transport system, from the drive motor shaft, to the tape capstan, pinch roller, etc. Now, when you up the speed of the tape to 3¾ ips, without changing anything else but DC motor rotation, that wow component now moves up to a frequency of 16 Hz. But 16 Hz is no longer considered "wow." It is a "flutter" frequency. Typically, when measuring "wow-and-flutter," we use a weighting curve that emphasizes "wow" frequencies (which are subjectively more annoying) and de-emphasizes the effect of higher, flutter frequencies (which the human brain can tolerate in greater amounts without finding them audibly intrusive). So, as you can see, it isn't the wow-and-flutter that's been reduced in a higher-speed cassette deck, it's a change in the wow-and-flutter content that has occurred.

Meanwhile, consider what has happened to wow-and-flutter in standard speed machines over the last four or five years. Whereas 0.1 percent wow-and-flutter was considered "state-of-the-art" just a few years ago (that's all that was claimed by the now-classic original Nakamichi 1000 three-head deck introduced in the mid-1970s), even low-cost consumer machines now boast (and actually measure) wow-and-flutter figures of 0.06, 0.05, even 0.04 percent! These are *real* improvements in wow-and-flutter, and not just shifts in frequency of wow components.

Mr. Giargiari cites Dolby C as one of the improvements that has appeared in standard speed cassette decks, but he fails to mention dbx as an equally significant one. A glance at the test results for the previously mentioned Tascam 234 Syncaset will disclose the degree of improvement in signal-to-noise as well as dynamic range and high-frequency headroom brought about by the incorporation of dbx linear companding in both professional machines such as the Syncaset, and in consumer cassette decks. It is for this reason that the owner's manual for the Tascam 234 Syncaset suggests leaving the dbx circuitry turned on all the time. In fact, they mount the dbx on/off switch on the rear panel of the unit, rather than up front.

It seems to me that if 3¾ ips cassette operation deserves to survive at all, it should do so in the world of professional or semi-professional audio recording and not in the already confused and confusing world of home audio. The Tascam 234 Syncaset is an outstanding example of how this higher cassette tape speed can be used most effectively. Here we are dealing with a "closed loop" situation, in which tapes made on the machine will be played back on the same machine, either for auditioning purposes, or for further electronic editing and mixing or multi-track overdubbing. The controls that are featured on this type of unit are primarily those that aid the serious recordist. It's for that reason that such features as tape selection, bias adjustment, azimuth adjustment and the like have been omitted, while gain and fader controls have been amply provided. Yes, there's still a place for 3¾ ips cassette decks—but it's not in the field of home audio.

Lab Report

len feldman

Sony CDP-701 Compact Disc Player

General Description: After a successful introduction in Japan last year and in Europe this past spring, CD players have finally arrived in the U.S. Debates concerning the wisdom of standardizing on a 16-bit, 44.1 kHz sampling rate continue, as do subjective discussions on the quality of sound delivered by this revolutionary laser-optical digital disc. Regardless of where you stand on these matters, CD discs are going to affect the recording industry in many ways, and, sooner or later, you'll need to equip your studio with one of these new players, if only to hear what the competition is doing. All this explains why we have chosen to test what is basically a new consumer record player for *MR&M*. The compact disc is going to become an important part of the modern recording scene, and we should all keep our eyes (and ears) on it.

Sony's first CD player (model CDP-101) went on sale in March of 1983 and is still out there, in more or less of a back-order position. Since then, other manufacturers have come up with more elaborate programming and cueing features. Sony must have realized that that's the price of being first, and so they readied a second-generation player, the Model CDP-701, which is the subject of this report. The new CDP-701 incorporates many programmable features which were not present on the earlier model but which have found their way into CD players introduced by some of Sony's competitors (who are, of course, also licensees). The new buttons and displays are all in addition to the controls and displays that were present on the older model, so let's quickly review those older controls and then I'll tell you about the new ones.

Controls & Switches: The power switch and timer/play switch are located at the lower left of the new front panel. After the power switch has been pushed, the digital display area at the right of the disc-loading drawer displays the numeral 1. Touch the OPEN/CLOSE button, and the disc drawer opens smoothly. Then, drop the disc you want to play into the drawer (no centering, no spindle to fumble with) and touch either the PLAY button or the OPEN/CLOSE button once again; the drawer closes slowly, centering the disc on the internal drive spindle. If you pressed PLAY, disc play begins in a couple of seconds. If you close the drawer by means of the OPEN/CLOSE button, you then have the option of touching the PLAY button to initiate play, or you can depress one of the track-advancing buttons (FORWARD, in this case) to advance the numeral on the display to any track of the disc you like. A PAUSE button is located just below the PLAY button. A reversing button next to the track-advancing button brings you back to the beginning of the track currently being played.

After a disc is loaded, it is automatically scanned and a ruler-like display appears, calibrated in

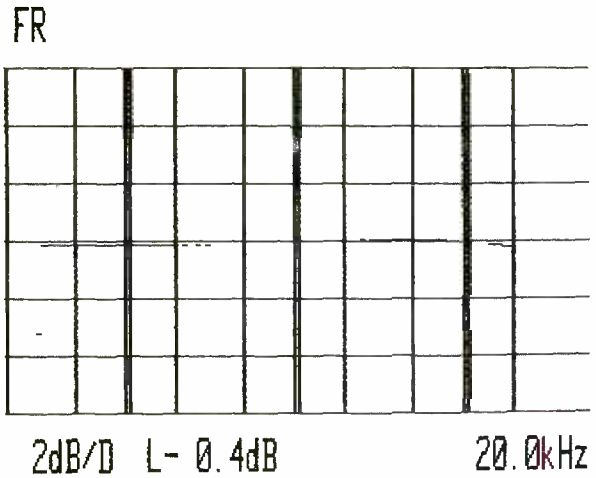


Figure 1. Frequency response of Sony CDP-701 player at 0 dB (maximum) recording level was virtually ruler-flat. Vertical scale has been expanded to 2 dB/division.

minutes. Highlighted markers appear along the ruler, showing you the approximate time at which breaks between successive tracks occur. During play, the marker corresponding to the point in the disc being played flashes continuously, with successive markers flashing as play moves along from minute to minute. Time into a track is displayed in minutes and seconds, while a second real-time display tells you how many minutes and seconds of playing time remain on the disc. In addition to displaying track numbers, the display area also shows Index numbers. Index numbers may be used in the future to sub-divide long selections or tracks on CDs, such as movements of a symphony, into smaller, identifiable subsections. Other buttons carried over from the earlier model include two with double arrows pointing in each direction and two with triple arrows. These allow you to fast-forward or fast-reverse play *while listening to the program material*. These functions are completely analogous to the fast-scan functions now found on most video tape recorders, except that as the buttons are held down, you don't hear a change in pitch of the music, just a fast sampling of what's on the disc so that you can elect to stop at a desired spot. Three repeat-play buttons offer that many options—specifically, repeat of the track being played, repeat play of the entire disc or repeat of specifically designated material (from one point to another) on the disc.

A RESET button cancels play at any point, returns the laser pickup to its internal rest position and stops the rotating of the disc. Adjacent to the elaborate display area just described is an infra-red sensor that detects instructions sent by the wireless hand-held remote control unit supplied with the CDP-701. This remote control unit is identical to the one supplied with the earlier model. This slim, hand-held module, powered by two AA batteries, duplicates just about all of the front panel functions, including the memory repeat options, fast audible scanning and track selection. In addition, a 0-to-9 keypad allows you to dial up any track number without having to step through lower or higher numbers to get to that track.

Three important new programming and accessing features, found on this unit, are particularly useful for those who plan to use the CD player as a source of music in a multi-track mix-down, as a source of background music, or in other studio applications that require fast and accurate cueing and timing. First, there is the feature called "Music Scan," activated by a similarly labeled touch button. When this button is depressed, the player offers you ten seconds worth of music from each track of the disc to be played, allowing you to audition the various selections. When you hear something you like, you have but to touch the PLAY button, and that track will be played all the way through.

The second new feature is known by its initials, RMS (Random Music Selection). Using a combination of numeric keypad entries and buttons labeled TRACK and START, you can program up to eight different tracks to be played in any order you wish. As the program track numbers are chosen, the display shows you how many choices you have made thus far, appending a minus sign (-) to the number, so as not to confuse it with a track number. A CLEAR button cancels all or any part of the program should you change your mind about what you want to hear and the order in which you want to hear it.

The third new feature allows you to initiate play at

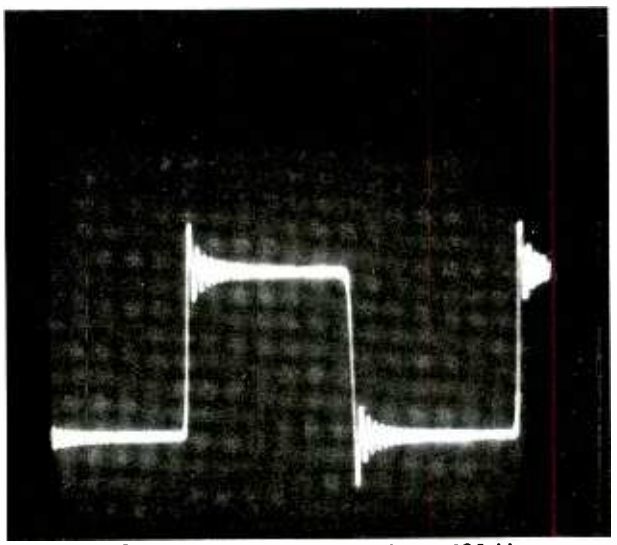


Figure 2. Square-wave response for a 400 Hz square wave shows some evidence of ringing caused by sharp cut-off low-pass filtering.

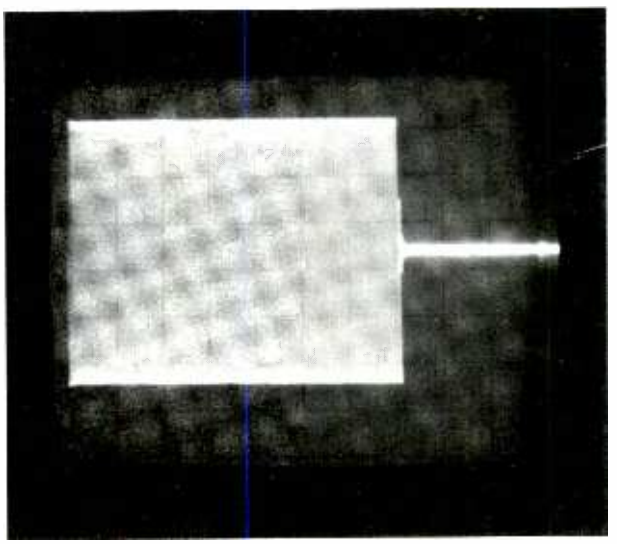


Figure 3. Tone burst signals on test disc were reproduced with some delayed decay at trailing edge of burst.

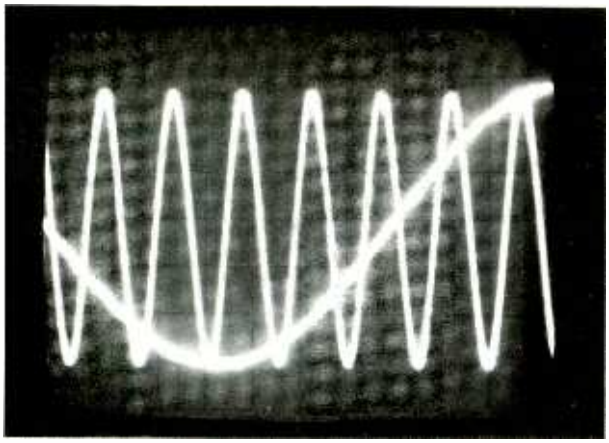


Figure 4. High-frequency (20 kHz) signal on one channel was reproduced with a moderate amount of phase-shift relative to lower frequency signal (2 kHz) on opposite channel.

any point within any track. The sequence of programming starts by touching a button labeled LOCATION. This is followed by keying in the track number desired and the minutes and seconds into that track when you want play to begin. The numeric keypad now serves the double purpose of keying in the track as well as the time in minutes and seconds. The previously mentioned START button initiates the search and play at the desired point in the disc. This feature, as well as the other new programming capabilities, is accompanied by appropriately informative displays within the enlarged display area.

A stereo headphone jack is located at the lower right of the front panel with an associated step attenuator that adjusts headphone level. The attenuator is calibrated from 0 dB to -24 dB. Decoded analog audio signals are accessible from the rear of the CDP-701 via a pair of phone-tip jacks. There's a "beep" on/off switch that works in conjunction with the hand-held wireless remote control unit supplied with the player. If you want the system to "beep" and inform you that remote instructions have been received, leave the switch on. A multi-terminal ACCESSORY PORT socket is intended for connection of a more elaborate system-remote control unit that will be available in the future.

Test Results: Trying to measure the performance of a CD player is somewhat of an exercise in futility. Most of the specs are better than the capabilities of the test equipment used to make the measurements. The test records I used were supplied by Philips and by Sony, who, incidentally, are the partners in the CD disc format that has become a world standard.

Frequency response plots for both the left and right channels of this CD player are shown in *Figure 1*. Note that the sensitivity of the amplitude (vertical) scale has been expanded to 2 dB per division. Maximum deviation from absolutely flat response is -0.4 dB at 20 kHz, the highest test frequency of the sweep.

Several tracks of the new Philips test record are devoted to measurement of total harmonic distortion over a wide range of frequencies from 41 Hz to 20 kHz. In a digital audio system, best distortion figures are obtained relative to maximum (0 dB) levels. Therefore, the disc also includes a series of test tones at -24 dB and at -30 dB so that the tester can measure how THD rises with decreasing signal levels. At mid-frequencies, THD measured 0.003 percent referred to maximum output. As expected, THD increases at lower recording levels, although even at a -30 dB level it was still acceptable with readings of 0.075 percent. Results were virtually identical for the left and right channels.

Signals are provided on the test disc at 997 Hz for checking linearity of the playback system. Levels are 0 dB, -1 dB, -6 dB, -12 dB, -24 dB, -60 dB, -80 dB and -90 dB. There was no significant deviation from perfect linearity until the test got down to -80 dB, which we read as -78.5 dB.

Using the appropriate tracks on a Sony test disc, we measured channel separation at four frequencies. Separation measured around 86 dB for both channels at all of the test frequencies (100 Hz, 1 kHz, 10 kHz and 20 kHz). These readings probably include some residual noise of the test setup, rather than actual crosstalk, which was not detectable as such on our oscilloscope display.

Two tracks are provided on the Philips test disc for checking SMPTE-IM distortion. We measured 0.0065 percent IM at 0 dB level; 0.04 percent at a -20 dB record level. Signal-to-noise ratio measured exactly 100 dB with reference to 0 dB (maximum) output level, A-weighted.

Square waves at frequencies of 100 Hz, 400 Hz, 1 kHz and 5 kHz are provided on the Philips Test Disc, as are various tone bursts, pulse signals and phase checking pairs of signals. The 1 kHz square wave signal is a good one to display (See *Figure 2*), since it clearly shows the degree of ringing that is produced by the sharp cutoff filter found in this and other players. Tone burst signals are also provided on the same Philips test disc, and in *Figure 3* we see a slightly delayed decay at the trailing edge of the signal, consisting of a 4001 Hz signal of 0.026 second duration with a repetition rate of 2 Hz.

Of the several phase-check tests available on the Philips test disc, the one we used has a 2 kHz signal recorded on one channel while a 20 kHz signal is recorded on the other. In this test, the positive-going crossing of the zero-axis of the lower-frequency signal is supposed to occur at the same instant that the higher-frequency signal crosses the zero axis, also in a positive-going direction. Examining the two sets of sine-waves in *Figure 4*, you can see that the low frequency signal crosses the zero axis in a positive-going direction a few microseconds ahead of the high-frequency signal's positive crossing. The high-frequency signal is displaced relative to the mid-frequency signal by approximately 20 degrees. That works out to a phase-error time delay of around 3 microseconds.

Philips has also created a test disc that is used to check the ability of a CD player to track properly even when certain user-caused defects are present on the disc. The disc contains several musical selections, much as a normal CD disc would. In addition, an opaque radial wedge (increasing in width from inner to outer radius of the disc) simulates a bad scratch on the disc. Several black dots of increasing diameter on the surface of the disc are meant to simulate dust particles. A third "defect" is a simulated fingerprint smudge near the outer edge of the disc. We found that the Sony CDP-701 played all the way through the 900 micron

width of the wedge with no muting (that's the greatest width of the opaque wedge, and about as good as any error correction system could be expected to perform); nor did the system mute when playing selections that were covered by any of the three black dots that were meant to simulate dust particles of increasing diameter on the surface of the test disc. As for the fingerprint simulation, the CDP-701 ignored it completely, playing right through the selections that were marred by this defect as if it weren't even present.

Comment: So far, it looks as though most of the CD players that have come to market produce equally good sound. To be sure, some makers are claiming circuitry innovations that are supposed to make their units sound better than the next fellow's, but I maintain that the software is still the determining factor when it comes to CD records. Perhaps some day, when a new breed of recording engineers learns to live with digital discs (*and* how to record for them), the software will become so good that we'll be able to "hear through" the discs to the more subtle differences between one player and the next.

Until that time, the only criteria I can use to judge the merits of a given machine are the ease with which one can operate it, the convenience features that it provides, and the degree of random access programmability that it affords. In all these respects, the second generation Sony CDP-701 is far ahead of its predecessor. And since that first Sony CD player introduced "way back then" was an outstanding piece of equipment to begin with, it follows that the newer CDP-701 is, to my mind at least, nothing short of a technological masterpiece.

SONY CDP-701 COMPACT DISC PLAYER: Vital Statistics

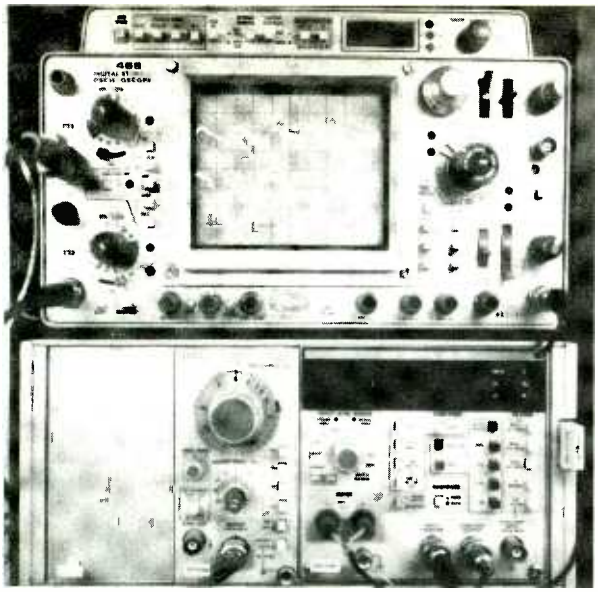
SPECIFICATIONS	MANUFACTURER'S CLAIM	LAB MEASUREMENT
Frequency Response	20 Hz-20 kHz, ± 0.5 dB	20 Hz-20 kHz, ± 0.4 dB
Signal-to-Noise Ratio	95 dB	100 dB, A-weighted
Channel Separation, 1 kHz	90 dB	86 dB
Harmonic Distortion, 1 kHz	0.003%	Confirmed
Wow-and-flutter	Unmeasurable	Confirmed
Audio Output Level	2.0 volts	2.13 volts
Headphone Output Level	28 mW/32 ohms	30 mW/32 ohms
Power Consumption	32 watts	30 watts
Dimensions	16.9" x 4.13" x 15.1" d.	Confirmed
Weight	25 lbs. 5 oz.	Confirmed
Price: Approximately $1500.00		

Circle 44 on Reader Service Card

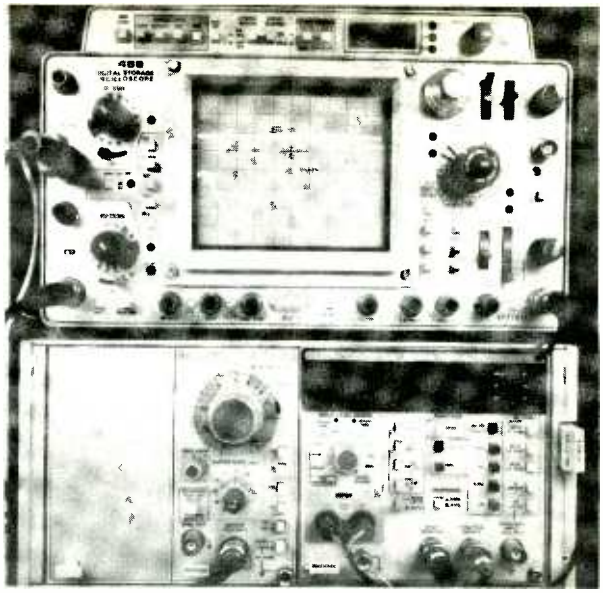
LT Sound CLX-2 Compressor/Limiter/Expander

General Description: LT Sound's CLX-2 is a compressor/limiter/expander that employs some of the most sophisticated circuitry and controls we have ever tested. It incorporates two significant features that are worth discussing in some detail. The first is what LT Sound calls a "ripple window." As they explain it, as the gain is changed in response to signal level, the compressor/limiter attacks when the signal volume needs to be lowered and releases when volume needs to come back up. This produces fluctuations in the gain control voltage and these fluctuations are referred to as ripple. Particularly in the presence of low frequencies, such ripple can produce high levels of distortion at low frequencies. The CLX-2's Ripple Window cancels out this ripple component during small gain changes, yet allows the unit to function normally for large gain changes. Since most musical signals aren't changing rapidly most of the time, the ripple window reduces distortion to less than one-tenth of that which would otherwise be produced.

The second innovation found in the CLX-2 acts to reduce other high-frequency gain control related distortions by as much as 20 dB and also reduces the side effects of noise gating with the expander section. This innovation is basically nothing more than a carefully tailored pre-emphasis/de-emphasis combination, not unlike those used in noise reduction systems for tape recording, but it is effective in reducing the distortion components in the mid- to high-frequency



(A)



(B)

Fig. 1. Without the two special circuits found in the LT Sound CLX-2, low-frequency signals are distorted at the output due to "ripple" effect, as seen in *1A*. With the Ripple Window and pre-emphasis/de-emphasis circuits, waveform is cleaned up, as shown in lower trace of *1B*.

region where they would be most audible. *Figures 1A* and *1B* were supplied by Lacy Thompson, Jr., the designer and owner of LT Sound. In *Figure 1A* we see a 50 Hz signal applied to a compressor with an attack time of 1 ms, a release of 300 dB/sec, a compression ratio of "Infinity," but with neither of the two LT Sound circuit innovations added. The upper trace is the ripple, while the lower trace is the distorted output, which measures over 10 percent under these conditions. Compare this with *Figure 1B*, in which all conditions are the same except that the LT Sound Ripple Window and pre-emphasis/de-emphasis circuits have been added. The distortion is now 0.163 percent, or less than one-fiftieth of its previous value.

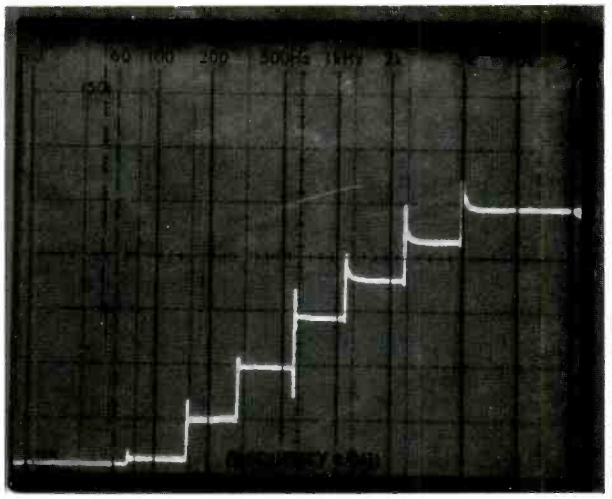
The CLX-2 is a two-channel compressor/limiter/expander with a great deal of flexibility in attack and release times and other functions. A better understanding of the versatility of the unit can be gained by considering, in more detail, the function of each of its many front-panel controls, most of which are duplicated for each of its two channels.

Controls & Switches: An on-off pushbutton for powering the unit is at the extreme left of the 19-inch rack-mountable front panel. A gain control adjusts the gain of the unit when the nearby IN/BYPASS switch is in the IN position. In that position, the switch will cause a nearby LED to light. THRESHOLD sets the input level at which compression or limiting starts to work, and this control is calibrated so that higher numbers will give you compression or limiting only on louder sections of program material. The switch labeled KNEE, when depressed, spreads the threshold over a wider volume range. The RATIO control determines the compression ratio above the threshold point. The larger the number, the more compressed the signal above the threshold.

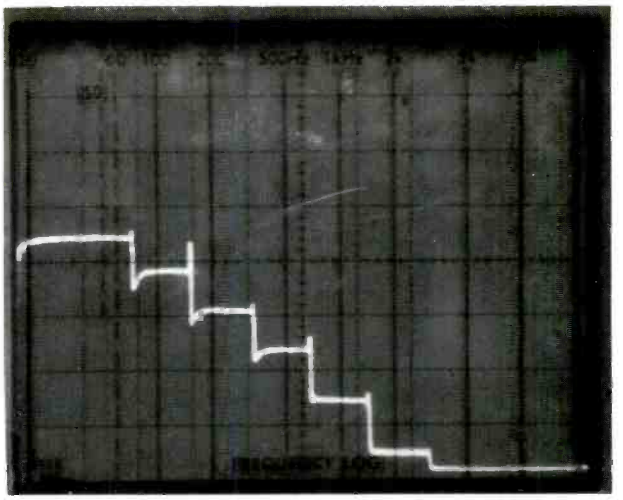
Technically speaking, the ratio is the number of dB you have to go above threshold to achieve a 1 dB change in output level. With this control set to infinite, no volume change at the output occurs at all above the threshold level, no matter how much you raise the input. A DE-ESSER control allows you to remove excessive hiss from vocal tracks (sibilance in "s" sounds) when they are individually processed through the unit. An ATTACK control determines the time it takes the compressor section to reduce the volume once the threshold level has been exceeded. Faster attack times produce less overshoot of volume, though they may sound more severe. Longer attack times produce a punchier sounding attack, but introduce a bit of overshoot. Five milliseconds has been found to be a good average setting for attack time, but obviously some experimenting is required for a given type of program material.

A control identified as DUCK allows you to reduce the volume of a guitar, for example, when a vocal is present. When the vocal is not present, the guitar track returns to normal original level. To achieve this effect, the instrument to be ducked (in this case, the guitar) is connected to the main input of the compressor while the vocal track in this case would be applied to an input labeled KEY.

A RELEASE control determines how quickly the lower volume signals are brought back to a higher volume after a loud passage has caused compression to take place. Higher numbers here represent a quicker recovery time, a louder sounding overall volume, and a more compressed type of sound. A switch labeled KEY, when depressed, causes the input signal to be expanded in accordance with the input signal present at the rear panel Key input, as already explained. The control identified as EXPAND determines at what level downward expansion is to take



(A)



(B)

Fig. 2. For these voltage "steps," Ratio control was set to Infinite, attack time was 0.1 millisecond, release time was a slow 2 dB/sec. The combination represents maximum compression, a fast attack time and a slow release time. *2A* is ascending; *2B* is descending steps.

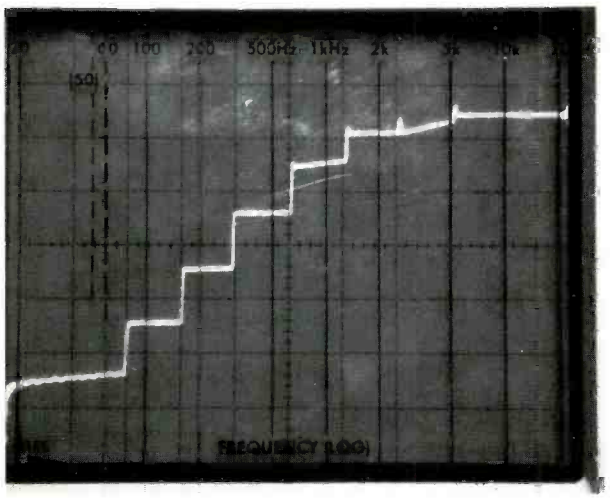
place. Higher numbers, such as -20, will turn the signal on and off at intermediate volume levels. Lower numbers, such as -60, will only expand the signal downward when its volume is very low to begin with.

A RANGE control determines how far down the signal volume is allowed to expand. Lower (more negative) numbers indicate greater volume decreases. This control, as well as the EXPAND control, works in conjunction with the Key and Duck functions.

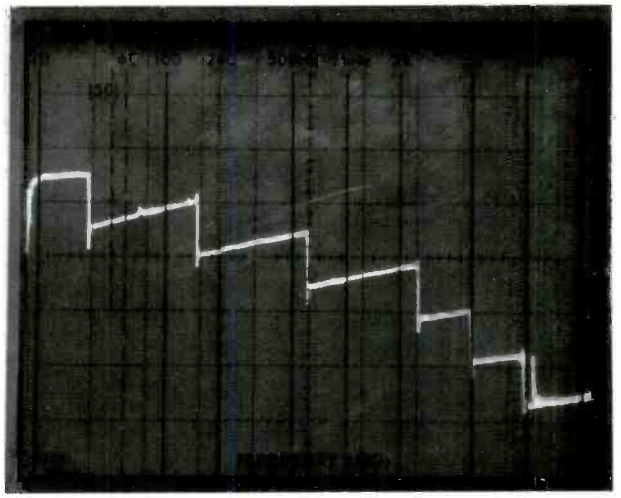
An LED display indicates the amount of gain reduction taking place in both the compressor/limiter and expander sections. A switch labeled DUAL/INDEP, when engaged, lights a nearby LED to denote stereo tracking mode. Stereo tracking would usually be employed during a final mix to preserve left-right panorama. The compression or limiting is applied such that whichever of the two signals is loudest will control the compression.

The rear panel is equipped with a balanced input jack for each channel, a balanced Key input jack, an unbalanced output jack and a jack labeled MOD, which is an unbalanced input that directly affects the voltage-controlled amplifier gain. Sensitivity of this control input is -20 dB per volt.

Test Results: As you can see from our brief table of VITAL STATISTICS, all of the more fundamental operating specifications were met or exceeded by our sample of the CLX-2 Compressor/Limiter/Expander, with the exception of the distortion spec. There, we measured 0.01 percent at 1 kHz for an output level of +4 dBv while LT Sound claimed a THD of 0.006 percent. Certainly the difference is not worth getting upset about! Of perhaps greater interest than the pure specs are the voltage step tests that we conducted using a spectrum analyzer set to a fixed frequency. Note that in *Figures 2, 3,* and *4* the frequency scale



(A)



(B)

Fig. 3. Here, ratio of compression is only 2:1, attack time is a slow 100 msec, while release is set to a maximum of 1000 dB/sec. *3A* is ascending voltage characteristics; *3B* is descending voltage steps.

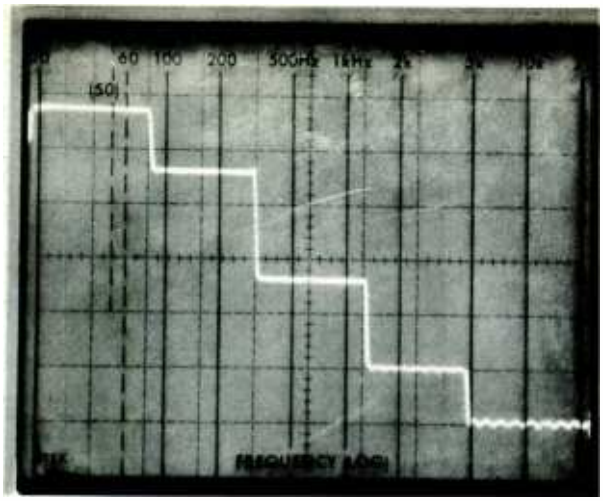


Fig. 4. Illustration of action of Expander of CLX-2. Downward expansion begins at point determined by EXPAND control on unit and ends at a point determined by setting of RANGE control.

inscribed on the tube face is to be ignored. We operated at a constant frequency of 1 kHz. Vertical sensitivity in each of these displays was 10 dB per division.

In *Figure 2A* we used an Infinite ratio setting, an attack time of 0.1 millisecond, and a release time of 2 dB/sec. These settings represent an example of maximum compression (above the threshold point), fast attack time, and slow release time. The upward voltage steps (10 dB change of input level per step), as seen at the output of the unit, are shown in *Figure 2A*. Notice that towards the right of the photo, the final two increases in input voltage result in virtually no futher increase in output voltage. In *Figure 2B* we show what happens if we start decreasing input voltage levels in 10 dB steps with the same control settings used for *Figure 2A*. The extremely slow release time is indicated by the tendency of each descending step to rise up to its previous level.

In *Figures 3A* and *3B* a ratio of only 2:1 was employed, along with a slower, 100 millisecond attack time and an extremely fast (1000 dB per sec) release time. Now we see evidence of some overshoot—in *Figure 3A*, the ascending steps of voltage—while in the descending voltage step pattern of *Figure 3B* each new voltage step downward is almost instantly stabilized at its final, desired value.

Figure 4 illustrates a particular downward expansion setting of the EXPAND control. The first step downward results in an output change of almost 10 dB (no downward expansion). The subsequent three steps downward show considerable downward expansion, while the final step is at a low enough voltage level so that downward expansion no longer occurs.

Comment: The most impressive aspect of the LT Sound CLX-2 was its extreme flexibility. I listened to a variety of unprocessed program material, feeding it through the CLX-2 and varying the controls of the compressor/limiter over a very wide range. The number of combinations was almost limitless. I was particularly impressed with the action of the de-esser function, which worked extremely well. I can think of some vocalists that I've listened to over the years who would have benefited greatly from the use of such a de-esser!

We were particularly impressed with the lack of distortion introduced by the unit, as well as by its ability to handle bass frequencies without introducing the familiar modulation effects commonly associated with less sophisticated compressors. While the Pro Net asking price of $895.00 for the CLX-2 doesn't put the unit in the "bargain basement" category (suggested list price is $1300.00), I suspect that to the small recording studio that can only afford one good stereo compressor/limiter, LT Sound's CLX-2 may well be the best answer.

LT SOUND CLX-2 COMPRESSOR/LIMITER/EXPANDER: Vital Statistics

SPECIFICATIONS	MANUFACTURER'S CLAIM	LAB MEASUREMENT
Gain	0 dB	Confirmed
Frequency Response	20 Hz to 40 kHz ± 0.5 dB	5 Hz to 100 kHz, ± 1 dB
Harmonic Distortion	0.006% @ 1 kHz, + 4 dBv	0.01%
Signal-to-Noise Ratio	90 dBv, "A" weighted	99 dB, "A" weighted
Dynamic Range	Better than 115 dB	119 dB
Slew Rate	13V/microsecond	Confirmed
Maximum Input	+26 dBv	+26 dBv
Maximum Output	+20 dBv into 2 kohms	-22 dBv
Output Impedance	47 ohms	Confirmed
Panel Dimensions	19" rack mountable x 1¾"	
Suggested Pro Net Price: $895.00		

Circle 45 on Reader Service Card

1/4 Notes

MAKING TRACKS

The **Eurythmics** are back in England, recording in a transformed 16th Century Church they purchased. Expected time of album release is January...**Hall & Oates** at **Electric Lady** in September to record two songs for a greatest hits package...**Grace Slick** recently finished an LP at the **Record Plant** with **Ron Neveson** producing...**John Cougar** is completing work on a new studio LP. The bulk of the tracks were recorded in a house under construction in Bloomington Indiana...At **Tres Virgos Studios: Stewart Copeland** of **The Police** produced the soundtrack for **Francis Coppola's** film *Rumblefish.* Stewart used a new computer device called a musync that maps out the film and records what is happening on each frame. **Johnny Cola** of **Huey Lewis and The News** is producing a new band called **Secrets.** Members include artists from **Tazmanian Devils, Norton Buffalo** and **Herbie Hancock.** Engineering the sessions are **Robin Yeager** and **Marian Lewis**...**Ronnie Montrose** is producing and engineering Bay area pop trio **Physical Ed.** Assisting at the board is **Gordon Lyon**...Currently at **Bullet Recording** in Nashville: **Jim Ed Norman** and engineer **Eric Prestige** are working on **T. G. Shepard's** upcoming album for **Warner Bros.** Producer **Garth Fundis** is wrapping up the mixing of **Russell Smith's** new album for **Capitol Records. Scott Hendricks** and **Phil Dihel** are assisting Garth at the board...Garth is also busy at **Sound Emporium** working with singer **Mac Davis.** They have resumed work on the album they began last spring for **Polygram Records**...**Luther Vandross** plans to begin producing his own solo LP in September...**Cheap Trick** is moving along with *Next Position Please,* their new LP produced by **Todd Rundgren** and recorded at **Utopia Studios**...L. B. Communications has been signed by **Peter Asher Management** and **Linda Ronstadt** to produce the premiere episode of *Albumflash,* a half-hour monthly Cinemax series focusing on new record albums by major recording stars...Recent recording activity at **Kendun Recorders: Mike Flicker** producing **Al Stewart,** with **Rolf Henneman** engineering and **Tim Dennen** assisting. **Crusaders,** with **Wilton Felder** and **Joe Sample** producing a new album for **MCA Records. Barney Perkins** is the engineer, with **Les Cooper** assisting. **Lonnie Simmons** is producing the new **Gap Band** album for his **Total Experience** label; **Mike Evans** and **Jim Stader** are engineering. **CBS** group **Third World** is doing a new LP with **Steve Stewart** and **Les Cooper** engineering...**Ronnie McDowell** finished up overdubs at **Soundshop Recording Studios** on his new album with producer **Buddy Killen** and engineer **Ernie Winfrey**...*Life is Something Special* is the debut album by the **NYC Peech Boys** on **Island Records.** The current single from the album is "Dance Sister," released in December due to radio's demand for the record. The album is produced by **Larry Levan** and **Michael de Benedictus**..."Mix Up, Mix Up" is the follow-up single to "Buffalo Soldier" from the *Confrontation* album by **Bob Marley.** The single is being re-mixed by **Shep Pettybone,** the inhouse master mixer at **WKRS** radio in NYC...Recent activity at **New River Studios: Atco** recording artists **INXS** doing audio post-production for an upcoming **MTV** special. The project was engineered by **Andrew Farris** and **Ted Stein.**

ON THE ROAD

Mick Fleetwood will begin touring in October or November to promote *Zoo,* his new solo LP... The **Little River Band** is resuming its American tour for six weeks beginning in September ....**Peter Gabriel** and recording artists **The Call** will be together on an upcoming tour of Europe from September 26 until November 15...**Prince** recently made an unpublicized appearance at a benefit in his home town of Minneapolis. Future plans for Prince include a film and soundtrack...The government of the tiny Caribbean island of St. Kitts has invited native daughter **Joan Armatrading** to provide entertainment at its first Independence Day celebration. Joan has just finished a five month world-wide tour...**George Russell,** a composer of modern jazz, will take his 14 piece **New York Big Band** on his first U.S. tour in October...Washington D.C. welcomes the fifth annual **New Music America Festival,** October 7-17. Highlights include: **the Philip Glass Emsemble,** Set and Reset, featuring **Trisha Brown/ Robert Rauschenberg/Laurie Anderson** and pianist **Anthony Davis**...**Windham Hill** recording artists **George Winston, Will Ackerman, Alex DeGrassi, Michael Hedges** and **Liz Story** are all embarking on tours beginning this fall.

& MUSIC...

ARETHA FRANKLIN: ***Get It Right.*** [Luther Vandross, producer; Michael H. Brauer, engineer; Michael Christopher, assistant engineer; recorded at Mediasound, New York, Record Plant, Los Angeles and United Sound, Detroit.] Arista 8-8019.

Performance: **Classy and resilient**
Recording: **Smooth**

The chemistry between Luther Vandross and Aretha Franklin resulted in the best r&b record of 1982: *Jump To It*, a bouncy romp that brought Franklin up-to-date with dance/soul conventions and provided the settings for her most relaxed, confident singing in years. Vandross, himself one of the most acclaimed singers of the Eighties, shrewdly produced Franklin from a singer's point of view, accommodating everything to her voice while keeping one ear tuned to the dance floor.

Get It Right sustains the momentum, and though it echoes *Jump To It* at times, *Get It Right* has slightly different song structures, arrangements, and a character all its own. The important thing remaining is the rapport between artist and producer. Vandross has such an understanding of Aretha's talent that every melody, every rhythmic phrase, every part of the mix is tailor-made for her, so that producer and artist can't help but click together.

The title track is a perfect example of how Vandross builds a sound environment for Aretha. The snappy bass (provided by Marcus Miller, who co-wrote four of the songs with Vandross), provides the backbone *and* the lead of the song, and its level is perfect—distortionless, nonsmothering and yet with plenty of kick. The keyboard and guitar embellishments follow in the same unassuming manner, letting Aretha take charge. She grabs the melody and ricochets back and forth with a lithe exuberance. The same playful spirit can be found in "Every Girl (Wants My Guy)," in which Aretha reaches for, and finds, a sly vocal in the lowest part of her range, and Vandross gets every last bit of resonance down.

Aretha's reflective side is also captured on *Get It Right*. She brings a soothing warmth to "When You Love Me Like That," which is state-of-the-art fusion/soul, only without the faceless vocal. Strings provide an elegant backdrop for some nice high notes on "Giving In," and the loping rhythm of "Pretender" builds with Aretha's vocal to a steamy rap that's as loose and self-assured as they come.

A remake of the Temptations' "I Wish It Would Rain" is the only major disappointment. As one of the three of four most gifted singers of her generation, Aretha has the right to cover any song she wants. But the promise of a slow-burn, gut-wrenching vocal goes unfulfilled. Too many instruments and background singers rush and crowd Aretha, forcing her to resort to chaotic, unlistenable histrionics. The other minor quibble is that the songs, four per side, run about two minutes too long, with unnecessary fills and chorus repeats; a little better editing would have made room for two or three more cuts.

But there will most likely be more collaborations between Vandross and Aretha because they have shown that *Jump To It* was no fluke. Vandross has proven himself again as an astute producer, able to keep Aretha's legendary talent unflinchingly contemporary; and she has proven that it's okay if they don't sing 'em like they used to, because they sing 'em like they do today.

rob hoerburger

NARADA MICHAEL WALDEN: ***Looking At You, Looking At Me.*** [Narada Michael Walden, producer; David Frazer, engineer; Ken Kessie engineer on "Never Wanna Be Without Your Love"; additional engineering by Wayne Lewis, Maureen Droney and Ray Pyle; recorded at the Automatt, San Francisco.] Atlantic 7 80058-1.

Performance: **Extremely uneventful**
Recording: **Earth-shaking dance floor dynamics**

Narada Michael Walden chose the Four Tops' classic "Reach Out (I'll Be There)" to lead off his seventh album, and he couldn't have made a better choice.

Whump! goes the kick drum at the start of the song, grabbing you by the chest. "Rrrrreach!" shout Frankie Beverly and Maze. "Now if you feel that you can't go on..." sings Walden lamely, and the song falls apart.

In fact, that kind of thing happens to every song on *Looking At You, Looking At Me.*

Walden is a great drummer but he is *not* a great singer. Unable to hold notes or hit high ones, Walden sings in a subdued manner, stripping highly emotional songs of their impact.

But there's nothing subdued about his drumming. He gets a fantastic

drum sound that you can both hear and feel. Ride cymbals ping cleanly, crash cymbals decay smoothly, evenly and without distortion. Snares and toms are always perfectly tuned. Excellent use of delay, compression and equalization on the kick drum produces a fierce sonic booming effect that could knock paintings off of a wall, or rattle the glasses off the tables at a disco.

Narada Michael Walden combines several of his talents into each of his records. As a songwriter, each album sees an improvement. He now fully integrates rhythm and melody, creating more soul. Arrangements are exciting, clever, and well suited to the songs.

As a performer, however, Walden has remained static. His performances satisfy, but never overwhelm. His voice is weak and restrained. His songs never reach a climax because his performance never hits a peak.

It is as a producer that Walden is at his best. His mixes are essentially perfect in terms of overall balance. All instruments are clearly located within the stereo image, and all retain their characteristic sound. A flugelhorn doesn't sound like a trumpet, a Fender Rhodes is a Rhodes is a Rhodes. With upwards of ten musicians playing on some cuts, it would be easy to lose some things in the mix. Yet everything is brought out for a full and clean mix.

Walden is also a master of effects. There's some of everything on *Looking At You, Looking At Me*, harmonizer, flanger, echo, reverb, you name it. But it's almost impossible, except in certain cases, to pick out the effects unless you're listening for them. Instruments are tastefully embellished by effects, not overcome by them.

But perhaps it is such a search for tastefulness that's making Narada Michael Walden's albums unexciting. There are no low points; nor are there any peaks. There's just a monotonous level of above average quality that never transcends into excellence.

steve jones

THE MANHATTANS: *Forever By Your Side.* [Morrie Brown, John Anderson, Steve Williams, Leo Graham, John McEwen, James Mack, Mark Blatte, Larry Gotleib, and George Tobin, producers; Steve Goldman, Steve Adaboo, Chuck Ange, Stu Walder, Mike Ferraro, Hal Hellerman, and Les Brockman, engineers; recorded at Celestial Studios, New York City, Studio Sound Recorders, Hollywood, CA, Universal Recording Studios, Chicago, IL; mastered at Frankford Wayne Labs, New York City, by Herb Powers.] Columbia 38600-1.

Performance: **Polished and richly textured**
Recording: **Crystalline**

Riveting four part harmonies, captivatingly sensitive lyrics and slick productions have long been trademarks of The Manhattans, a vocal group from New York City. Like so many groups that got their start when they did—nearly thirty years ago—the foundation of The Manhattans' smooth style was laid while singing doo-wop on local street corners. Many of the groups they did battle with back then have not survived. They, however, did.

Although The Manhattans have been singing for almost three decades, it was not until 1973, after signing with Columbia Records, that the group acquired international attention. The song that set the wheels in motion was "There's No Me Without You," their first gold record. What followed were more hits, many of which became classics. Among them are: "Kiss And Say Goodbye" (a platinum record), "It Feels So Good (To Be Loved So Bad)," and "I'll Never Find Another Find (Another Like You)."

Their latest album, *Forever By Your Side*, is a reaffirmation by one of the world's great vocal groups. Its production was split among three

sets of producers—Morrie Brown (known for his work with Evelyn King and Kashif) and his four assistants, George Tobin (known for his work with Smokey Robinson), and the team of Leo Graham and James Mack, who wrote and produced The Manhattans' Grammy award-winning smash "Shining Star."

The songs, whether uptempo bouncers or easy ballads, have a combo-like sparseness to them, which can be attributed to impeccable equalization and balancing among the various instruments. Throughout, one is able to keep track of what the keyboard player, horn players, string players et al are doing. The opener and first single, "Crazy," may, on the surface, seem like just another r&b dance song, but when Sonny Bivens, Gerald Alston, Blue Lovett and Kenney Kelley embody it with their brand of singing—close harmonies generated by sweet melodies—it is lifted to another dimension. A fluid bass synthesizer line heard throughout this song gives it a tremendous sense of buoyancy, but you'll want your feet firmly planted on the ground for this one is a cooker.

Some of the tunes on *Forever By Your Side* sound like previous Manhattans' material. "Start All Over Again," for example, sounds strikingly like "Kiss And Say Goodbye," and "Love Is Gonna Find You" recalls "I'll Never Find Another Find (Another Like You)." Unexpected melody and harmony changes, however, give these songs their own special character. On the title track, with the bass, drums, piano, and strings mixed low, The Manhattans' voices become the prime focus of attention; they blend them in perfect consonance. Now *that's* singing!

havelock nelson

RICKIE LEE JONES: *Girl At Her Volcano.* [Rickie Lee Jones, producer; Mark Linett, engineer; recorded at Jennifudy Studios, L.A., The Roxy, L.A., Perkins Palace, Pasadena, CA.] Warner Brothers 23805-1 B.

Performance: **Quirky yet endearing**
Recording: **Smooth as velvet**

My logic must be failing. Here is a ten inch (like those old jazz records of the pre-LP days) EP of old jazz standards sung in the most eccentric phrasing possible by a non-jazz singer. I can think of a dozen purely *logical* reasons to reject this recording immediately. What is a singer/songwriter in the rock genre doing releasing a record with "My Funny Valentine"? And "Lush Life"?! And mixing these standards with that rock classic from the Sixties, "Walk Away Rene"? What is going on?

What is happening is that Rickie Lee Jones is in the process of returning to the roots of her singing/songwriting. And this eccentric mash of jazz and vintage rock is an accurate reflection of her past and present musical loves.

A key toward comprehending the performances on *Girl At Her Volcano* can come by comparing it to a seminal record by another woman singer/songwriter, Laura Nyro. After recording a few stunning original albums of her own music, Nyro recorded *Gonna Take A Miracle*, a jivey and street-wise collection of early Sixties rock with a heavy accent on Black soul music. The record received extremely mixed reviews and sold poorly. Had Nyro run out of ideas for original songs? Why the nostalgic censura?

Nyro has since continued her recording career with several more albums of original songwriting. And now her recasting of Sixties covers sounds not like a swan song of a fading inspiration but rather a reaffirmation of musical roots. A journey to the past with a singular and even visionary intent: to personalize songs in the rock canon in order to freshly rehear their resonances.

All of the comments about Nyro's *Gonna Take A Miracle* apply totally to this Rickie Lee Jones release. After all, they are extremely close in spirit and substance: Jones' version of "Under the Boardwalk" sounds nearly like an outtake from Nyro's album; listen to those emphatic piano chords on "Walk Away Rene"; note the wobbly melodrama a la Nyro on "Lush Life." And her occasional lapses in articulation, again, like Nyro, make you wish for an accompanying lyric sheet.

But also hear all those quirky yet endearing moments that make Rickie Lee Jones a stunningly individual talent: the bittersweet breathless tone on "Rainbow Sleeves"; the controlled sentimentality in her singing on "Hey, Bub."

Here's this oddly sized record of seven songs clocking in at a little over twenty minutes that makes me overwhelmingly appreciative of Jones' singing six covers of material totally unsuited to her usual vocal range and sensibility. Yet she manages this triumph. It is as if Nina Simone released a live album of gut-busting arias from Wagner. Will wonders never cease? Will her next recording set Malcolm Lowry's classic novel *Under the Volcano* to jazz?

Warner has nicely backed this strange project with a ravishingly smooth sound. Surface noise is exceedingly low thanks to their "Quiex II" processing. Rarely have I heard live performances captured with such well-balanced brightness. Ace engineering for a first-rate act.

This recording will shock Rickie Lee Jones' fans who were expecting a continuation of the sound established of her first two albums. It will terribly upset jazz fans of a purist bent. It will probably receive little air play from rock stations, and jazz stations won't know how to present it.

That's one way to judge original and lasting music: by its refusal of easy classification. In a music industry replete with fakes, Rickie Lee's the real thing. A torch singer tapping the fire and light of her interior volcano.

norman weinstein

PHILIP GLASS: *The Photographer.* The Philip Glass Ensemble. [Kurt Mankacsi, Philip Glass, producers; Kurt Munkacsi, Michael Riesman, engineers; recorded and mixed at Greene Street Studios; mastered at Master Disk by Bill Kipper.] CBS FM 37849.

Performance: **Beautiful**
Recording: **Carefully molded, first rate.**

The Photographer is a music/theater piece composed for the 1982 Holland Festival about the British-born photographer Eadweard Muybridge. In 1874, Muybridge, who was then working in San Francisco, discovered that his wife was having an affair with one Colonel Larkyns. The photographer shot Larkyns and, subsequently, was acquitted at his murder trial. In fact, although his name and samples of his work (particularly his nature photography) can still be found in photography reference works, the incident between him and Larkyns seems to have slipped quietly into history.

Other than in the liner notes, we don't learn much about this incident from Glass's recording of the piece—partly because the LP presents only the audio, and not the visual, aspect of the work. Even so, as anyone who has seen either of Glass's operas *(Satyagraha* and *Einstein on the Beach)* can attest, Glass's interest is in non-narrative theater; in his theater works, it is the feelings that surround the narrative, rather than the narrative itself, that are captured and presented. In the case of *The Photographer*, the first act is a play about

the Muybridge-Larkyns incident, and the subsequent trial, to which Glass contributed three short incidental pieces. Two of these—a bittersweet and rather attractive choral song, "A Gentleman's Honor" (with lyrics drawn from the trial transcript by David Byrne, of the rock band Talking Heads), as well as an instrumental version of same—are included on the LP; the third, a dance, is absent.

The meat of the disc, however, comes from the second act, which is presented as a concert in its staged version, and the third act, a dance piece involving the characters from the play in the first act. Without the visual side, we obviously lose something of the work's impact, but, taken as abstract music, these pieces stand up nicely and are easily as captivating as Glass's earlier and more truly abstract explorations of tonality and rhythm. Glass fans who were disappointed with the collection of uncharacteristically short, melodic pieces that made up his last LP, *Glassworks*, will probably be better satisfied by *The Photographer*.

Anyone with even a glancing familiarity with Glass's music will recognize elements of his unique compositional voice here: the arpeggiated themes, the repeated phrases, and the gradual building of textures during these repetitions, all culminating in the dense rhythmic webs Glass likes to create. What is most striking in *The Photographer*, however, is how far Glass has managed to take this style, and how sophisticated his manipulation of these seemingly simple elements has become. His penchant for slow change has given way to a considerably greater density of "events," and even to rapid shifts of themes and textures. In the third act music, for instance, the characteristic opening arpeggiation soon gives way to an electrifyingly fast gallop for full ensemble—hardly in keeping with the soporific image Glass's detractors have attached to his work; but this tornado-like movement is suddenly interrupted for brief a cappella vocal interludes that lead to other material.

Similar devices are used in the second act music, in which Paul Zukofsky's tensely played, arpeggiated chord progressions are set against an organ sequence built on the first two chords of Beethoven's "Moonlight" Sonata (which later come back in blocked form, sounding more like the old ? and the Mysterians

In a Mellow Groove: Alvin Queen and Maxine Sullivan

nat hentoff

A jazz collector I know, who hears just about everything that comes out because he's compulsive, was trying to describe Alvin Queen's *Glidin' and Stridin'* (Nilva Records). "It's like one of the early Blue Notes," he said, "and you know, hardly anyone makes jazz dates like that any more."

He's right. In its early years, some four decades ago, Blue Note was a jazz label that specialized in dates that had the unhurried feel and spaciousness of after-hours sessions. The players stretched out as long as they felt good about the results. And so it is in this new release by drummer Alvin Queen (a Horace Silver alumnus), pianist Junior Mance, and bassist Martin Rivera.

Although there's ample improvisatory space for Queen and Rivera, the imaginative center of the set is Junior Mance. He's played with a wide range of talents and temperaments—Lester Young, Gene Ammons, Dinah Washington, Dizzy Gillespie, Cannonball Adderley, et al—but in recent years he's been much too infrequently recorded. Here, the fullness of Mance's experience and the authoritativeness of his time, touch and spirit combine in a series of deeply satisfying distillations of a life in jazz.

The engineering is unobtrusively skillful, but I would have wished for a somewhat less bright piano sound. This is a very minor flaw, however, in an album that will give and sustain pleasure so long as we're not automators.

Also of perennial value is *Maxine*, a Holland session by Maxine Sullivan with Ted Easton's Jazzband, recorded in 1975 and now released in America on Audiophile.

As an admirer of the lithely swinging, crisply articulate, and highly economical singing style of Maxine Sullivan, I have been waiting for years for an album by her that equals the delight of her "live" performances, including her radio broadcasts of yore. At last, this is it!

For one thing, the tasteful, unpretentious Dutch jazz band behind her is clearly content to accompany Maxine rather than compete with her for attention in the front line. Accordingly, secure in the gentle flow of the players in the background—with occasional, deftly apt commentary by the horns—Maxine can just enjoy and be herself.

The result—from "Someday Sweetheart" and "I've Got The World On A String" to "Wrap Your Troubles in Dreams"—is one of the classic definitions of jazz singing. Her voice is like a horn but does not try to be a horn, because she also cares about illuminating lyrics. And Maxine's beat is so supple, so subtle, and so sure that it's a continuous source of gentle exhilaration. This is a set to put aside as a gift for a jazz aficionado who has almost everything.

The quality of sound is first-class, starting with a perfectly balanced rhythm-section sound whose natural tang is reflected in the true sound of the horns and of Maxine herself.

ALVIN QUEEN: *Glidin' and Stridin'*. [Alvin Queen, producer; Ron Carran, engineer.] NILVA RECORDS NQ 3403, distributed by New Music Distribution Service, 500 Broadway, New York, NY 10012.

MAXINE SULLIVAN: *Maxine*. [Ted Easton, producer; Joop Pieete, Sr., engineer.] AUDIOPHILE AP-167, 3008 Wadsworth Mill Place, Atlanta, GA 30032.

hit, "96 Tears"). The violin plus stretches of wordless choral accompaniment and bass synthesizer and trombone lines create an eerie, Gothic atmosphere.

The production is first rate, and boasts a clean but nicely ambient sound in all registers, and a comfortably broad stereo image. As detailed in *MR&M's* Glass/Munkacsi interview a few months back, this large ensemble work, like all of Glass's recordings, was created by layered multitracking, one instrumental group at a time. Even so, the reconstructed performance has a vibrant, "live" feel to it that lacks nothing in cohesiveness, energy, and even the illusion of ensemble interaction. CBS's pressing is clean and quiet. Allan Kozinn

PETER TOSH: *Mama Africa.* [Peter Tosh, producer; Lancelot McKenzie and Geoffrey Chong, engineers; recorded at Dynamic Sound Studios, Kingston, Jamaica.] EMI Records SO-17095.

Performance: **Enthusiastic and committed**
Recording: **Compressed yet unhampered**

This is Peter Tosh's finest album since his *Equal Rights* album on Columbia many moons ago. The reason? Unusually fine melodies, exquisitely complex and thoughtful arrangement, vibrant backing musicians, pithy lyrics. This is Tosh's "comeback" album, and for those of you who haven't followed Tosh's career, a review might be in order.

Peter Tosh was one of the original Wailers, the most commercially successful group in the history of reggae. Tosh left the Wailers to Bob Marley in the early Seventies and pursued a solo career with varying degrees of success. Two albums were issued by Columbia: *Legalize It* (a reference to Tosh's relationship to marijuana) was lackluster and inconsistent due to the fact that Tosh's vocals bordered on comatose rumblings; *Equal Rights* found Tosh singing with greater spunk and drive about political conflicts through the eyes of a committed Rastafarian.

Then Tosh met the Rolling Stones and ended up releasing a number of uneven records on their label. Tosh faced accusations similar to those faced by fellow Jamaican Jimmy Cliff: He was accused of "selling out" to the American rock industry for big bucks.

Mama Africa should quickly put an end to rumors of Tosh abandoning reggae for rock. True, there is a version of "Johnny B. Goode" on this album, but it is a charmingly eccentric remake of Chuck Berry's classic. The Johnny in Tosh's song is one tough leader of a reggae band and the beat of the song confirms the lyrical transposition in time and space. The title cut, "Mama Africa," blends African polyrhythms (brilliantly supplied by percussionists Scully Sims and Sticky Thompson) with a reggae backbeat supplied by drummer Carlton Davis and bassist Lebert Morrison. Tosh sounds positively ecstatic as he croons to his mother Africa, making her sound simultaneously like a symbol of Rastafarian freedom and a literal woman to be romantically embraced.

Song lyrics cover a spectrum of topics ranging from the topically political ("Peace Treaty") to the Biblically timeless ("Where You Gonna Run"). Several of his songs sound like simple Old Testament parables set to hummable tunes ("Glasshouse" among them). Tosh's indebtedness to former Wailer singer and composer Bunny Wailer is particularly evident in those songs with a spiritual feel.

Arrangements are stunning with exciting horn charts by one of Jamaica's finest composers, Clive Hart. Female backing vocals—while not up to the stellar standards maintained by Bob Marley's "I-Threes"—are supportive of Tosh's efforts.

The recording is dense with tracks on top of tracks on top of...yet each voice and instrument maintains its distinct voicing in the final mix. Tosh's vocals are not as dominant in the mix as they were in his Columbia days, but he never sounds buried in an unseemly hash of horns or female backing singers. His talent, both as vocalist and producer, looms largely over this recording.

Give this record to a friend who thinks all reggae sounds the same. *Mama Africa* will unclog his or her ears. As the Rastafarians would say: This is a *roots* recording; a soundtrack for finding one's way back to an Africa spiritual imagination knows best by its rhythms. norman weinstein

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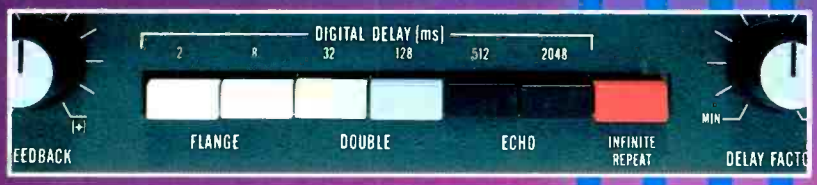

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