Profile: JOAN ARMATRADING

MODERN RECORDING & MUSIC

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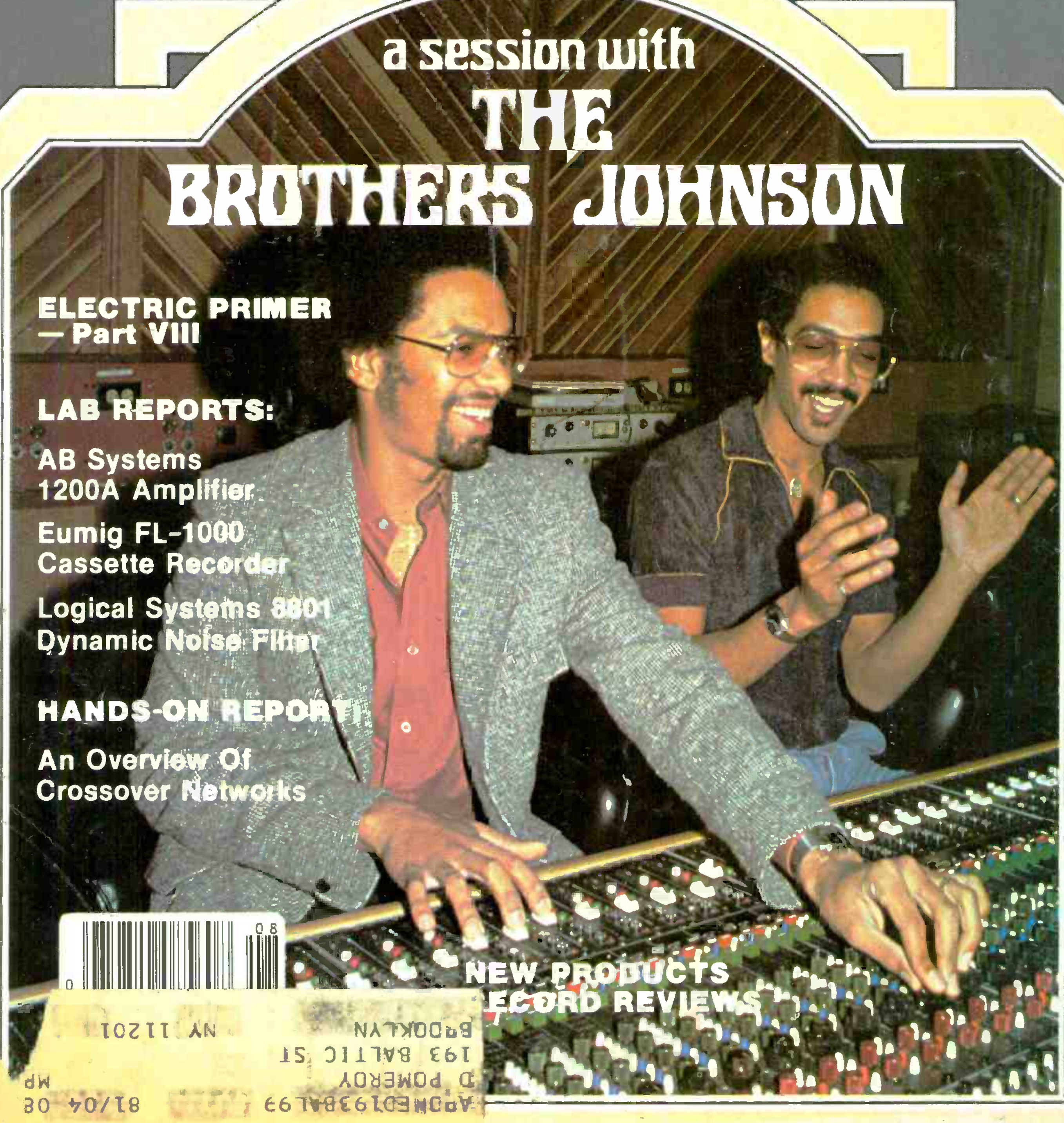
AUGUST 1980



a session with THE BROTHERS JOHNSON

ELECTRIC PRIMER — Part VIII

LAB REPORTS:

AB Systems 1200A Amplifier

Eumig FL-1000 Cassette Recorder

Logical Systems 8801 Dynamic Noise Filter

HANDS-ON REPORT:

An Overview Of Crossover Networks

NEW PRODUCTS

RECORD REVIEWS

How serious are you about a synthesizer?

Even if you don't have an unlimited budget, you still want virtually unlimited expression from your synthesizer. Yamaha knows this. We also know what kinds of features and functions give you this expression. The musicians who evaluated our prototypes told us. And we listened. The result is a new line of affordable synthesizers from Yamaha built especially for live performances. They are capable of many of the sounds of our larger, costlier models, and have all of the quality and reliability.

CS-40M. Duophonic, programmable and highly portable describes this top model in the new line. It has four VCO's, two VCF's and two VCA's plus a Ring Modulator, an Attack/Decay EG for the LFO and Ring Modulator, and a unison mode which converts the unit to mono operation by doubling up the VCO's for richer sound. The keyboard has 44 keys.

The CS-40M can store and recall, at the push of a button, up to 20 sounds that you've created, even after the power is shut off. Interface with a tape recorder requires just two patch cords.

CS-20M. Up to 8 voices can be stored and recalled in this model. The CS-20M has two VCO's, an LFO, a noise generator, a mixer (for the VCO's and the noise), a 3-way VCF and a VCA. It is a monophonic instrument with a 37-note keyboard.

Both models have keyboard trigger in/out jacks and control voltage in/out jacks for convenient use with a sequencer. Rear panel jacks are provided for ON-OFF foot switching of Sustain and Portamento/Glissando effects, and for foot-pedal control of the filter and volume.

CS-15. This compact, very affordable synthesizer has two VCO's, two VCF's, two VCA's, two EG's and one LFO. One-touch knobs and switches free you from complicated patch work. Sawtooth wave, square wave, white noise, and triangle wave give unique tonal characteristics.

MODEL	KEYS	VCO	VCF	EG	NOTES	DIGITAL MEMORIES
CS-5	37	1	1	1	1	N/A
CS-15	37	2	2	2	1	N/A
CS-20M	37	2	1	2	1	8
CS-40M	44	4	2	2	2	20

CS-5. This is our most compact monophonic synthesizer. It has 37 keys, but with the 6-setting Feet selector switch, the instrument's range is extended to a full 8 octaves. A Sample and Hold circuit allows you to automatically play a continuous random pattern. There are many other features that make this model's very affordable price even more attractive.

For more information on the full line, write: Yamaha, Box 6600, Buena Park, CA 90622. (In Canada, write: 135 Milner Ave., Scarb., Ont. M1S 3R1.) Or better yet, visit your Yamaha dealer for a demonstration of the synthesizers that take both your creative desires and your budget considerations seriously.

Because you're serious.

YAMAHA

CIRCLE 99 ON READER SERVICE CARD

Studiomaster is Expanding

We have never been ones to sit around and watch others progress. nstead, we prefer to be the leaders. And in our field, that is how nany regard us. So we apply our energy to expansion . . . growth in very phase of the development of our products.

Our mixers are expandable. *Studiomaster* was first with the add-on nodule concept in affordable professional grade mixers. Whether you elect our 12X2b or 16X4 units, neither will ever limit your group or tudio to its original capacity. By our four-channel-at-a-time expander nodules, a 12X2 can become a 24X2 in minutes. A 16X4 will change nto a 20X4 almost immediately, and without the hassle of dangling ables that can cause shorting or transporting difficulty.

Our features are expanding. As pioneers in the more-feature-per-lollar area, *Studiomaster* was the first to offer 5 way equalization on n affordable mixer. We still are. Our 16X4 mixer was the first studio uality desk that allowed the operator to have a separate stage (or udio) mix for the performers while still sending an independant, naffected mix to the tape recorder. No competition yet. And now, ur mixers come with direct in/out channel patching as standard quipment, just one example of our expanded features.

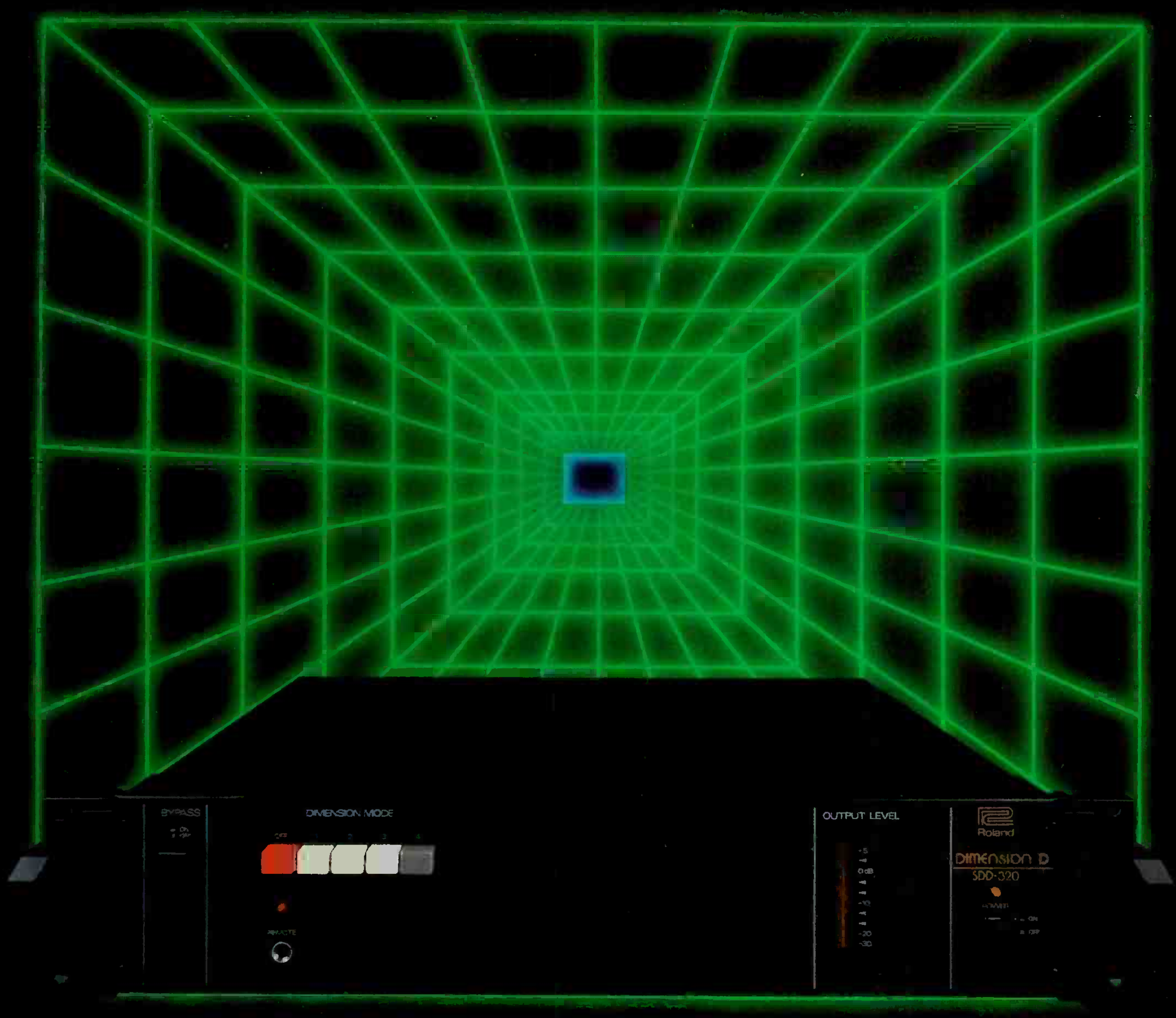
Our product line is expanding. *Studiomaster* now has a 16X8 mixing console designed for 8 track studios and live performance. We offer the professional touring band the most affordable 20X8 monitor mixer available. And we now manufacture three and five way stereo crossovers which solve the complex speaker system problems of large concert arenas. We even make the coolest running, smoothest sounding 225 watt/channel amplifier around, too. Did we say we're expanding?

The only thing that really hasn't kept pace with our other phases of expansion is our price tag. It has escalated only a fraction in two years . . . and that only to justify the extras we offer. So we are still the most affordable 16X4 mixer that money can buy in our performance category.

Maybe you should consider expanding your studio's or group's ability to create. Give us a call at (800) 854-6219 nationally or (714) 528-3360 in California. We will give you the name of the *Studiomaster* dealer closest to you. If you visit him this month for a *Studiomaster* demonstration, you can receive a *Studiomaster* T-shirt free from us. Visit him soon to see the continually expanding possibilities of *Studiomaster* products.

For more information about Studiomaster products, please write to Craig Bullington, National Sales Manager, Studiomaster, Box 55, Atwood, California, 92601.

CIRCLE 83 ON READER SERVICE CARD



The Dimension D

The problem with many special effects is that they have a tendency to become specialized effects. Rather than being integral to a performance. They become simply the frosting on the cake, used only occasionally because their very nature makes constant use tiresome.

But what if the fundamental sound itself is lacking? What if you want to give it some life without having to put it through a wringer for improvement? For you Roland offers Dimension D—the Uneffect.

The Dimension D changes the life of your sound without changing the sound itself. Electric and electronic musical instruments are enriched with a depth and liveness that they never had before. . . fundamentally the same, but deeper and richer. Likewise, acoustic instruments and vocals are richened and thickened when you amplify or record them. They're given back the brilliance they had live.

The Dimension D is not a device that will be relegated to sitting in a bank of equipment, rarely used. On the contrary, the Dimension D is likely to become as integral to the creation of sound as any other element. Once you turn it on, you won't turn it off.

The Dimension D can work on either a mono or a stereo signal, and also has the capability to generate a stereo field. And if you don't want to turn the Dimension D off, you won't have to, because its signal-to-noise ratio is better than 90 db.

The Dimension D is not an effect, it's really an experience. It won't blow you away—it isn't supposed to. That's because some effects are not measured by their intensity, but by their subtlety.

Enclose $1.00 for a copy of the Roland Rack catalog.

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(213) 685-5141

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We Design the Future

CIRCLE 144 ON READER SERVICE CARD

AUGUST 1980
VOL. 5 NO. 11

MODERN RECORDING & MUSIC

THE FEATURES

COMING NEXT ISSUE!
A Session with Smokey Robinson
Profile: Guitarist Pat Travers
Plus more!

Cover Photo: Lynn McAfee
Session Photos: Lynn McAfee
Armatrading Photos: Courtesy Howard Bloom Organization

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Letters to the Editor

Beloved Rogue

Thank you for the Joe Klee review of the Carmen Leggio album in the March 1980 issue of *Modern Recording*. It's always a pleasure to see one's name in print and since Mr. Klee usually likes my work, it's doubly pleasurable.

Some questions and comments for Joe, though:

I was curious about your phrase "well known tendency to overmic the drummer." First, do you mean just too many mics, or do you mean the drums are too loud on the record? And secondly, how well known is this tendency, in fact?

Lastly, you might notice that different recordings of mine sound different depending on the producer of the date. My records (produced by me), like Gil Goldstein's, which you reviewed very favorably, sound completely different from a Hank O'Neal (although he gives me amazing freedom) or a Gus Statiras (Progressive) or any of the other producers I've worked with. Realize that they all have their particular preferences for balance. One will like more drums, another no echo, another forbids overdubbing, and yet another insists on total isolation of instruments! An engineer is, after all, just there to make the producer's sound a reality. As much as we do have a certain amount of control, I feel that a lot of reviewers place a bit too much emphasis on the engineer's role in determining particular sounds and balances.

Of course, this is not to deny my vast experience in jazz recording (over 350 albums), nor my impeccable taste, good looks or charm, but you must realize that there's no autonomy in the control room unless it's the producer's.

So write and explain yourself, you rogue! Cordially,

—Fred Miller
Studio Registry
New York, N.Y.

Joe Klee asked us to respond for him: He agrees with you—he did in fact overstate the case for the engineer's role. The bit about tending to overmic the drummer, on retrospect, is a moot point considering that the producer and the artists were the same in the instances he was referring to your work and "well known tendencies." Forgiven? Say yes; Joe's a swell fellow.

Review Review

I would like to make a suggestion concerning your record reviews. There must be at least a dozen or two publications where one can read a review of the artistic and commercial merits or demerits of a record. Yours is the publication that critically touches on the engineering and production strengths and weaknesses. My suggestion is that the reviews give us more "meat" concerning production and engineering techniques, and less on the reviewers' likes and dislikes. Slanting reviews more towards the tone of your "A Session With . . ." articles would be more constructive for the kinds of readers

your publication attracts. Dissecting the arrangement vertically and horizontally as to the engineering and production "tricks" would be most helpful. I further suggest that the reviewers at least phone interview the engineers, the producers and the artists to verify what is heard on the record.

For example: I would like to know the equipment and technique used to get, say, the lazer shots on Dolly Parton's "Baby, I'm Burnin'." On the other hand, I don't really care to know the reviewer's opinion on how far he thinks Dolly has strayed from her Country roots. I can get this kind of information from all the other reviews. From you I want to learn how what I hear was constructed, and the recording quality.

I make this suggestion only to improve further an already excellent publication. Sincerely yours,

—Hank Strasser
Reality Productions
Montclair, N.J.

Point well taken. Our reviewers have been asked to indulge less of their own tastes and attend more consistently to the recording, rather than to the subjective qualities of new releases.

But we must argue that your suggestion of phone interviews with engineers, producers and artists, while ideally sound, would pose a few problems. First off, "Groove Views" would necessarily expand. And while we know our readers enjoy reading the reviews, we like to concentrate our efforts less on critical review than on dynamic involvement in recording arts. On another level, engineers, artists and producers are hard enough to track down for in-depth interviews such as those we tackle in our "Profile" features. Additionally, phone costs, road tours and new projects, although we may be mistaken, would probably be prohibitive. It is a great idea, though, and if wishes were horses . . . .

Colorsound

I bought your Winter '80 *Buyer's Guide* and found Colorsound effects devices listed under Scanlan Music Sales. I wrote to Scanlan about Colorsound effects devices and they sent me a letter saying the Colorsound line of effects is no longer available in this country.

Can you send me the address of Colorsound or a distributor that handles Colorsound? I believe the company is located in England. Thank you.

—Dave Wharton
Fairbury, Ill.

The people at Scanlan Music Sales (at 15 E. Park Blvd., Villa Park, Ill. 60181, phone (312) 833-5464) tell us that they are trying to work out distributorship for Colorsound products anew and they hope to be handling the line again soon. In any event, their spokesman suggested you contact them once again: if you have a particular item in mind, perhaps they will be able to special order it for you.

Fan Dance

First, I would like to say "thank you" for the best magazine in the field of music and music reproduction. In the February 1980 issue, by the way, I was overwhelmed by the review of Spyro Gyra's *Morning Dance* album. At my first opportunity, I made the three-

Technics

Performance and reliability. That's why 73 of the top 100 radio stations that use turntables use Technics direct drive turntables. In fact, of those stations surveyed by Opinion Research Corporation, Technics was chosen 6 to 1 over the nearest competitor.

Why did station engineers choose Technics direct drive: "Latest state of the art." "Reliability and past experience." "Low rumble, fast start." "Wow and flutter, direct drive and constant speed." To quote just a few. And you'll choose Technics for the same reasons.

The D-Series. Three turntables that start at $125.* Each with 0.03% wow and flutter and −75 dB rumble. That's unsurpassed performance for the price.

The Q-Series. Two turntables with speed accuracy of 99.998%, wow and flutter of 0.025% and −78 dB rumble. No wonder so many radio stations choose Technics quartz-locked turntables.

The MK2 Series. Three turntables with a startup time of 0.7 seconds, or ¼ of a revolution. And the accuracy of Technics quartz-locked pitch control. That's performance good enough for the most demanding professional.

Then there's the SL-10 ($600). The turntable of the 80's. Not much bigger than a record jacket, it has a quartz-locked direct-drive motor and a servo-controlled linear tracking arm. But what really makes it unique is what it can do: Play with all the accuracy Technics is so famous for, even on its side or upside down.

Technics direct drive. The turntables that top radio stations use.

* Technics recommended prices, but actual price will be set by dealers.

You should buy a Technics direct drive turntable for the same reasons 73 of the top 100 radio stations did.

CIRCLE 98 ON READER SERVICE CARD

hour drive to see (and hear) them at the Front Row Theatre in Cleveland. To say the least, I was thoroughly impressed with the show. Many thanks for the insight on Spyro Gyra.

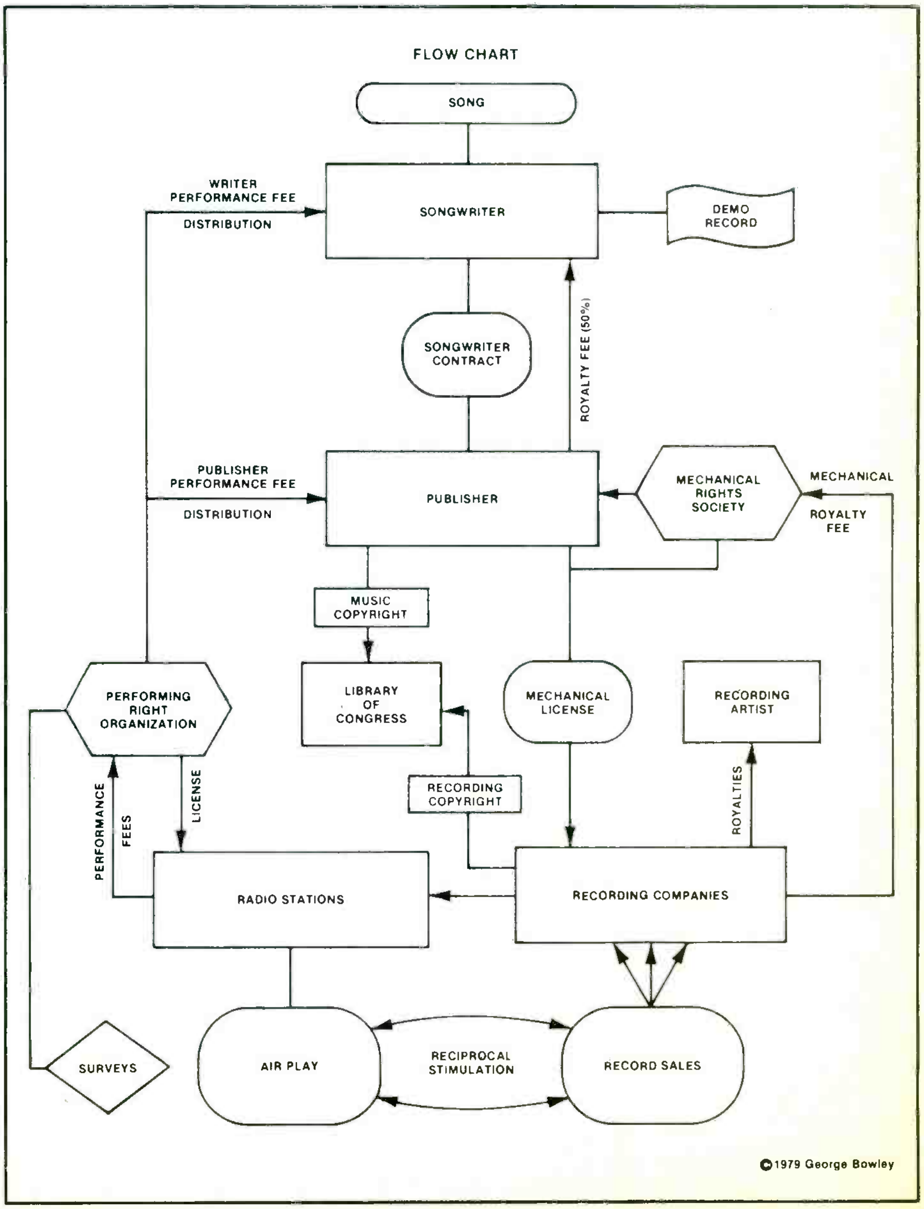
—Don Imlay
Lanesville, Ohio

Flow Chart Art

I enjoyed the opportunity to contribute to your magazine last year, and I thought your readers might also be interested in the accompanying graphic. It was developed for inclusion in my prospectus for a studio equipment upgrade for the purpose of educating the lender in some of the basic relationships and major money flows in the music business (from the songwriter/publisher viewpoint).

My intent was to illustrate 1) the role of the publisher as the songwriter's primary interface with the rest of the industry, 2) the income generated by performing right organizations through surveys, which is equally shared by the songwriter and publisher, 3) the income generated from record sales through mechanical licenses, which is shared equally by the songwriter and publisher, 4) the protection of rights through the copyrighting of both music and sound recordings, and 5) the frustratingly factual "Catch-22" iterative process in which record sales stimulate air play, and air play stimulates record sales.

The chart is very basic. It does not address all facets of the business, nor all the fees and monies that change hands. It does not reflect a publisher's



interfaces in promoting, licensing and administering its catalog, nor does it identify a recording studio's expenses connected with studio time, musicians, arrangers, producers, pressing, distribution, promotion, and the like. If all relationships and economic factors were addressed, the chart would resemble a spider's web. The intent was to show only basic relationships.

Keep up the good work on your excellent magazine. You walk a very fine line between educating the completely uninformed and giving the seasoned professional something to read also. You do it beautifully.

—G. A. Bowley
President
Sync Records
Springfield, Va.

A Long Note from Shortstop

Just wanted to thank *MR* and Ken McKinn of RPM for his response to my letter printed in the April '80 issue. I don't feel like I'm alone in left field anymore (seems like shortstop now).

Based on what Ken had to say, I opted for a dbx 155, and for someone who has never used one before, the results are astounding. When Uncle Sam returned last year's tax refund, I picked up a Tascam 3 board to go with the dbx, so now I'm even starting to think of myself as having a 4-track *pro* studio (I've always had delusions of grandeur).

This brings me to the other subject on my mind—the James Rupert articles on small studios. Since I picked up a degree in accounting somewhere in my wild past, the business end isn't too hard for me. I even have contracts with my clients—something that I don't think Mr. Rupert has mentioned so far. It doesn't hurt to lay out who is doing what for (to?) whom for how much, and what the final product will be. His ideas on promotion, advertising, and establishing contacts score 100%. When I first started up, I advertised in the local entertainment paper, talked to four or five of the most successful bands, and sat back to wait. What I discovered was that a) hardly anyone had the ambition to answer the ad, and b) the really successful bands all *knew* beyond the shadow of a doubt that even a four piece band needed at least 8 tracks to record on. So with heavy heart and a great deal of embarrassment at having only a four track set-up, I crawled into a corner with some back issues of *MR* to dream of the day when I would be able to establish a recording business. I was talking the situation over with my tape recorder one day (*anything* is better than talking to yourself), trying to decide how I could have misjudged the desire to record that I thought most bands have (had things changed that much since my band-playing days?). Anyway, I decided that things hadn't changed. The big, successful "you need an 8-track studio" bands all wanted to make records—that is, the idea around here seems to be that the band rents studio time, produces a master tape, and then produces its own record album for sale in local stores and air-play on the radio. Presumably, they send the home grown albums to various record label A&R people also. Under these circumstances, I'd agree that a band would be better off in at least an 8-track facility. But I didn't give up, I sort of restructured my own thinking. Going the "do-your-own album" route seemed to be an awfully expensive way for a band to break into even a regional market, but rather than argue my viewpoint, I simply ignored the bands with that attitude.

Going from bar to bar on Friday and Saturday nights to talk to working bands is time consuming and expensive (and sometimes fun—hic!). Driving to the local music equipment dealers and copying band names and phone numbers from their referral boards is a *whole* lot easier. So is answering musicians' ads in the newspapers. And that's what I did. Of course, it took a few calls to develop a decent phone rap, but nobody hung up on me. I emphasized that I do *demo sessions*, that my equipment is high caliber, that I *didn't* make record album masters, and that the price was right. Bottom line? I have more business than I ever dreamed possible. I still run ads in the local papers, but virtually all of my clients have been developed through telephone contacts. I'm no longer embarrassed to have "only" a 4 track set-up, I'm having fun, making money, and have developed some good friendships to boot. My clients use their demo tapes to audition for new bookings and to contact A&R people. A few have come back to record albums that they sell at performances, distribute in record stores, and send to radio stations. Only these are *cassette* albums, not records. (It's less expensive and a lot less can go wrong).

This started out as a simple thank you letter and I didn't intend to become so verbose, but perhaps there are other semi-pros out there who have been talking to their tape decks lately and need some encouragement. Don't give up! Evaluate your capabilities, set realistic goals, and above all know the market you want to reach and don't be afraid of doing the necessary legwork.

I don't claim to be an expert (not by a long shot) on any of this, but I'll be glad to answer any questions from semi-pro recordists via the mail provided a stamped, self-addressed envelope is included.

Please include my address should you decide to print this letter. And please bear with my typing . . . it never was my forte. Thanks again.

—Mark Simmons
Phase One Recordings
2006 Piney Point Rd.
Brunswick, N.Y. 12180

Snowmen Looking for Drivers

I have just purchased your Winter '80 *Buyer's Guide*. As the soundman for a rock group I was disappointed in the fact that there was no section on "drivers".

In my opinion, speakers are the most

important sections in the signal flow. Even the most sophisticated front-end and amplifier equipment will sound terrible through the wrong, or through poor quality drivers.

I would appreciate your comments on this matter. Thank you.

—Philip Thomas
Snowmen
Torrance, Ca.

Yeah, it's true, we have no section on drivers plain and simple, but there are sections on "Studio Monitors" and on "P.A. & Stage Monitors." Driver type and size are included in each of those sections. Is okay?

Commercial Audio

My partner and I have enjoyed your monthly and consider it very valuable to our new studio.

We have a small but well-designed 6-track studio, ready for 24 when we get the bucks, and we have started up the beginnings of a commercial audio production company.

We put together original jingle packages and slogan concepts for local companies. Our first major jingle is being considered by a large frame shop corporation which is expanding its company.

Could you folks, if you have not already, give all of us in the same sort of position an in-depth article(s) about the commercial audio production industry both nationally and locally?

If one can get a foothold in his city or the chance of national exposure, it's a way to make a living in the small studio.

—Kenneth W. Darling
Portland, Ore.

You betcha. Our own Jim Rupert (Rupert to his many admirers) has kindled a candle in the mysterious jingle jungle in a recent article ("Stalking the Wild Studio Customer," April, 1980) and is working on another that we expect will answer many questions on juggling the jingle (okay, okay, we'll stop) business.

Public Domain or Permission?

What I'm interested in knowing concerns the method of obtaining permission to use a recording in a motion picture. For example, a scene in part of my film involves a car radio. The music being broadcast that day is recorded in the soundtrack of the film. Is not the broadcast a matter of public domain and therefore not subject to copyright laws? A similar circumstance occurs in everyday life when a TV news program tapes a local scene and broadcast music is part of the telecast.

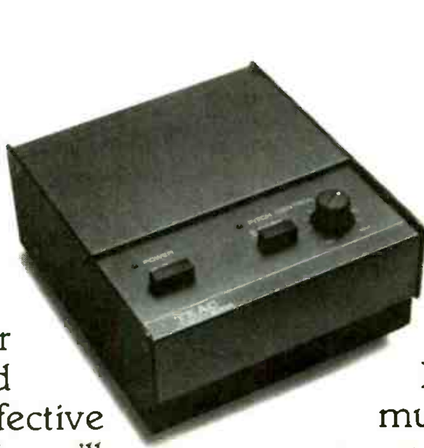
If permission and royalties are necessary, who does one contact to use contemporary radio music?

Your publication has a fine and intelligent focus. Thanks for the help.

—P. Panza
Earthrise
Beaver Falls, Pa.

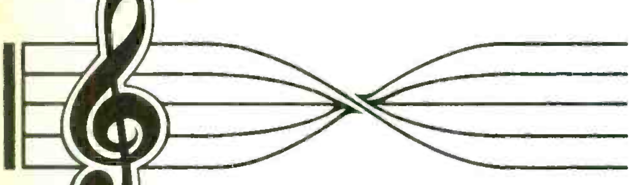
Honestly, we're stumped. But the sources when questioning rights and permissions are ASCAP (American Society of Composers, Authors and Publishers), 1 Lincoln Plaza, New York, N.Y. 10023 (212) 595-3050 and BMI (Broadcast Music, Inc.), 320 W. 57th St., New York, N.Y. 10019 (212) 586-2000. Contact them with your question. Best of luck to you.

OUR VARIABLE SPEED CONTROL WILL MAKE YOU CHANGE YOUR TUNE.



If you're already working with an 80-8 or 40-4, our Variable Speed Control is a very cost-effective addition. For just $350* you'll adjust 15 ips to the tune of ±20%.

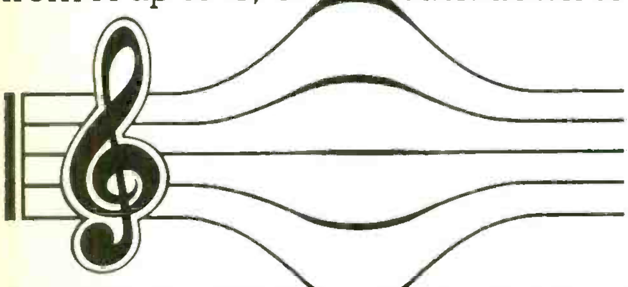
And you'll get a brand new single speed servo-controlled DC motor in the deal. Your multichannel recorder becomes more versatile. And it ends up last-ing longer. Remember trying to over-dub a piano only to find it out of tune with the track? Or sweat-



ing through three hours with a singer who flatted the last note of an otherwise flawless performance? You'll turn these late-night horror stories into lullabies with Variable Speed Control.

Try it for adding a "tunable tom" effect to your song. Then experiment with other rhythmic twists.

Turn two singers into a chorus of eight. Add harmonies. Transpose from A up to C, or back down to

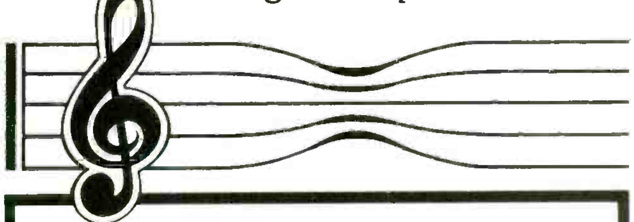


F#. With the 80-8, you have eight tracks to build your song.

When you're working with synthesizers, you can spend hours experimenting. Or seconds repairing an out-of-tune tone. Try creating your own special effects, bending and shaping other instruments to fit your ideas. Whether you have an 80-8 or 40-4, you have the capability to turn basic music into complex arrangements.

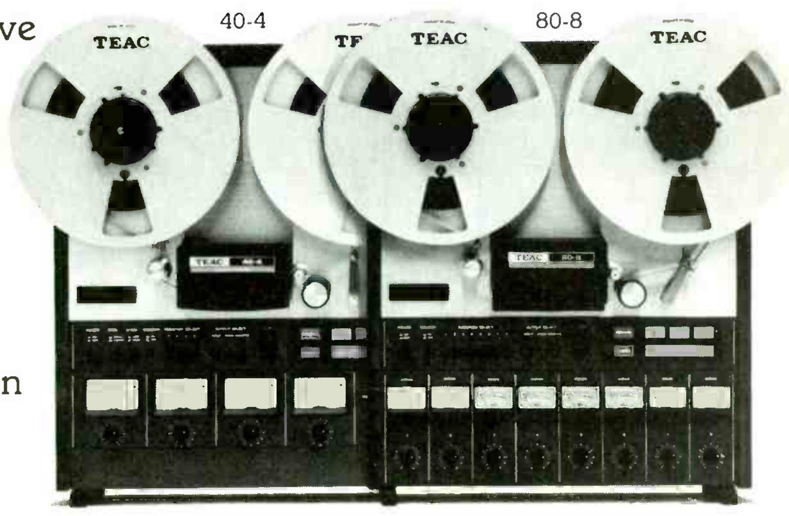
As a production aid, our Variable Speed Control becomes Executive Producer when that beautiful radio spot comes in at 32 seconds. Just rewind the tape, set the control and 28 seconds later you're right on the money.

For audio-visual soundtracks, slide or filmstrip audio tracks, Variable Speed Control lets you solve tough cueing and timing problems. Without re-recording, wasting time and losing money.



If you buy an 80-8 before October 31, 1980, you can get the Variable Speed Control and the new DC servo-controlled motor free of charge—plus arrangements for free installation. Get all the details at your participating TASCAM Studio Series dealer today, and discover how easy it is to sing a new tune.

*Suggested list price, optional with dealer; installation required.



TASCAM STUDIO SERIES

TEAC Professional Products Group

TALK BACK

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

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

Putting Up a Good Front

We recently acquired a pair of Electro-Voice RE-16 dynamic cardioid microphones. Since these are generally considered to be high-quality microphones, and they have integral blast filters, we decided to use them as "front" vocal mics in a sound reinforcement system for a small club. Although the entire system is balanced for maximum level before feedback, as soon as anyone places a hand on, or even near, one of the RE-16s, the system begins to *howl*. How can we correct this problem?

—L. Le Phante
Bob Boone
Nowas, Ark.

Your statements about the RE-16s are correct, but the application you've chosen them for does not exactly suit this type of directional microphone.

The directional characteristics of the RE-16 are "created" by the slotted openings along the body of the microphone. If these openings are covered up when the microphone is hand-held, the microphone becomes *non*-directional.

If your system balance (maximum level before feedback) is adjusted with the RE-16s on stands, that is, in their normal cardioid condition, the system *will* feed back when the microphones are hand-held and become non-directional.

The best solution to this problem is to use the RE-16s in some other application, preferably a fixed location such as drum overheads, instrument amps, etc. For the vocal mics you can use models whose directional characteristics do not depend on full-length openings in the microphone body. Some examples: Electro-Voice PL-91 or PL-95, AKG D330 BT, Shure SM-58, Beyer M500.

—Anyad Pitchaya
Chief Engineer
Gypsy Sound Studios
Montreal, Quebec

The Confusing Question of Comfortable Curves

I own a medium-size 8-track recording studio. We have a 3M 8-track and 4-track along with Scully and Teac recorders. Our other equipment consists of an Audio Designs console, dbx noise reduction, a White equalizer, SAE power amps and JBL 4343 monitors.

According to technicians and service people I've spoken to, the control room (which measures 16' x 10' x 9½') has been EQ'd flat according to real-time analyzers. I have heard from sound reinforcement technicians that in "live" sound reinforcement, a flat EQ is not desirable nor "comfortable." Moreover, I was told that a slight dip in the midrange, with a boost in both the low- and high-frequency ranges results in a more desirable sound.

As for my studio, should my equalization show that slight curve, or is a flat EQ necessary in a studio application? I have noticed that some of the tapes we have done have sounded a bit "bassy." Do you think it's possible that we have been adding more bass in the mixdown to achieve that "comforting" curve, resulting in a tape that is too bassy when played back in another room? Confusing?

—Michael L. Boelter
Studio Manager
Music Factory Studios
Mequon, Wisc.

Room equalization is subject to a wide variety of factors and opinions. The development of a "house curve" for your studio will evolve from a combination of instrumentation tests and listening evaluations.

Often, in sound reinforcement applications in a large hall, the room will be tuned with a deliberate high-frequency roll-off. This equalization usually is in the form of a slope at a rate of 1-3 dB per octave starting around 1,000-3,000 Hz. This gradual attenuation of the high frequencies is generally perceived as being more listenable in large spaces.

In your control room with an enclosed volume of approximately 1500 cubic feet, a sound reinforcement roll-off would probably not be desirable. Most studios equalize for flat response up to 8-12 kHz. Above that frequency, a high-frequency roll-off of 2-4 dB per octave is often applied to reduce harshness and to acknowledge that most other non-professional listening environments have reduced high-frequency capability. While one might well choose a program equalization with a dip in the midrange and a boost in the lows and highs, I would not recommend such an equalization for your monitors. Tapes produced in such an environment might well sound too thin on other systems.

The problem of "bassy" tapes may be a monitoring problem or it may be caused by some other factors. First, when recording multiple tracks, it is easy to pick up spurious or exaggerated low frequency information that may not be obvious when listening to soloed tracks. Such low-frequency signals may be caused by leakage, sub-harmonics or proximity effect from cardioid microphones. When this information is combined during the mixing process, the low-frequency energy can be substantially increased. The use of careful microphone technique and high pass filters, when needed, can help solve this problem.

Another possibility is poor low-frequency response on your tape recorder. All tape recorders, ¼-track recorders in particular, are subject to a rise in low-frequency playback level, which is caused by the relationship between the path of the tape over the head and the frequency of the recorded signal. This phenomenon is knows as "hyperbolic head bump." At fifteen ips, this rise in response can be as much as +4 dB at a frequency between 75-125 Hz. Most professional recorders have a low-frequency playback equalizer to compensate for this. Some home and semi-pro recorders lack the low-frequency equalizer. As an example, the often-used Teac 3340S has a specified response curve that allows up to a +5 dB increase in low-frequency playback level. You should verify the performance of your recorders and the recorders you are using for evaluation.

As for adding more bass to compensate for this and create a "comforting" sound, the more likely reaction would be to fail to add adequate high-frequency response/equalization.

It certainly is true that equalization of all types can be confusing. Once you know your room response, keep a careful log of all your procedures on several projects and compare your notes with your observed results. The written notes can help you focus on the correct procedure for your studio.

—Van Webster
President
Digital Sound Recording
Los Angeles, Ca.

Soundman Wanted

I am a musician in a road band playing bars, small clubs, etc. We have two Biamp 12-channel mixing boards and we run everything through them. This includes six keyboards, bass, guitar, and drums. We mix the band from on stage

and using a good sound check and a lot of luck our mix is usually adequate. It would be very useful to have a metering system that would indicate the relative percentage of each channel or group of channels relative to the total mix. The Biamp boards have channel outs, so the signal from each channel would be easy to get. Thus, as the band got louder or softer, we could look at the meters and see if the mix was still together.

We also have six vocal mics on stage. Most of the time, only one is in use. There are never more than four in use at any one time. Do you think using noise gates on the mics would improve the sound? Perhaps some kind of proximity switch could be used to turn the mic on when the singer approaches it. What do you think of this idea?

I've also found that in "live" sound reinforcement, a couple of dB boost at about 2-4 K really makes the vocals pop out. What's your opinion on this?

—Chris Michael
Chicago, Ill.

The problems of mixing sound on stage are well known and, unfortunately, still unresolved. To date, there is no effective means of monitoring a front system from on stage because there are more factors involved than simply gauging relative volume levels of instruments and voices. Frequency response, for example, is greatly effected by the room acoustics which, under typical circumstances, change drastically as the night goes on and crowd sizes vary. Also, certain subjective qualities, such as punch and cut through capability, are greatly affected by the same factors. The types of equipment needed to properly monitor these characteristics are normally beyond the economic means of a group your size—and this is assuming all the components of your system are working properly! When an amplifier fails on the low-end portion of your system, it is highly probable that your stage volume will not allow you to discern this failure until the audience begins to react.

In conclusion, then, you have two choices. Continue mixing from on stage, using as much discretion as possible, or acquire a qualified sound technician. My inclination, given the size and complexity of your group, would lean toward a soundman. If he knows his job and what the group expects of him, he will become an increasingly invaluable member of your group.

As for your last question, most compression drivers have a 2-2.5 kHz peak in them and my inclination would be to—rather than *raise* that particular band on the equalizer to get more snap out of the system—lower it and raise the 4 and 8 K bandwidths just a bit. You'll get the same snap, but you won't get any distortion.

—Tim Nole
McCauley Sound Co.
Puyallup, Wash.

An Organically Sound Combination

Could you please advise me as to whether or not the T-2 Crumar organ is compatible with the Leslie Model 760 speaker?

—Oscar O. Webb, Jr.
Toledo, Ohio

The Leslie Combo Preamp 7370 is required when using a Crumar T-2 organ with a Leslie 760. The typical Leslie speaker system only has an amplifier and needs an external preamp to increase the gain of the signal from the organ. The 7370 also contains the Leslie motor controls which are necessary in creating the Leslie sound.

—John Vilagi
Product Specialist
Music Technology, Inc.
Garden City Park, N.Y.

A Medley of Mics for Musicians

Can you recommend a good microphone for amplifying a flute? We are currently using a cardioid microphone, I believe. We are also looking for a different mic for our drummer who has a simply huge drum kit. The mic we use now picks up the cymbals and the snare to a greater degree than we find acceptable. Would a parametric equalizer filter this out without affecting his voice?

—Mike Nibert
Taylor, Mich.

The important thing to remember here is that to obtain professional results, you must use professional equipment. The microphone, of course, is where good sound reinforcement begins.

I strongly urge any group that takes the quality of their musical performances seriously to use good reinforce-

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ment equipment, and that includes professional studio microphones. Microphones bearing brand names such as Neumann, Beyer, AKG, Shure, Electro-Voice or Sennheiser will never let you down.

To answer your first question, I would recommend using a good quality cardioid condenser mic for the flute, perhaps a Neumann KM-84 (make sure you put a windscreen on it). If something less expensive would suit you, then check out the Sony line of cardioid mics.

As for your question regarding a good mic for your drummer, I would recommend a cardioid dynamic microphone such as a Shure SM-58 or an Electro-Voice RE-16. Both of these models have directional characteristics and have good built-in "pop" filters permitting close-range vocal miking and rejection of unwanted sound.

—David Kalmbach
Chief Engineer
Atlantic Sound Productions
Marietta, Pa.

Fixing That Frustrating Flutter

I have recently subscribed to *Modern Recording & Music*, having found it to be a superb reference for both amateur and professional recording enthusiasts. My question may seem rather simple, but it has caused me many hours of grief and frustration while trying to produce a good, listenable sound. My difficulty involves the recording of six- and twelve-string guitars onto an open-reel tape deck.

I am using a Kenwood 5066 deck, Maxell LX-35-90 tape, a Shure Dynacom B microphone and a tape speed of 19 cms. During the recording process, the track sounds crisp and clean. However, during the playback the tape is plagued with flutter. I have made numerous attempts to remedy the problem with very little success. Do I have to compress the signal to fit the tape response, or is there an easier way to succeed without the use of a compander?

—Barry G. Coulton
New Britain, Pa.

There really can only be one cause for tape flutter, and that is the tape recorder itself.

You see, flutter is caused by a faulty tape path. Anything from the reel motors to tape guides, capstan roller and pinch roller on the tape transport can be at fault. There is *no* remedy outside of correcting the tape transport! My suggestion is to contact the tape recorder manufacturer concerning this problem.

There is an equal amount of flutter recorded on every instrument, but the frequencies generated by six- and twelve-string guitars make flutter far more apparent. Unless the tape recorder is functioning perfectly, you'll always have problems when recording these instruments.

Good luck with your work,

—Jack Malken
Secret Sound Studio, Inc.
New York, N.Y.

Tricky Switch

I have a very unusual problem concerning my Kenwood KX-620 Stereo Cassette Recorder.

When I first turn on my set, the right channel does not record, erase or playback. However, after about 30 minutes of operation, the right channel cuts back in, and functions normally.

I suspect that the problem lies in a small gap in the right channel circuit board. When the set warms up, the gap is closed by the heat-expanded electical connectors.

Since my recorder is no longer covered by warranty, I would like to attempt repairs myself, and feel that if my diagnosis is correct, I have a better than even chance of fixing the recorder.

My question to you is: Should my diagnosis prove correct, is it possible to fix my machine using everyday tools, such as a multi-tester, soldering gun, and electrical pliers? I do not have extensive tools, such as an oscilloscope, but I do know how to handle the ones I have.

—Tina Schlereth
Forest Hills, N.Y.

The problem you described with your KX-620 Cassette Recorder in your letter points to, in my opinion, the record-playback switch as being at fault.

However, the record-playback switch is not a simple item to replace or fix. I strongly recommend you take your unit to our authorized service center in your area.

—Raul V. Hernandez
Technical Specialist
Kenwood Electronics, Inc.
Carson, Ca.

4680.
More power.
Fewer boxes.

You need cumbersome piles of equipment to have exciting, powerful sound—right? Not anymore.

The JBL 4680 Cabaret Line Array gives you higher efficiency, broad bandwidth and wide coverage without mountains of equipment. One JBL Line Array probably puts out twice the sound of the P.A. you're now using!

So your audience gets clean, crisp sound. And you have fewer boxes to lug around.

The 4680 houses an array of four full-range E Series speakers built with the new JBL Symmetrical Field Geometry magnetic structures which reduce magnetic circuit distortion to the lowest levels of any known speaker made today. The 4680 reproduces sound through four 10" cones rather than squeezing it through a small horn throat, which drastically reduces air nonlinearity. It also features two ring radiators that extend the high frequencies. And low-frequency venting for extended bass response.

JBL 4680 Line Array cabinets are crafted from multilaminate, cross-grained hardwood rather than soft pine plywood or particle board. Flush-fitting grille covers make the cabinets their own road cases. And the finish is a triple coat of durable black polyurethane. Tough, molded corners allow for stacking in a variety of configurations.

Even the voice coils have unique protection: To minimize bounce while traveling, they're automatically shorted when there isn't a plug in the input jack.

Cabaret Line Array systems are fully portable. They'll fit into a standard-size station wagon or van.

And JBL quality minimizes costly "downtime." It's built to the quality standard so many pros have come to rely on.

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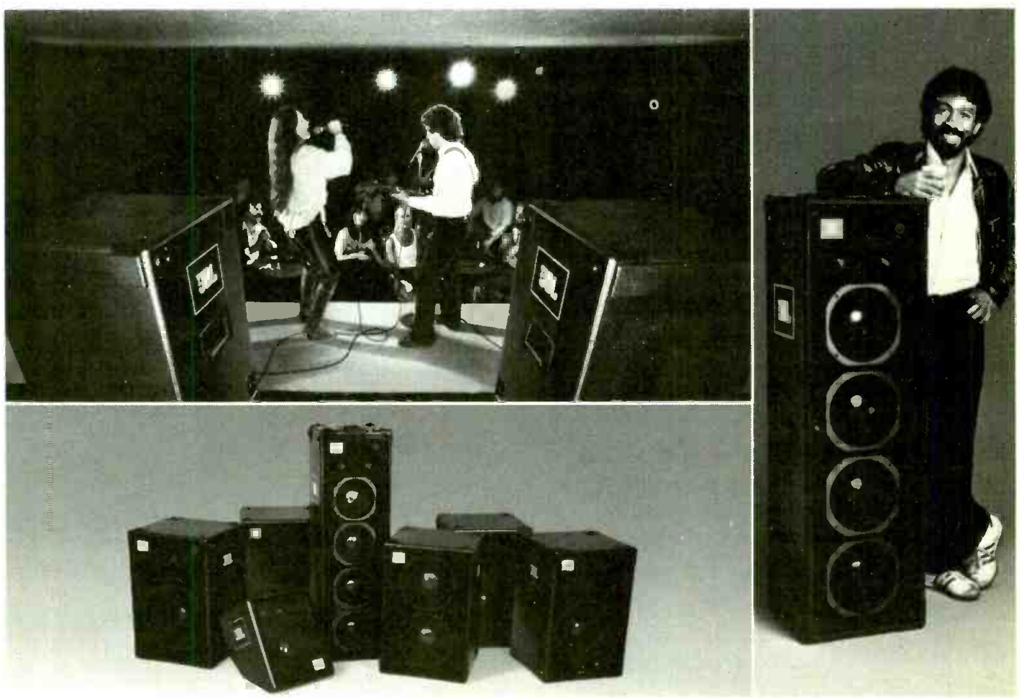
Model	Frequency Range	Power Capacity (Continuous Sine Wave)	Sensitivity (1W–1M)	Continuous Program	Crossover Frequency	Enclosure Volume
4680	55 Hz-15 kHz	300 W	103 dB SPL	600 W	3 kHz	142 litres 5 ft³



JBL First with the pros.

THE PRODUCT SCENE

By Norman Eisenberg

MIC/LINE DISTRIBUTION AMPLIFIER

Self-powered and rack-mountable, the new Model 7823 from Modular Audio Products is a microphone/line distribution amplifier that includes this company's model 4003 transformer-coupled mic preamp with adjustable gain to 65 dB, and its model 4820 balanced output amplifier which drives eight 600-ohm lines at +20 dBm. Uninterrupted output from the mic preamp (via a separate line-level transformer-coupled output) is always available. A source select switch enables the 7823 to be alternately used from a line-level source through its balanced, bridging differential input. Isolation between outputs is rated at 80 dB, and the manufacturer claims extremely low noise and distortion through the use of MAP audio op-amps.

CIRCLE 16 ON READER SERVICE CARD

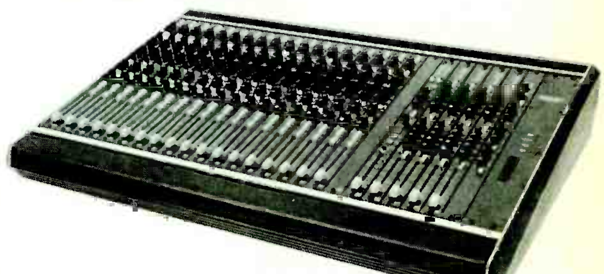
MIC CABLE TESTER

The model TE-2 from Wireworks Corp. combines all features of its predecessor, the TE-1, with an added test mode and direct plugging of a second connector type. The new test mode enables the user to check for conductors that may be shorted in the case of XLR-3 type connectors—something that can cause ground-loop or problems. The TE-2 also incorporates quarter-inch phone connectors so that it now has the capability to directly plug and test phone-to-phone cables as well as any combination of XLR-3-to-phone cable arrangements. Providing LED displays of shorts, opens and out-of-phase wiring, the TE-2 is a compact-sized, 9-volt battery-powered, steel-enclosed unit. Price is $89.

CIRCLE 17 ON READER SERVICE CARD

MODULAR MIXING CONSOLE

Soundcraft Electronics Limited's Series 400 is described as a fully modular 4-bus mixing console for up to 8-track recording and for "sophisticated" sound reinforcement. Two mainframe sizes are available: one for up to 18 input channels, the other for up to 26. According to the manufacturer, the Series 400 has more facilities than any other console in its class. Controls have forty-one detented positions except for the EQ and pan which have a single center detent. Metering is by sixteen-segment LED bar-graph displays, switchable individually to VU or peak readings. The transformerless differential mic preamp is claimed to produce "ultra low noise, and gives a transient performance better than conventional transformer-coupled preamps."



CIRCLE 18 ON READER SERVICE CARD

VERSATILE RACK CABINET

The RL200 tape transport console from Ruslang is designed to accept tape decks 19 inches wide by up to 21 inches in depth. Available in wood-grain finishes as well as solid colors, the RL200 is built with standard EIA tapped steel rails and it is available with rolling casters. Overall height less the casters is 31¾ inches.

CIRCLE 19 ON READER SERVICE CARD

HOT OFF THE PRESSES

Dolby Laboratories is offering, free on request, a four-page illustrated pamphlet that explains the theory and operation of its headroom-extension (HX) circuitry. HX is, essentially, a means of using the recording signal to control a cassette deck's bias and EQ in order to increase the dynamic range at high frequencies by an additional 10 dB.

CIRCLE 20 ON READER SERVICE CARD

"How to Build A Small Budget Recording Studio from Scratch . . . with 12 tested designs" is the title of a new book by F. Alton Everest, published by TAB Books and priced at $8.95. Soft-bound, it contains 335 pages and is heavily illustrated. The book covers such topics as the size of a studio, room effects, acoustical treatment, working layouts, construction plans, how to create studios in such places as a residence and garage and special considerations for different purposes such as campus radio, budget recording, radio program production, television and so on.

CIRCLE 21 ON READER SERVICE CARD

Of special interest to classical super-disc buffs is a catalog called "Classics Only" published by a new company of the same name, whose address is given as Box 14186, Columbus, Ohio 43214. The catalog costs 50 cents. Apparently, Classics Only is some kind of specialized disc marketing outlet, dealing in digitally-mastered classical recordings. Labels included in the catalog are Angel, Chalfont, Delos, Denon, DigiTech, London, RCA, Real Time, Sound 80, Telarc, Unicorn PCM and Varese Sarabande.

CIRCLE 22 ON READER SERVICE CARD

The Consumer Electronics 1980 Annual Review, published by the Electronic Industries Association (2001 Eye Street N.W., Washington, D.C. 20006) has just arrived. This 42-page booklet contains an overview, with plenty of hard sales statistics, of the whole consumer electronic field. According to a price list printed on the inside back cover, 1 to 5 copies are complimentary; 6 to 24 cost 25 cents each; 24 to 99 cost 20 cents each; 100 or more copies bring the cost down to 15 cents each.

CIRCLE 23 ON READER SERVICE CARD

ONKYO POWER AMP

From Onkyo comes word of the model M-5060 power amplifier, a stereo unit rated for 120 watts output (per channel) into 8 ohms, 20 Hz to 20 kHz, at no more than 0.005 percent THD. Priced at $800, the M-5060 uses dual-line construction; that is to say, it is designed basically as two mono amps sharing one chassis. Power supplies and circuitry for each are independent. Featured are power output meters with peak-hold, and a special protection circuit. Onkyo explains that since this amplifier is capable of large current delivery, heavy silver-plated parallel-connected relays are used to avoid a potential current "bottleneck." The amplifier has outputs for two sets of speakers, and separate gain controls for each channel. The circuit design emphasizes mini mizing of switching distortion, crossover distortion, thermal alteration or supply distortion, and other sources of distortion.

CIRCLE 24 ON READER SERVICE CARD

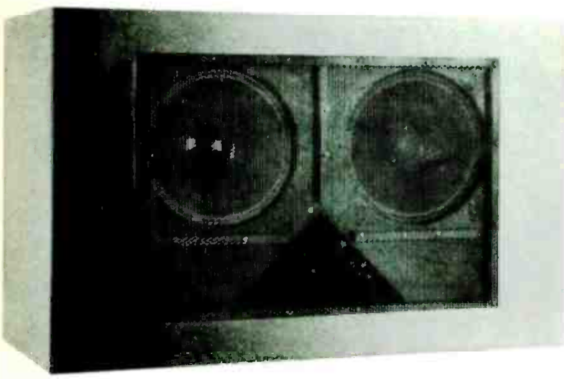
ELECTRET LAVALIER MIC

Calling it the "almost invisible" microphone, AKG has introduced its C-567 electret-condenser lavalier microphone, claimed to be the smallest lavalier obtainable while providing professional quality and durability. The transducer system is field replaceable. No battery compartment is provided in the C-567. Instead, the mic may be phantom powered from the mixer or recorder to which it is connected, or by an AKG external supply. Accessories include tie bars for one or two mics, a single mic tie-tac, a belt clip and a wire mesh windscreen. Response is omnidirectional, and frequency range runs from 20 Hz to 20 kHz. Price is $195 with carrying case.

CIRCLE 25 ON READER SERVICE CARD

SUBWOOFER FROM DBX

The firm of dbx enters a new product area with its Model 510 integrated subwoofer system, offered as a component in the dbx sound enhancement series (which includes the Boom Box and various dynamic range expanders). The new subwoofer features what dbx calls an exclusive speaker sensing circuit that provides complete overtravel and thermal protection for the drivers, accomplished by reducing the input signal with minimum disruption of the output signal. The drivers in the model 510 are two long-throw 15-inch speakers. Also in the system are the vented cabinet, internal electronic equalization, a 500-watt power amplifier, selectable high-pass and low-pass filters and rms and peak level limiters. Output at 23 Hz is listed as up to 120 dB/SPL. The model 510 weighs 210 pounds and stands 4 feet high. Price is $1200.



CIRCLE 26 ON READER SERVICE CARD

COMPACT SPECTRUM ANALYZER

A hand-held spectrum analyzer, the model ASA-10, has been announced by Gold Line. Claimed to be of professional quality, the ASA-10 is offered as an aid in balancing a sound system, making checkouts of equipment and systems and determining best placement for speakers. Used in conjunction with an equalizer, the unit—after "fine tuning"—may be plugged into the line for visual monitoring of performance. The ASA-10 also is suggested for use in measuring various noises and room sound levels. It incorporates ten filters on ISO center frequencies, and is powered by 8 AA cells or external supply.

CIRCLE 27 ON READER SERVICE CARD

HEADROOM METER

Inovonics has introduced the Gordon Headroom Meter, described as a third alternative to program monitoring. The device incorporates the traditional aspects of the VU meter and the European peak-program meter. It meets the UK/EBU standard for response to program peaks, but maintains what Inovonics describes as "a more conventional and artistically desirable 'syllabic' response to music and speech." Price is $122; a VU conversion option is available for $69.

CIRCLE 28 ON READER SERVICE CARD

SANYO "PLUS" SERIES

Sanyo is making a concerted bid for the high end and advanced audio market with its "Plus" series of components (one of which, the Super D Noise Reducer, was covered in our test reports in the June issue). The growing line of products includes new cassette decks with metal-tape capability, three head configuration and solenoid-controlled transport. One deck, the Plus D65, features auto-reverse in recording. Also in the line is the model Plus P55 stereo power amplifier, rated for 100 watts per channel into 4 or 8 ohms, with no more than 0.009 percent distortion across the range from 20 Hz to 20,000 Hz. The amp may be strapped for 200-watt mono operation. Peak power output on each channel is shown on a front-panel LED display arranged in a 12-section "staircase" pattern. The panel also has a speaker selector switch and a headphone jack.

Also of special interest is the Plus E55 audio program timer which uses dual and independently programmable channels. The E55 may be used for preprogramming unattended recording or playback, as well as for timer-activation of other appliances, lights, etc.

The new Sanyo "Plus" series products are all standard EIA rack-mountable.

CIRCLE 29 ON READER SERVICE CARD

SPECTRA SOUND FLANGER

Spectra Sound (a subsidiary of Spectra Sonics) has introduced the model 4000 flanger, designed to create effects such as positive and negative flanging, double tracking and speaker rotation simulation. In addition, the 4000 can create chorus, vibrato and tube echo effects. According to the manufacturer, the device produces over five octaves of flanging without input aliasing, output quantization noise, or

the introduction of high-frequency clock components. Modulation of the delay time can be accomplished by an internal variable sweep oscillator, or by the use of an external control voltage from a conventional foot pedal, joystick, synthesizer or computer device. Standard features include balanced and unbalanced inputs and outputs, LED overload indicators and inputs and outputs for slaving several other units. Price is $695.

CIRCLE 30 ON READER SERVICE CARD

KIT FOR BUILDING REAL TIME ANALYZER

Said to take two evenings to assemble is the new model 1081 kit from Logical Systems (Vancouver, Wash.) for building an audio real-time analyzer. Once assembled, the device displays a twenty-one dB range of signal energy in ten octaves, the latter matching the display of most graphic equalizers. Also featured are a diagnostic sweep signal and an input jack for dynamic microphone. Price is $179.

CIRCLE 31 ON READER SERVICE CARD

PAGES FROM AN AUDIO DIARY . . .

Martinique is a sun-drenched tourist's paradise where the food is tops and the bathing topless. That ought to be enough for someone seeking rest and relaxation after surgery. But I've just discovered something less exotic while no less pleasantly surprising. It's a Peavey sound-reinforcement system installed in the Meridien Hotel, and being put to good use by some highly talented musicians, including the greatest steel band I have yet heard. The ensemble effects and transients created by these performers are something terrific, and it all sounds loud and clear, even over the inevitable babble of audience voices and the tinkling of fruit-juice-and-rum-filled glasses. A real good job, and even if it reminds me of work waiting for me back home, it's a nice familiar touch while away from home.

On board the DC-8 it's amazing how two different persons using the same amplified intercom can manage to get it to sound so different. One voice, pointing out the scenic high spots, seems to float gently through the cabin, with excellent intelligibility. Another voice, explaining about the emergency exits and such, seems loaded with distortion. Can it be that the very subject matter a person transmits over a sound system influences the way he or she modulates the voice? I will not speculate on the possibility that the sound system itself is reacting to the different kinds of message—that's just plain spooky. Of course, it all could be simply how close the mic is placed to the speaker's mouth. I'll have to check this out more carefully on future flights.

Home again, and waiting for my attention—among other items—are the latest Koss stereophones. The model HV/X is a surprise as I unpack it. This may be the lightest-weight headset I ever have seen from Koss. It also is an obvious departure in general design from the well-known PRO-4 series since the cups do not fully enclose the ears, and the "open back" does permit some room sounds (the telephone ringing, for one) to be heard while you wear them. How will this headset sound?

Well, in a word, great. The low end is all there, even without the cups sealing in your ears completely. And the middles and highs take on a very nice "well-aired" quality that seems to lend depth and space to the reproduced stereo. I make the smoothest range of the HV/X to run from about 40 Hz to beyond audibility, with response still evident below 40 Hz albeit rolled off somewhat. Even with bass boost, however, there is very little evidence of low-frequency doubling. For direct monitoring off a tape deck, you probably will want to crank up the deck's playback level control, but for many users what the Koss HV/X lacks in ultimate efficiency it may well make up for in sheer tonal clarity.

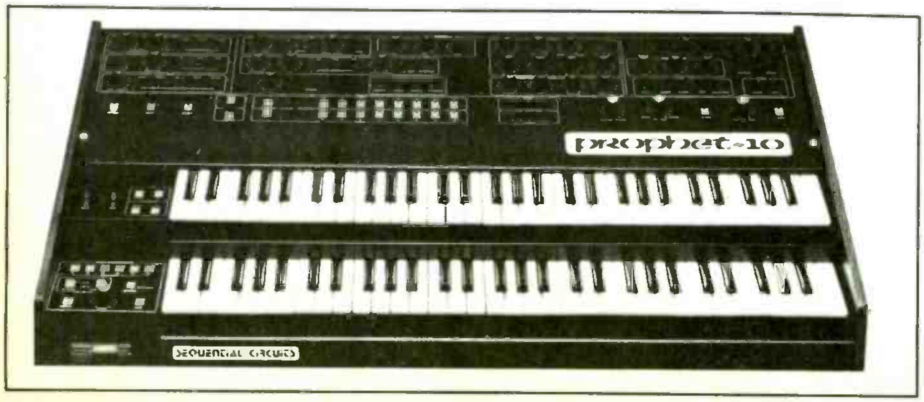
CIRCLE 32 ON READER SERVICE CARD

MUSICAL NEWSICALS

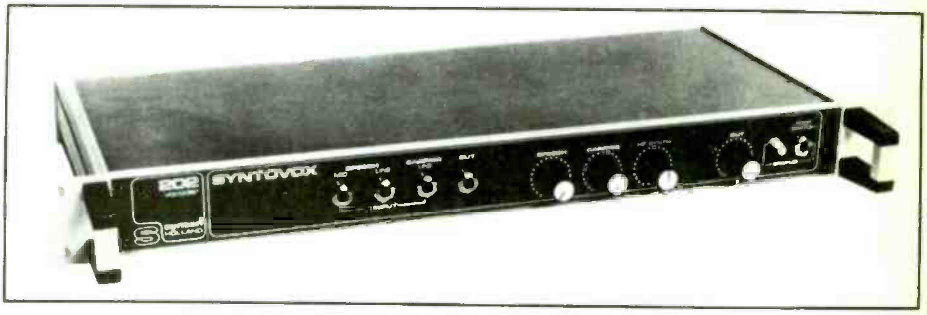
SYNTHESIZER EQUIPMENT

Sequential Circuits Inc. has announced that its very popular Prophet 5 programmable synthesizer now has a big brother, the Prophet 10. The new Prophet is fully programmable like its predecessor, but is a two-manual polyphonic design with five voices per manual for a total of ten voices. Each set of five voices has its own program so that completely different sounds are available on each manual. Each voice has two VCOs, a fourpole low pass filter, two ADSR envelope generators, a mixer and independent modulation capability. Additionally, the lower keyboard may be controlled by an optional polyphonic sequencer which includes built-in cassette data storage. Other features of the Prophet 10 include pitch-bend and modulation wheels, octave switches, polyphonic modulation, four voice-assignment modes with indicator LEDs, a program increment footswitch, two programmable and assignable control voltage pedals, three-band programmable volume control and a nonprogrammable master output volume control. Mono and stereo outputs are provided in both balanced and unbalanced formats.

CIRCLE 1 ON READER SERVICE CARD



As we have noted in this column before, vocoders seem to be one of the hottest items in both the accessory and synthesizer markets. One of the most comprehensive units available is the Dutch Syntbvox 221, a sophisticated 20-channel system marketed in this country by Parasound, Inc. Parasound recently announced the introduction of a new, simplified model from Syntovox which is oriented toward "live" performance use. The Syntovox 202 uses eight filter sections to produce an effective and intelligible vocoder effect without complexity. The front panel controls are likewise simple so that the musician has effective yet convenient control over the unit, and a footswitch makes it easy to switch the effect in and out even during a "live" performance.



CIRCLE 2 ON READER SERVICE CARD

Aries Music, Inc. has announced the introduction of the Aries Instrument Modification System II, which is a simple, compact synthesizer designed to be controlled by any conventional single-note musical instrument such as trumpet, saxophone or other horns. The new Aries system is a modular synthesizer which follows both the pitch and the dynamics of the source instrument to extend the range of sounds available to the musician. The basic synthesizer section of the Instrument Modification System includes a dual VCO, a multi-mode filter, and a dual VCA for basic tone generation and shaping. Additional modules include a balanced modulator (ring modulator), and phasing and reverb modules. Also included as part of the system are two control voltage foot pedals which can be patched into any module for additional control of the overall sound. The Aries Instrument Modification System II is available in both assembled and kit versions as are most of Aries' products.

CIRCLE 3 ON READER SERVICE CARD

MICROPHONES AND ACCESSORIES

Countryman Associates has announced the introduction of the EM-101 miniature electret condenser microphone, which is this country's first entry into the microphone field (unless you want to count their electrostatic piano pickup systems as microphones). The EM-101 is an omnidirectional mic with a small .2″ × .3″ diaphragm for excellent transient response. Its small overall dimensions (.6″ × .3″ × .15″) make it particularly useful for such applications as miking acoustic instruments where it will often produce better sound quality than typical piezoelectric or accelerometer type contact pickups. Its maximum sound level of 149 dB SPL allows its use on even the loudest instruments without distortion. Another possible application is to position the mic very near a reflecting surface to take advantage of the pressure zone effect. Frequency response is specified as 20 Hz-15 kHz ±1.5 dB. The mic has a low impedance (150 ohm) balanced output and requires 48 volt phantom powering for its preamplifier.

CIRCLE 4 ON READER SERVICE CARD

In the last few years there has been a renewal of interest in various single-point stereo miking techniques, and a new accessory item from Shure should make it more convenient for recordists to try such techniques with two separate mics rather than having to invest in special stereo microphones. The Shure A27M is an adapter which allows two microphones to be mounted in close proximity on a single stand. The two sections of the adapter may be rotated with respect to each other to accommodate a wide variety of positioning techniques including the common X-Y and M-S patterns. To accommodate microphones of varying diameters or for special effects, the two sections of the adapter may be assembled to provide vertical separation between mics of 1⅛″, 2⅝″ or 4″.

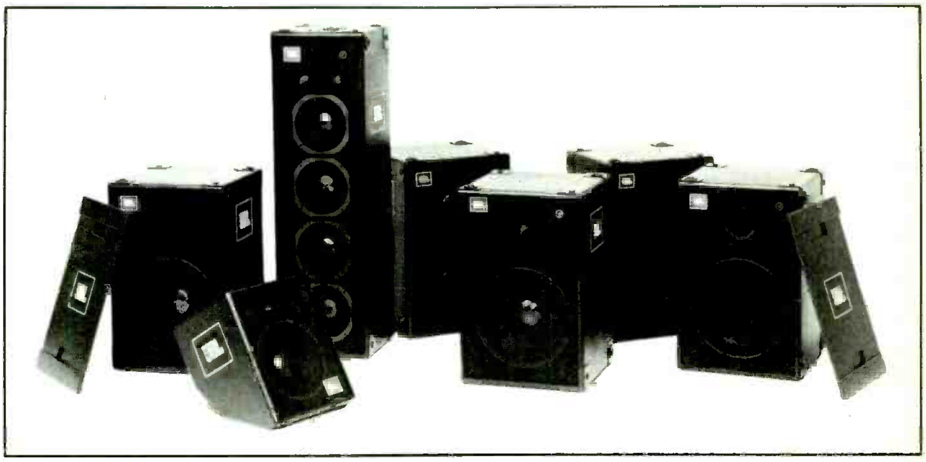
CIRCLE 5 ON READER SERVICE CARD

SPEAKER SYSTEMS

For incredibly flexible monitoring, Mirage Audio offers the Shortstack MiniMonitor speaker system. The Shortstack houses a pair of high-efficiency 4½-inch cone drivers in a compact enclosure of Duron. The system is small enough (10½ × 5½ × 4½″) and light enough that it can be mounted on any mic stand via the 5/8-inch threaded flange on the bottom of the enclosure, thus allowing total flexibility in locating and aiming the speaker. The Shortstack is rated at 100 watt RMS power handling capacity, and frequency response from 100 Hz to 15 kHz ±6 dB, and produces 87 dB SPL at a distance of 1 meter on axis with a 1 watt input.

CIRCLE 6 ON READER SERVICE CARD

JBL has once again expanded its line of Cabaret Series speaker systems. The four new models were designed for musical instrument amplification applications and include the Model



4621 single-speaker lead guitar system; the 4623 acoustic guitar system; the 4625 bass guitar system; and the 4627 keyboard system. All Cabaret Series speaker systems use JBL's E Series loudspeakers for high efficiency and power handling capability, and feature Baltic birch cabinets with flush-mounted transport covers and handles for ease in handling. The four new additions to the Cabaret Series augment the existing models which include a dual-speaker lead guitar cabinet, a stage monitor system and the Model 4680 line array sound reinforcement speaker.

CIRCLE 7 ON READER SERVICE CARD

MIXING CONSOLES

A new line of stereo mixing consoles in formats ranging from eight inputs through 24 inputs has been announced by Peavey Electronics. The new Peavey Mark III Series was designed for professional performance and versatility and incorporates several features not commonly found in sound reinforcement mixers. The input channel circuitry is transformer balanced and has a wide gain range to accommodate a wide variety of input signals and has switchable +48-volt phantom powering. Each channel also features four-band equalization, two pre-fader monitor sends, two post-fader echo/effects Comprehensive patching facilities are provided with pre- and post-fader insert patch points in each input channel and insert patching for each mix bus and output bus. Effects and reverb returns are provided for both monitor busses as well as the main busses. The line outputs for both main output channels, both monitor channels and a summed mono channel derived from the stereo main busses are transformer balanced. Ten-segment LED level indicators are used for the main and monitor outputs to display peak levels.

CIRCLE 8 ON READER SERVICE CARD

Tour Sound Products manufactures a full range of P.A., monitor, and instrument amplification systems. On the electronic end of its line, Tour Sound Products has announced the introduction of the Model 450 SC powered mixing console. The 450 SC is a four-input stereo mixer integrated with a 40-watt per channel (RMS at 1.0% distortion) power amplifier. Each input has bass and treble EQ, an effects send, panpot and rotary fader. Also available is the model 850 SC which has eight inputs and output of 80 watts per channel.

CIRCLE 9 ON READER SERVICE CARD

Road Electronics has introduced a new 8-input stereo mixer, the Model RS-2308. Each input channel of the new model has a transformerless differential input amplifier which accepts balanced microphones via XLR-type connectors and unbalanced line signals via ¼-inch phone jacks. Each channel has bass and treble EQ with a selectable midrange EQ section with a 500 Hz to 5 kHz range, a monitor send which is switchable pre- or post-EQ and fader and channel send/receive patch points for external effects devices. The unit also includes two 8-band graphic equalizers which are externally patched for versatility; 15 dB of cut or boost is available in each octave band and the uppermost (8 kHz) and lowest (63 Hz) bands are shelving type rather than peaking/dipping sections.

CIRCLE 37 ON READER SERVICE CARD

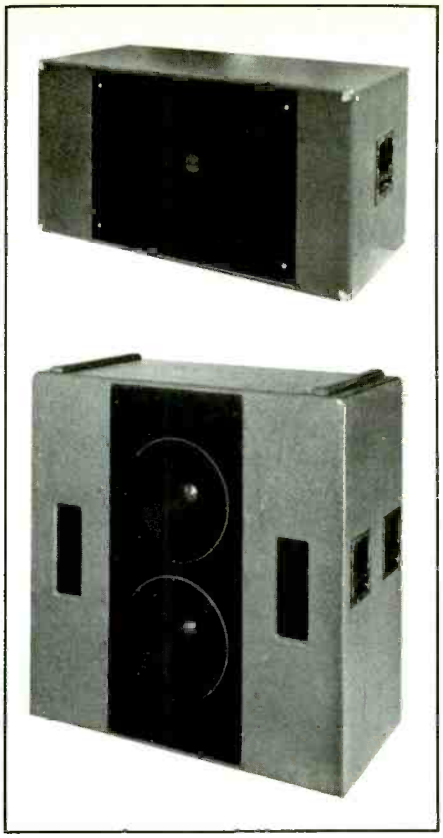
MUSICAL INSTRUMENTS

ARP Instruments, Inc. has also done some branching out from its established synthesizer market with the introduction of its new 16-Voice Electronic Piano. ARP intends to continue its commitment to research and development in the synthesizer field but will supplement that with the development of a series of products oriented toward the electronic piano and organ customer. The ARP Electronic Piano is a 73-note instrument with a specially designed maple keyboard mechanism which accurately reproduces the feel and response of a traditional grand piano action. The instrument is touch-sensitive and its sixteen voices offer a wide range of sounds including piano, vibes, harpsichord, harp and electric piano among others.

CIRCLE 38 ON READER SERVICE CARD

SOUND REINFORCEMENT EQUIPMENT

Those of our readers who were fortunate enough to see and hear Pink Floyd at the group's recent concert appearances in Los Angeles or New York have heard first-hand what Stanley Screamers can do for a P.A. system. Stanley Screamers are a line of professional sound reinforcement products manufactured by Altec Lansing and designed by Altec in collaboration with Stan Miller of Stanal Sound (Kearney, Nebraska and Van Nuys, California). Stanal has been a staunch supporter of Altec products for years, and the system Miller furnished to Pink Floyd used all Altec speaker components including several Stanley Screamer models. The first two Stanley Screamer products to be made available to the public through the Altec dealer network are the Model 1020-R subwoofer system and the Model 3210-R three-way, tri-amp



speaker system. The 1020-R uses a pair of Altec Lansing 421-8LF drivers in an enclosure that measures approximately 4′ × 4′ × 2′ to produce a high acoustic output in the 20 Hz to 80 Hz range. The 3210-R system uses a pair of Altec 604-HPLN 15-inch duplex loudspeakers to cover the bass and midrange plus an Altec 291-16F compression driver with Altec's exclusive Tangerine radial phase plug coupled to a Mantaray constant directivity 90° × 40° horn for the treble. The 3210-R system does not have any crossover networks and thus must be used with a low-level crossover and tri-amp amplifier set-up. Both systems feature fiberglass-covered plywood enclosures and are available in Utility versions for permanent or semi-permanent installations or in Road versions complete with steel corners, recessed handles and, in some cases, fiberglass lids and hardwood skids for touring use.

CIRCLE 39 ON READER SERVICE CARD

Many sound reinforcement systems are based around horn-loaded or bass reflex bass enclosures and compression driver/horn combinations for midrange and treble. But some designers, including the folk at Bag End Modular Sound Systems feel that a better sounding system can be designed around cone-type drivers in sealed or semi-sealed enclosures. Bag End offers a wide range of speaker enclosures for a variety of applications. The Bag End Bass Series includes 1 × 12″, 2 × 12″, 1 × 15″, 2 × 15″ and 1 × 18″ models available either empty or loaded with JBL or Gauss drivers; applications for this series include keyboard and bass guitar amplification as well as low frequency sound reinforcement use. The midrange series includes 10-inch, 12-inch and 15-inch systems available empty or loaded and double systems in the same driver sizes available unloaded only; applications for this series of semi-open backed systems include guitar and keyboard amplification plus midrange use in multi-way P.A.'s. A more unusual approach is seen in Bag End's Wedge Series which is designed as a three-way system which will stack in an arc of 3-foot radius for high power capacity and wide dispersion; a 6-foot-wide array, for example, will cover a full 180°. The bottom section of the Wedge Series houses 10-inch speakers for midrange reproduction from 125 Hz to 1 kHz, the middle section uses 5-inch speakers such as the JBL 2105 to cover high frequencies, and the top section holds very high frequency units such as the Electro-Voice T350 or the JBL 2402 to cover the range from approximately 4 kHz on up.

CIRCLE 40 ON READER SERVICE CARD

MUSICAL INSTRUMENT AMPLIFIERS

St. Louis Music Supply Co. has announced several additions to its Crate line of compact guitar and bass amps. The CR65 and CR65DL are 60-watt amps with a single 12-inch speaker (a Celestion in the case of the CR65DL) which feature two input channels with distinctively different tonal qualities. One preamp channel is a BiFet integrated circuit design with gain, bass, parametric midrange, treble and master volume controls, while the other preamp channel uses computer-type CMOS integrated circuits and has controls for gain, bass, treble and master volume. The two channels are selected by means of a footswitch with front panel status indicator lights. Other features include a combined volume control for when the two channels are used simultaneously, a gain switch for high volume/low distortion operation and a

three position bright switch. Also new in the Crate line is the MiniCrate model CR-M, a budget-priced 20-watt amp with a single 10-inch speaker. This simple model has gain, bass, treble and master volume controls, two input jacks and a line out jack in addition to the speaker output. Like all Crate amps, the Mini-Crate is housed in a ¾-inch Ponderosa Pine cabinet with steel corners.

CIRCLE 10 ON READER SERVICE CARD

MUSICAL INSTRUMENT ACCESSORIES

RolandCorp US recently announced two new additions to their Boss Rocker line of electronic sound modifiers. The first of these is the PD-1 Rocker Distortion, the first distortion device to be housed in Roland's rugged Rocker type pedal, which uses solid-state Hall Effect devices rather than potentiometers or photoelectric devices to control the pedal circuitry. The PD-1 Rocker Distortion uses pedal position to control the amount of distortion effect. A light touch on the heel of the pedal activates a silent FET switch which engages the effect with the minimum degree of distortion set by a special control; pushing the toe of the pedal increases the gain and hence distortion up to a maximum gain of 43 dB at the full forward position where another silent FET switch activates an additional 20 dB of boost for super distortion effects. In addition to the minimum distortion control, the PD-1 has level and tone controls to shape the overall sound of the effect. The other new Boss Rocker product is the PW-1 Rocker Wah, which was designed to offer extra flexibility in addition to a classic wah effect. The flexibility comes primarily from a range switch which selects bass, normal or treble ranges, and a width control which varies the width of the range of frequencies covered by the throw of the Rocker pedal. Both units use silent FET switching throughout, have LED on/off indicators, and can be powered by two 1.5 volt batteries or an optional AC adapter.

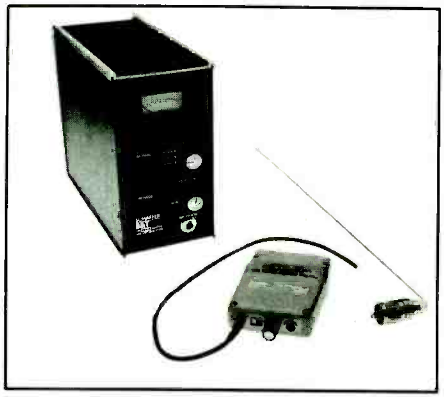
CIRCLE 11 ON READER SERVICE CARD

Silent Partner is a low-priced, compact headphone amplifier which can be used for practice, tuning or audio system troubleshooting. The unit has a 50 K ohm input impedance for minimum circuit loading and will drive any headphone from 8 ohms to 2000 ohms. Frequency response is 40 Hz to 20 kHz, and the unit will deliver ¼ watt into 8-ohm phones. Silent Partner, from Artists X-Ponent Engineering is powered by a single 9-volt battery, and weighs only 4 ounces.

CIRCLE 12 ON READER SERVICE CARD

WIRELESS SYSTEMS

Many professional users of wireless systems for "live" performance applications consider the Schaffer-Vega Diversity system to be the Cadillac of wireless systems, but unfortunately the system also carries a Cadillac-like price tag somewhere on the far side of $3400. The Ken Schaffer Group recently announced the availability of a new, lower-cost wireless system known as the Schaffer B&T.



The basic difference between the Schaffer B&T and its expensive big brother is that the new system is not a true diversity system. (There has been considerable confusion about what is and is not a true diversity system, but what it all boils down to is how many *receivers* the system has. Two or more receivers and the appropriate circuitry to switch between them automatically constitute a true diversity system; two antennas connected to a single receiver does not constitute a true diversity system.) The Schaffer B&T system uses a single antenna receiver which is virtually identical in audio performance to the Schaffer-Vega Diversity system since it shares the same crystal-controlled frequency system, 4-stage helical resonator receiver circuitry and audio compander system to provide a 90 dB or better signal-to-noise ratio without resorting to interference-causing wide band transmission. In practical terms the primary difference between the new system and the Diversity system is that the user must allow a few minutes during sound check to test for the absence of RF "dead spots" which the diversity system is designed to eliminate automatically. If dead spots are encountered during sound check, it simply means that the antenna must be repositioned until it receives an adequate signal from the wireless transmitter from any location the musician is likely to occupy during the performance. Once the proper location is found, the Schaffer B&T system is capable of delivering top quality performance with minimum noise and interference.

CIRCLE 13 ON READER SERVICE CARD

KEYBOARD MIXERS

Dallas Music Industries has announced a new model, the 6+2 Keymix, which has been designed specifically for "live" use by keyboard players who require on-stage mixing facilities for their various instruments. The Kelsey 6+2 Keymix incorporates 6 mic level inputs, 6 line level inputs, stereo and mono outputs in both low-level balanced and high-level unbalanced formats, patching facilities for input and output busses and 10-segment peak level LED arrays on the stereo outputs. The Kelsey Keymix comes complete with a rugged, professional SMG road case for complete protection on the road.

CIRCLE 14 ON READER SERVICE CARD

A recent addition to the product line from Music Technology, Inc., is the MTI Polypatch System, a new mixer/amplifier system specifically designed for the keyboard player. The Polypatch System features four input channels, each with sensitivity to accept mic or line level inputs. Each channel has controls for level, high-, mid and low-frequency EQ, reverb send (reverb is built-in) and external effects send. The reverb send is post-EQ for greater control of the reverb sound while the independent effects send is pre-EQ and is patchable channel by channel. On the master side, the Polypatch has controls for master EQ, reverb return and effects return. One thing that makes the Polypatch unique is the inclusion in the unit of a high power amplifier capable of delivering 180 watts into 4 ohms or 250 watts into 2 ohms; distortion for the amplifier section is given as less than 0.2% at 150 watts into 4 ohms.

CIRCLE 15 ON READER SERVICE CARD

The ELECTRIC PRIMER

Part VIII

By Peter Weiss

At the end of Part VII, we looked at a circuit containing inductance and resistance, and introduced a "vector" method of analysis for A.C. circuits of this kind. We can apply the same method to circuits containing capacitance and resistance, and this is what we will do in this penultimate installment of the "Primer."

To begin, we have to examine the behavior of a series combination of capacitance and resistance in response to a sinusoidal alternating voltage. In Part III (Dec. 1979 issue), we described the time-related characteristics of a combination of capacitance and resistance. These same characteristics determine the circuit parameters that interest us.

Let us review the characteristics mentioned above. When a steady (D.C.) voltage is applied across a capacitor that is in a series combination with a resistor, the current flow starts at a maximum value and diminishes as the charge is built up on the plates of the capacitor. Then, when the charge on the capacitor plates (and thus the voltage measured across the capacitor) is at a maximum, the current flow is zero, since no more charge can move onto the plates. The relationship between the voltage across a capacitor and the current "through" a capacitor in an A.C. circuit is basically the same as that just described.

At this point, one question that might arise is, "How can current flow through a capacitor, when a capacitor consists of two metal plates separated by a non-conducting material?" It is true that a capacitor presents a virtually infinite resistance to the flow of electrons. However, when an alternating voltage is applied across a capacitor, an alternating current *seems* to flow through the capacitor. The process is as follows: As the charge on the plates builds up during the first positive 90° of an alternating voltage cycle, the apparent current (due to the movement of charge to the plates) goes from a maximum initial value (when there is not yet any charge on the plates) to zero as the charge already accumulated on the plates prevents any further accumulation. At this point the voltage across the capacitor is at its maximum positive value. As the alternating voltage across the capacitor enters the second 90° of the positive half of the cycle, it begins to decrease from its peak value, in effect *discharging* the capacitor. The current that results from this discharge starts at zero and increases smoothly, flowing in a direction *opposite* to that taken during the first 90°. As the voltage reaches zero again, at the completion of the first 180° of the cycle, the current flow (in the new direction) reaches a maximum. As the voltage enters the second (negative) 180°, it begins to charge the capacitor again, this time with a polarity opposite to the charging polarity during the positive half of the cycle. The "charging current," which was at a maximum at the 180° point of the voltage cycle, begins to decrease until it reaches zero at the same instant that the voltage cycle reaches its negative peak at 270° into the cycle. From this point, the discharge process takes place again; the voltage decreases from the negative peak towards zero, and the current goes (in the original direction) from zero to its maximum value. A result of this process is that the apparent current "leads" the voltage by 90°, or one quarter of a cycle. The relationship between voltage across (V_c), and the current through (I) a capacitor is shown in *Fig. 1.* (From this point on, we will refer to the apparent current through a capacitor as if it were an actual current flow.)

The magnitude of the alternating current through a capacitor for a given applied alternating voltage is dependent on the value of the capacitance and the frequency of the applied voltage. The opposition of a capacitor to the "flow" of current is called *capacitive reactance.* The symbol for capaci-

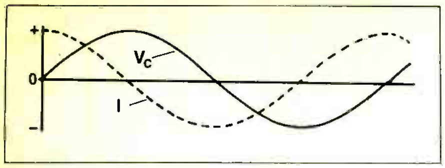


Figure 1

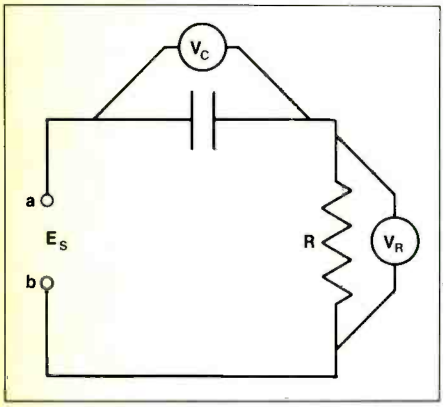


Figure 2

tive reactance is "X_c," and the unit is the ohm. The formula for this is

$$X_c = \frac{1}{2\pi \cdot f \cdot C}$$

where $2\pi \cong 6.28$, f is the frequency in Hertz and C is the capacitance *in farads*. [*Note:* This is the kind of situation for which scientific notation (see Part VI, April 1980 issue) comes in handy. Most capacitors have values given in microfarads (millionths of a farad, written μf) or micromicrofarads ($\mu\mu$f, also called picofarad, pf). These can be expressed in scientific notation as 10^{-6}f and 10^{-12}f, respectively.] To get a feel for working with this circuit parameter, let's calculate the capacitive reactance of a .47 μf (4.7×10^{-7}) capacitor when the applied signal frequency is 1000 Hz (10^3 Hz).

$$X_c = \frac{1}{2\pi \cdot f \cdot a}$$

$$X_c = \frac{1}{6.28 \times 10^3 \times 4.7 \times 10^{-7}}$$

$$X_c = \frac{1}{6.28 \times 4.7 \times 10^{-4}}$$

$$X_c = \frac{1}{29.53 \times 10^{-4}}$$

$$X_c = \frac{1}{2.95} \times 10^3$$

$$X_c = 340 \text{ ohms}$$

In *Fig. 2* the .47 μf capacitor has been placed in a series circuit with a 340 ohm resistance. An alternating voltage with a frequency of 1000 Hz (E_s) is applied at terminals **a** and **b**. We can use the vector method introduced earlier to compute the impedance (Z), or total opposition to alternating current, of this circuit. The vector representations of X_c, R and Z are shown in *Fig. 3*. To find Z:

$$Z = \sqrt{R^2 + X_c^2}$$

$$Z = \sqrt{(340)^2 + (340)^2}$$

$$Z = \sqrt{2 \times (340)^2}$$

$$Z = \sqrt{2} \times 340$$

$$Z = 480 \text{ ohms}$$

The phase angle, θ, is 45°, since $X_c = R$.

The sum of the voltage drop across C, V_c, and the voltage drop across R, V_r must equal E_s, the applied voltage, but this sum is the result of a vector *addition*. To accomplish this vector addition E_s is laid out as a vector as shown in *Fig. 4*. The length of the E_s vector represents the magnitude of the applied voltage and the angle θ represents the phase angle between the voltage and current in the circuit. To find V_c,

$$\sin 45° = \frac{V_C}{E_S}$$

Looking up sin 45°, we find that it is 0.707. So,

$$V_C = 0.707 \times E_S$$

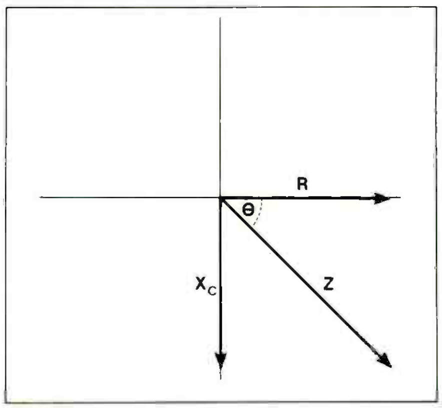


Figure 3

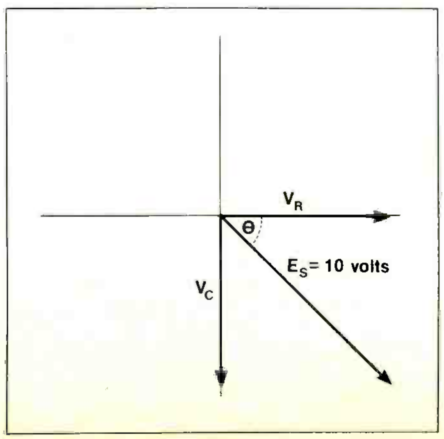


Figure 4

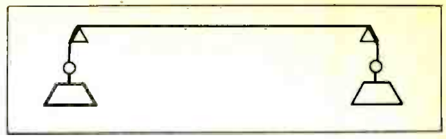


Figure 5

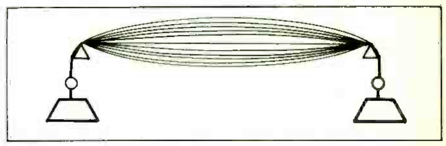


Figure 6

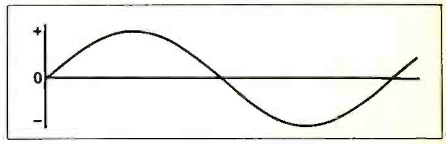


Figure 7

A similar calculation can be made for V_r:

$$\cos 45° = \frac{V_R}{E_S}$$

$$0.707 = \frac{V_R}{E_S}$$

$$V_R = 0.707 \times E_S$$

Remember that the conditions here were set up for easy calculations and simple visual presentation (i.e., equal values for X_c and R). This type of situation is helpful when explaining A.C. circuit analysis. Actual situations encountered in practice will generally be more complex, with impedances and other parameters calculated for several different frequencies.

Before we progress to some practical audio applications of the information already given, we must cover some fundamental facts and present some definitions of audio terms. We will depart briefly from purely electrical considerations and investigate some mechanical phenomena.

Vibrating Terms

First, let's examine *Fig. 5*. It shows a string stretched (by the weights) across two fixed supports. We know from experience that if such a string is picked or struck it will vibrate and produce a musical sound. But exactly what kind of sound, and how is this sound produced? Well, we can easily imagine that the string, after being "energized" by picking or striking, will vibrate as a whole as shown in *Fig. 6*. The rate or frequency of this vibration depends on the length of string between supports, the thickness of the string, and the tension in the string.

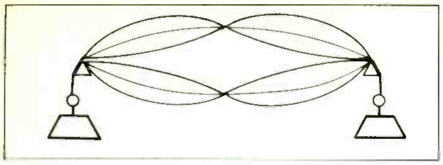


Figure 8

The frequency at which the string vibrates as a whole is called the *fundamental* frequency of vibration of the string. The sound wave it produces can be represented (for discussion purposes only, since sound waves in air are actually moving pressure variations) by the sine curve in *Fig. 7.* The frequency of this fundamental mode of vibration determines the pitch of the musical sound we perceive. If a vibrating string produced a sound containing only the fundamental frequency, we would hear (after the initial click from picking or striking) a "pure" tone, like a whistle. We know from experience that a vibrating string does *not* sound like a whistle. The reason it does not is that a vibrating string vibrates not only as a whole, but also in two separate *halves,* as shown in *Fig. 8.* This vibration of the halves, which happens simultaneously with the fundamental vibration, occurs at a frequency which is twice that of the fundamental. This higher frequency is called the *second harmonic.* Musically, it is one octave above the pitch of the fundamental. The sound wave that results from a string vibrating at its fundamental and second harmonic frequencies is shown (representationally) in *Fig. 9.*

Besides vibrating as a whole, and in halves, a vibrating string vibrates in thirds, each third vibrating at a frequency *three times* the fundamental. This frequency is called the *third harmonic.* A vibrating string also vibrates in fourths, fifths, sixths, etc. Each of these modes is also called a harmonic, and is numbered according to which multiple of the fundamental it is. Theoretically there are an infinite number of harmonics produced by a vibrating string. However (for simple stretched strings), the higher the number of the harmonic, the smaller its contribution to the total sound volume. Not all harmonics are spaced at musical intervals from the fundamental. As mentioned before, the second harmonic is an octave above the fundamental. The third harmonic is a twelfth above the fundamental, the fourth harmonic is *two* octaves above, but the *seventh* harmonic is not at a normal musical interval above the fundamental. Some instruments, pianos for example, are constructed in such a way as to mechanically limit the contribution of the seventh harmonic of each string to the total sound.

Just about every *musical* sound we encounter is made up, not of a single frequency tone, but of a complex jumble of frequencies, harmonically related. How and why do we hear differences between the following musical sounds: a soprano vocalizing a high C, a piano sounding a high C and a violin playing a high C? These musical sounds all have the same pitch, and therefore all contain at least the same fundamental frequency (in this case, 523.25 Hz). What distinguishes these sounds from one another is the different proportions of harmonic content in each. It is the presence and relative intensities of the harmonics (at least the first several) in each musical sound that enable us to identify the source. (**Note**: The way in which a tone begins and ends is also an important cue in determining the origin of a musical sound.) Therefore, in order to be useful, audio devices should not alter the harmonic content of the signals passing through them. A more general way of stating this, since all sounds, musical and otherwise, contain a number of different frequencies, is that an audio device should not emphasize or minimize any frequency or range of frequencies within the audio spectrum. A device with this characteristic is said to have a "flat" frequency response. This terminology is used because the frequency versus output graph of such a device is a straight horizontal line.

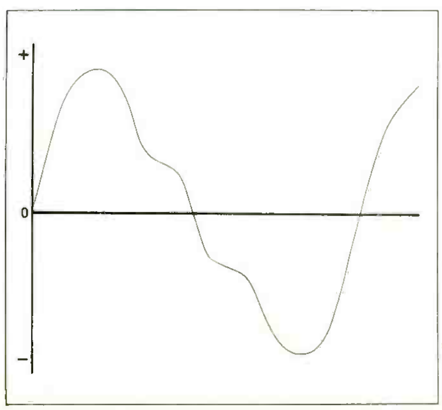


Figure 9

With the information we have so far we can describe the operation of two important devices encountered in audio practice: the fixed pre- or de-emphasis equalization circuit and the capacitor (condenser) microphone.

Audio Devices

Fixed equalization networks are employed in the audio sections of FM broadcast transmission systems and in all phases of audio recording and reproduction. In the case of audio tape recording, a boost in the treble end and "cut" (reduction) in the bass end of the audio range are necessary so that the high-frequency portions of the recorded signal are impressed (magnetically) on the tape at a level substantially higher than the pre-existing random high-frequency noise level on the tape. There are also other effects—like overloading in the bass portion of the spectrum—that this fixed equalization is designed to minimize or prevent. When the recorded signal is played back, it passes through another fixed equalization circuit having a frequency response that is complementary to the frequency response of the equalization circuit used during recording. That is, the playback frequency response has bass boost and treble cut. This playback equalization cancels out the effect of the recording equalization on the signal and, at the same time, because of the cut in the treble end, the high-frequency noise present on the tape is considerably reduced in level. The total result of the use of the two complementary equalization circuits is an overall "flat" frequency response and a substantial reduction in the level of tape noise.

The fixed equalization circuits mentioned above are generally contained

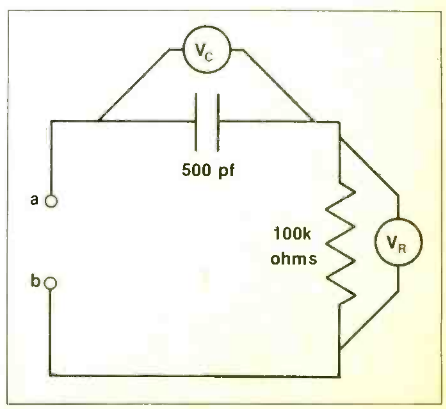


Figure 10

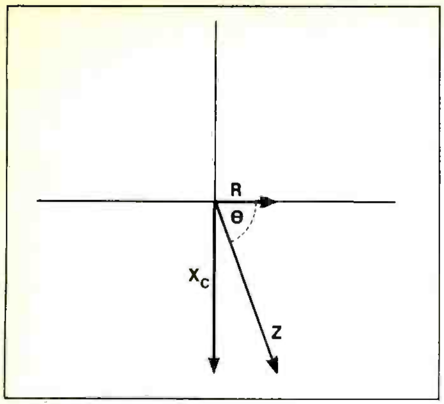


Figure 11

within the record/playback circuitry of audio tape recorders, with provisions for calibrating the record and playback frequency responses. One way in which a fixed frequency response circuit of the required type may be specified is by describing a resistance-capacitance network in terms of the R-C time constant such a circuit would have. (Remember: time constant = resistance in ohms × capacitance in farads.) A single simple R-C combination will not have the exact frequency response required for the entire audio range, but we can examine the frequency-dependent characteristics of an R-C circuit having a frequently specified time-constant, 50 μsec. (micro seconds, or millionths of a second) or 5.0×10^{-5} sec. We must first come up with the proper values of resistance and capacitance. The product of these values must equal 5.0×10^{-5} seconds. We can start by choosing a resistance of 100,000 ohms. What will be the value of capacitance necessary to make the time constant of the series combination equal to 5.0×10^{-5} seconds?

$$t = R \times C$$

$$5.0 \times 10^{-5} \text{ sec.} = 10^5 \text{ ohms} \times C$$

$$C = \frac{5.0 \times 10^{-5}}{10^5}$$

$$C = 5.0 \times 10^{-10} \text{ farads}$$

or

$$C = 500 \text{ pf.}$$

If we now place a resistance of 100,000 ohms and a capacitance of 500 pf in series, as shown in *Fig. 10.*, apply an alternating voltage at **a** and **b**, and calculate the voltage drop across R (V_r) as the signal frequency varies, we can plot the frequency response of the circuit. We will assume a constant value for the signal voltage, E_s = 10 v. Thus, the vector sum of V_c and V_R must

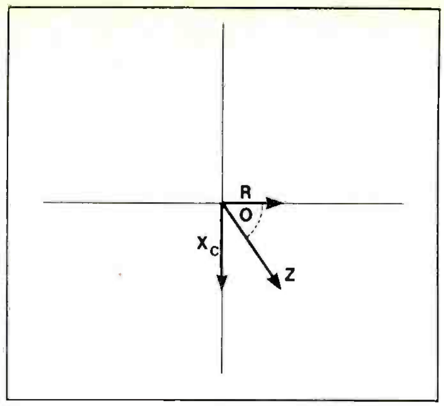


Figure 12

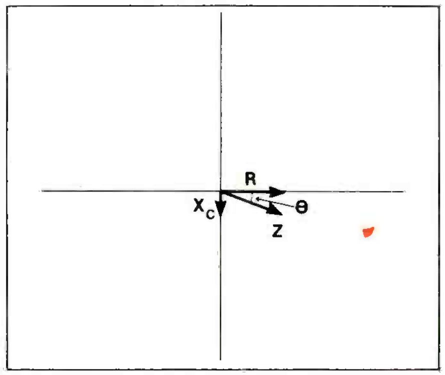


Figure 13

always equal E_s. We can determine the phase angle, θ, from the calculations for X_c. Let's pick three frequencies, one each in the bass, midrange and treble portions of the audio range: 100 Hz, 1000 Hz, and 10,000 Hz. First, let's compute X_c at each frequency. At 100 Hz,

$$X_C = \frac{1}{2\pi \cdot f \cdot a}$$

$$X_C = \frac{1}{6.28 \times 10^2 \times 5.0 \times 10^{-10}}$$

$$X_C = \frac{1}{3.14 \times 10^{-7}}$$

$$X_C = 3.19 \times 10^6$$

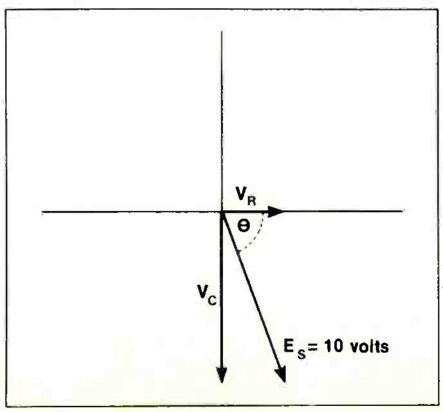


Figure 14

At 1000 Hz,

$$X_C = \frac{1}{6.28 \times 10^3 \times 5.0 \times 10^{-10}}$$

$$X_C = 3.19 \times 10^5$$

At 10,000 Hz,

$$X_C = 3.19 \times 10^4$$

Now we must find the phase angle between V_c and V_R at each frequency. We can do this by setting up the relationship

$$\tan \Theta = \frac{X_C}{R}$$

(See Figs. 11-13.)

For 100 Hz,

$$\tan \Theta = \frac{3.19 \times 10^6}{10^5}$$

$$\tan \Theta = 3.19$$

From "trig" tables or an electronic calculator,

$$\Theta = 88.2°$$

With similar calculations we can find that at 1000 Hz, θ = 17.7°, and at 10,000 Hz, θ = 1.8°. Using the information on phase angles we can compute and compare the voltage drops across C and R. We will do this with vectors, as shown in *Figs. 14-16.*

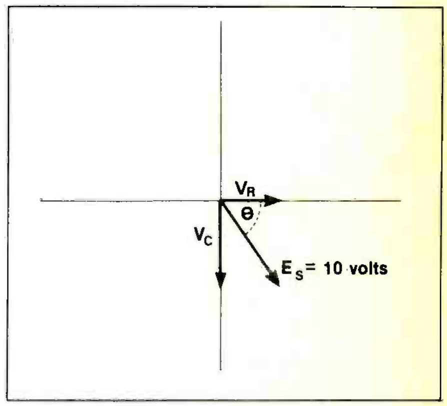


Figure 15

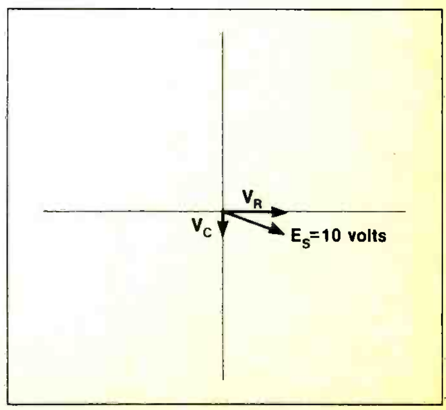


Figure 16

PERFECT HARMONY!

Peavey Electronics' transducer technology, in conjunction with our CS Series power amp program, has created what we consider the finest portable monitor package available to keep your onstage sound clean (and closer together).

The "heart" of the system features Peavey's new EQ-27 graphic equalizer combined with the field proven CS-400 stereo power amp to provide a system producing 400 watts RMS with twenty-seven bands of equalization. This combination of power and EQ enables intense sound pressure levels on stage while virtually eliminating feedback.

Add to this the M-400's special DDT® compression to maximize apparent headroom, transformer balanced inputs, electronic crossover, and bi-amp capability. The result is a highly versatile package with more than enough punch to get your vocals above even the loudest stage situations.

To deliver the M-400's punch, our engineers have designed a new series of enclosures with bi-amp capabilities built around our Black Widow speaker and the advanced 22A high frequency driver/horn combination.

The 1245 and 2445 monitor enclosures were created for optimum onstage monitoring, producing tremendous clarity, projection, and wide frequency response. Kick-proof grills and flite case type covers protect the units from damage and road abuse.

See the system at selected Peavey Dealers in your area or write us for more detailed information. You'll see one more reason why Peavey is ahead of its time,....and the competition.

PEAVEY®

PEAVEY ELECTRONICS CORPORATION / 711 A Street / Meridian, MS. 39301

CIRCLE 89 ON READER SERVICE CARD

For V_c at 100 Hz,

$$\sin 88.2° = \frac{V_c}{E_s}$$

$$.999 = \frac{V_c}{10 \text{ volts}}$$

$$V_c = \frac{10 \text{ volts}}{.999}$$

$$V_c = 9.99 \text{ volts}$$

Performing similar calculations with the values at other frequencies, at 1000 Hz, V_c = 9.54 volts, and at 10,000 Hz, V_c = 3.03 volts. Calculating V_R for the three frequencies using the formula

$$\text{cosine } \Theta = \frac{V_R}{E_s}$$

V_R at 100 Hz is 0.31 volts, at 1000 Hz V_R is 3.0 volts and at 10,000 Hz, V_R is 9.52 volts. As can be seen from the results of the calculations, at 100 Hz the greater voltage drop is V_c. At the other end of the spectrum, at 10,000 Hz, V_c is about 3 volts, but V_R is up (from 0.31 volts at 100 Hz) to 9.52 volts. *Fig. 17* displays the frequency response curves of the circuit just described by plotting relative values of V_c and V_R along the vertical axis versus frequency along the horizontal axis. The solid line represents relative values of V_c plotted against frequency, and the dashed line represents values of V_R plotted similarly. If an R-C circuit containing a capacitance of 500 pf and a resistance of 100,000 ohms is connected as shown in *Fig. 18*, it will provide the frequency response displayed as the solid line in *Fig. 17*. If the

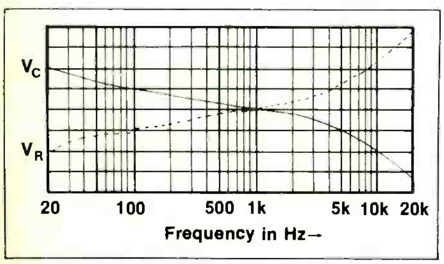


Figure 17

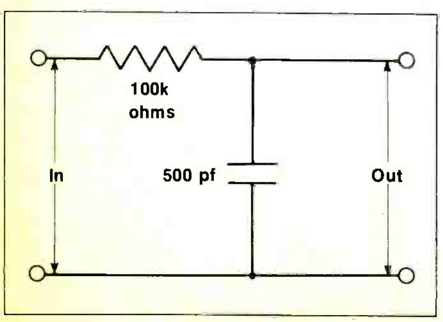


Figure 18

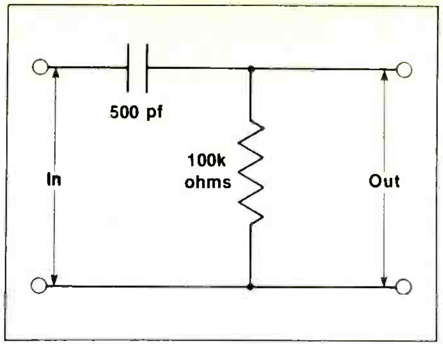


Figure 19

same elements are arranged as shown in *Fig. 19*, the frequency response, as measured at **c** and **d**, will be that represented by the dashed line in *Fig. 17*. The two frequency response curves shown in *Fig. 17* are complementary in the same way that the overall record and playback frequency responses of a tape recorder are complementary. In actual practice, different time constants are specified for different parts of the audio range. The N.A.B. tape recording equalization standard calls for a time constant of 3180 microseconds for the lower portion of the range and either a 50 or 90 microsecond time constant for the treble portion of the range.

The Condenser Mic

In order to explain the operation of a capacitor (condenser) microphone, we need to recall the three factors that determine the value of a capacitor: 1), the area of the plates; 2), the distance between the plates; 3), the material between the plates. A condenser microphone contains a capacitor made up of a heavy, stiff plate and a thin, stretched plate. The areas of the plates are fixed, and the same material —air—is always present between the plates. The thin plate is thin and flexible enough to be able to vibrate in response to sound waves. As the thin plate vibrates, the distance between the heavy plate and the thin plate changes at a rate equal to the frequency of the sound wave striking the thin plate. If this capacitor is placed in a circuit with a resistance and a charging voltage, the changing distance between the plates will cause the capacitance to vary at an audio rate. This change in capacitance will cause a changing current to flow through the circuit, and, consequently, a simultaneous change in the voltage across the resistance. This change in voltage across the resistance is an exact electrical picture of the sound wave that caused the vibration. An additional capacitor is used (C_B) to block the D.C. portion of the output signal. With the D.C. portion removed the output signal is a "pure" A.C. voltage that accurately represents the original sound wave. Before solid-state circuitry was generally available, one could always recognize a condenser microphone in a studio by the power supply to which it was connected. The power supply box (about the size of a small loaf of bread) provided a high (about 500 volts) D.C. voltage for charging the capacitor, and also provided power for a small preamplifier located inside the microphone case. The preamplifier was (and

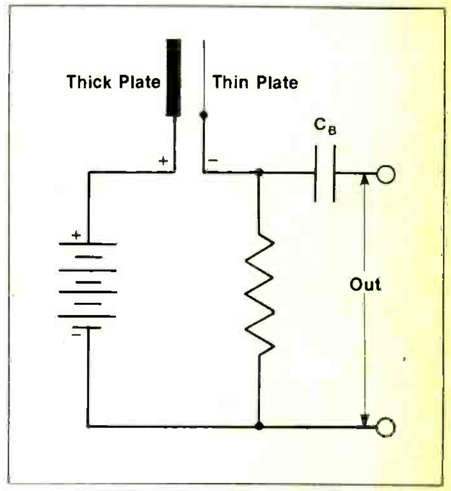


Figure 20

still is) necessary because the actual output voltage from a circuit like the one in *Fig. 20* is extremely low. If this low signal were sent directly to a recording console or tape machine, its actual level would not be much higher than the random noise present in the input circuitry of the console or recorder. Power for modern condenser microphones, which have solid state circuitry, can be provided either by batteries or a D.C. supply sent *to* the microphone over the same cable that carries the audio signal *from* the microphone. This last method is called "phantom" powering.

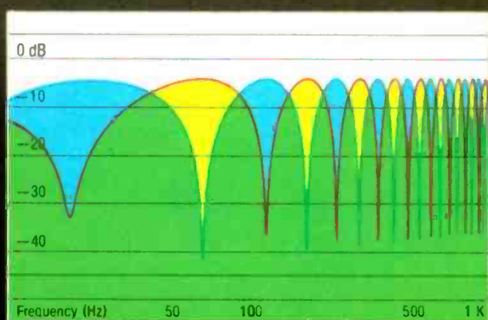
In the next and final installment of the "Electric Primer," we will discuss circuits containing capacitance, inductance and resistance, and apply that information to the design of a two-way speaker system crossover. In addition, we will examine the subjects of transformers and impedance-matching.

More than a chorus.

The new MXR Stereo Chorus stands alone among conventional chorus devices. Its unique design and sound almost defy description.

The Stereo Chorus utilizes advanced time delay circuitry providing a doubling effect for voices and instruments; one voice will sound like two singing in unison, two like four. As in nature, the voices can vary subtly in pitch. What you get is a natural choral effect. It can be used to thicken the sound, so that six-string guitars sound like twelve-strings. Through the introduction of extreme pitch bending, the Stereo Chorus will produce an intense vibrato, normally unattainable.

Not only is the Stereo Chorus musical and versatile, it also provides a means of achieving realistic stereo enhancement. We've included two outputs with complimentary notches and peaks in the frequency spectrum. The exact frequencies which are notched in one channel are boosted in the other. The graph shown is an actual response plot of the two outputs, which illustrates this concept. The Stereo Chorus transforms tones, harmonics and sounds into a lush and shimmering musical environment which surrounds the listener. This differs from similar devices which have one processed output, and one dry output. The Stereo Chorus provides a true stereo image which sounds more vibrant and alive than conventional chorus effects.



The manual control varies the delay time; a width control determines the amount of sweep; and a speed control adjusts the rate at which the delay is swept. A bypass switch provides noiseless accessibility to the dry signal in both outputs.

The MXR Stereo Chorus is equally geared for the studio or the road. We've included an internal switch which allows the selection of instrument or line level at both the input and output. Its high input impedance reduces the effects of loading when using long lines or other effects devices. Its low output impedance allows you to drive long lines and any other equipment. Superior circuit design has enabled us to maintain a wide bandwidth and dynamic range, ensuring signal fidelity.

The MXR Stereo Chorus is AC powered, ruggedly constructed to withstand the rigors of professional use, and is backed by MXR's reputation and commitment to the music industry. But reading about a product that has no equal is not enough. To appreciate what your music has been missing, see your MXR dealer.

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

MXR Musical Products Group

a session with THE BROTHERS JOHNSON

By Bruce Swedien

It has been four years and three platinum albums since the Brothers Johnson emerged on the recording scene. It all began when they received a call to play on a Quincy Jones session for "Q's" album *Mellow Madness*. Bassist Louis Johnson recalls, "The day that Q called, I had just gotten a gig working at a pizza parlor. I had to ask the boss for time off on my first day at work to go do the session with Quincy. I still don't know how I pulled that one off!"

These first sessions led to Quincy hiring George and Louis for his tour of Japan, and the Brothers' first album. That album, titled *Look Out For #1*, met with astounding success. It was certified gold within three months of its release and platinum (for sales of one million units) shortly thereafter. This first album also brought the Brothers a Grammy nomination as Best New Artist of 1976.

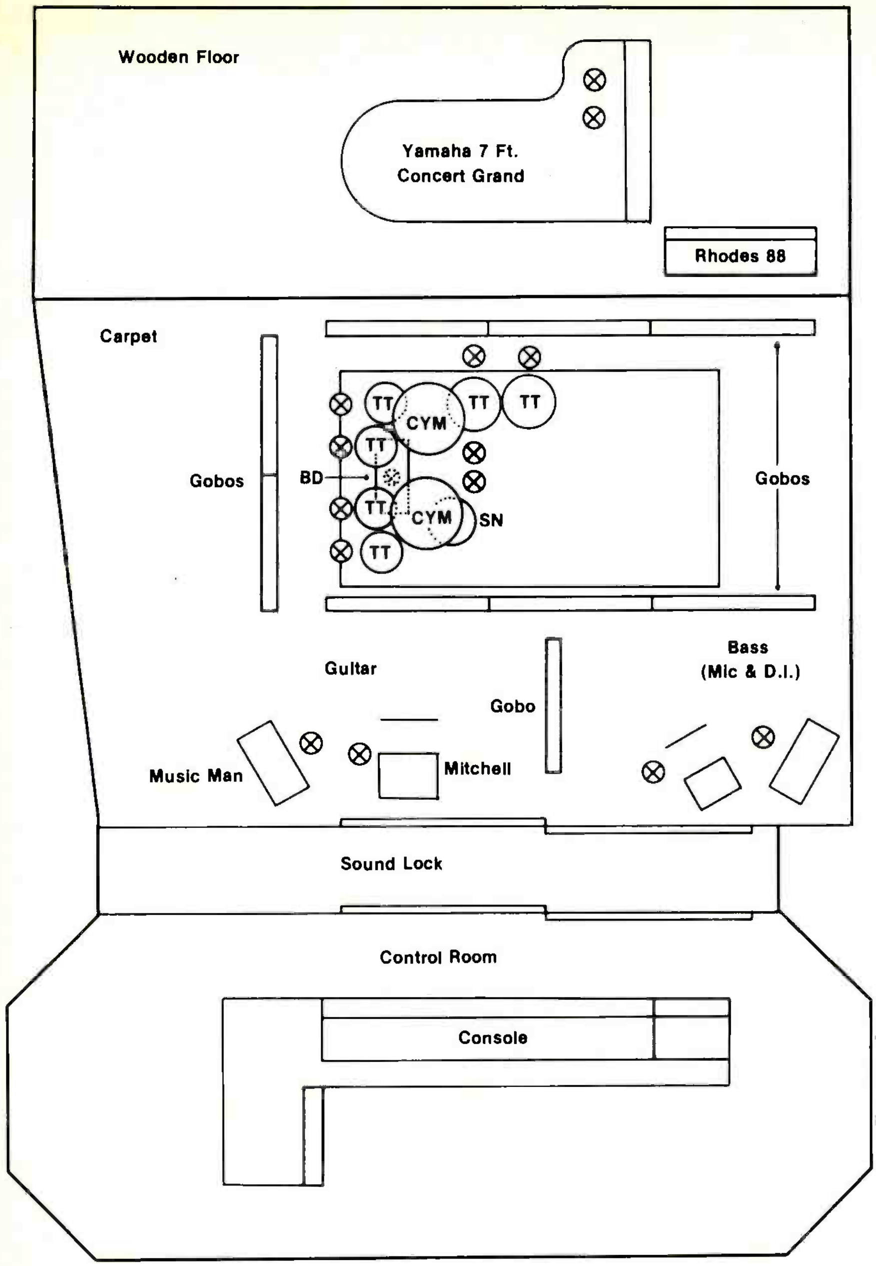
Their second album, *Right On Time*, hit the streets in the spring of 1977 and was certified platinum seventy-seven days after its release.

With the release of their third album, *Blam*, the Brothers faced the challenge of topping their previous achievements. Their philosophy has been to keep growing musically and to express as many musical influences as possible. That philosophy really worked on *Blam* because it was platinum in eight weeks time. That's a hard act to follow, let alone to top.

But top it they will! Producer Quincy Jones will see to that! George Johnson has been experimenting with new guitars and electronic effects. George's clean, articulate style has never sounded better than on this new A & M album titled *Light Up The Night*.

Louis Johnson has added keyboard synthesizer bass to his list of achievements. Both George and Louis share the composing of material for the album. Rod Temperton, of Heatwave fame, wrote two songs for the album and Michael Jackson wrote one song.

I chose Kendun Recorders new Studio D in Burbank for the album. Actually we have been using this studio for several months now. It is not only "apple-squeaky" clean and new, but it has every possible tool that a world-class, state-of-the-art recording studio should have. It also



gives us one more very valuable advantage that few studios have these days. Studio D at Kendun is the only studio in that particular building. It is separate and very private. This privacy is a big help to Quincy Jones and the artists that he produces.

The mixing console in Studio D is an SSL (Solid State Logic, Ltd. Oxford, England) 48 input 32 output console. It is totally computerized. The computer section of the console is set up to run all mixing and recording functions. That involves computer-memory mixdown and even tape function control. It is all done from a little computer keyboard in the center section of the console. The computer talks to the Studer A-800, 24-track, 2-inch master tape recorder and will implement any of the functions that the machine is capable of in any sequence you desire.

We made no special modifications to the studio to do the Brothers album with the exception of building a drum platform in the middle of the studio. This we made of two sections, each four feet by eight feet and eight inches high, making a finished platform eight feet square by eight inches above floor level. Drummer John "J.R." Robinson plays with a large set so this gave him plenty of room.

Bruce Swedien was the engineer on the Brothers Johnson sessions. He is a rare individual for someone with such an illustrious track record: he even returns your phone calls.

Getting to the Bottom

Recording Louis Johnson's unbelievably powerful bass style has been of real interest to me because I'm sort of a frustrated bass man myself. Louis has a nickname of "Thunder Thumbs." There is a real challenge in getting that tremendous sound clearly on the record. I tried all the conventional direct boxes, combinations of mic and direct pickup and all of these fell far short of his real sound.

One day while we were in the studio recording a bass solo with Louis, I put a high-impedence electronic volt meter across the output of his electric bass. I was amazed at how high the actual output level was. You could almost record his bass without a mic preamp. Looking at one of those puny little transformer direct boxes . . . no wonder there was something lacking. Those little transformers are able to handle about +8 dBm maximum level and apparently that isn't enough. I scrounged around my shop and in a big box labeled "to be organized" found a large, old transformer whose weight is measured in pounds not ounces. I made Louis a special "Thunder Thumbs" bass direct box. It has really worked well. I have tried the electronic "FET" active boxes but in an area of high rf and electrical noise, they are a bit undependable.

Louis has many different basses that he plays, including keyboard bass, all of them sound terrific through that big old transformer. Occasionally I mic his amp and mix a bit of that sound in with the direct pickup.

• • •

George Johnson seems to have a different guitar for every day of the year. His super-clear, rhythmical style is a pleasure to record. He is frequently seen in a corner of the studio changing strings on a guitar and tuning an instrument for one particular sound. Most of his recording, as is Louis's, is done in one or two takes, if it has the right feel, with few, if any, punch-ins.

I usually mic George's amplifiers, because that way any coloring he does with the amp will be on the tape. I use a wide variety of mics on him from a Shure SM-57 to a Neumann U-87. This depends on the effect we are after. Placement of his set-up in the studio is important. Studio D at Kendun has a "live" end with a high ceiling which we have used with great success for George's guitar solos.

Before recording the rhythm tracks for the album, all material is written and well rehearsed. Many changes are made during these rehearsals that would otherwise eat up expensive studio time. As it is, much revision is done during the rhythm track recording, but when the Brothers Johnson rhythm section goes in the studio the material to be recorded is well organized.

I have had the good fortune of working for twenty-three years with a friend—Quincy Jones. "Q's" musical qualifications require no explanation. Quincy's musical depth makes his work an audio engineer's dream come true.

Quincy's rich orchestrations and kaleidoscopic musical concepts have provided a real opportunity for me to develop an engineering concept that has its roots in all areas of music. In my work with "Q" we cover all types of sounds from groove funk to jazz and on to classical music. With such a wide range of sound sources to work with, it became obvious to me a few years back that standard multi-track single machine recording technique would not be enough. I began experimenting with "Mag-link" time code to run two multi-track (16-track) machines together in sync. This offered some real advantages, but since then I have expanded my system to use SMPTE time-code and two 24-track tape machines. At first the obvious advantages come to mind, lots of tracks, more overdubbing, etc. With a little experience I soon found that the real advantage of multiple machines—with Quincy's work—is that I can retain a lot more true stereophonic recording right through to the final mix. An additional major advantage is that once the rhythm tracks are recorded, I make a SMPTE work tape with a cue rhythm mix on it and then put the master tape away until the mix, thus preserving transient response that would be diminished by repeated playing during overdubbing.

Producer Quincy Jones during an unusually quiet moment in the control room of Studio D at Kendun Recorders.

Instrument List

GUITARS:
Gibson Les Paul
Gibson ES-35
Fender Telecaster
Fender Stratocaster
Guitar Amps:
Mitchell Pro 100
Music Man 212-8E
BASSES:
Music Man Sting Ray
Music Man Sabre Bass
Music Man Thunder Bass
All taken direct.
DRUMS:
Slingerland drums
Zildjian cymbals
Remo heads
KEYBOARDS:
Moog Prodigy
Oberheim 12-voice synthesizer
Prophet 5

Learning the Trade

It is during the rhythm sessions that the real professional approach to recording an album becomes apparent. The Brothers Johnson have obviously spent a good deal of time in the studio under the right leadership. Learning from Quincy Jones has given them an advantage that few other young musicians have been fortunate enough to have. They are indeed two young men, but they have had a great deal of good studio experience playing not only on their own product but on other artists' albums as well.

Louis and George Johnson are not fashionably late (at great expense) to the studio. Many times one of them comes one half-hour before session time to put new strings on an instrument or to rehearse a tough part.

Louis and George rarely accept a technically correct "first" take. Different tempos are tried to find just the right groove. "It's not acceptable till it's in the pocket." When a rhythm track is finished to everyone's satisfaction, Quincy will usually ask George to put a rough vocal performance on the tape. This gives a much more realistic idea of what the number really is going to be than if we evaluated it on the basis of a rhythm track alone. It is these rough vocals that give me a chance to try a new mic or a new gimmick to get an idea of how it will effect the overall image of a song. If my idea or new mic doesn't make it, I can depend on getting an opinion in chorus the next day.

After the rhythm tracks are finished and rough vocals recorded, I make a quick rough mix of each tune and then cassette copies are given to Quincy, George and Louis. The next few days (usually two to three days) I spend in the studio making SMPTE time-code work tapes. The master tapes are then put away and not played again until the mix.

• • •

The next step is to "sweeten" the tracks with vocal chorus backgrounds. This we did with L.A. studio singers, and being superb pros they completed all the vocal backgrounds in two days time (a total of four sessions). That's really incredible when you consider not one note of music had been seen by the

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24
BASS DIRECT	DRUMS KIT		KICK	SNARE	SOC (or hi-hat)	GUITAR mic	DIRECT	FENDER-RHODES		PIANO		GUITAR OVERDUBS 11-4-79 mic	DIRECT	BASS mic	SYNTHESIZER OVER-DUBS 11-5-79	BACKGROUND VOCALS BASIC		BACKGROUND VOCALS STACK		SYNTHESIZER OVER-DUBS 11-8-79	CLICK TRACK 11.6 FR CL	–	SMPTE –
																12-1-79		12-1-79					

LABEL: A+M RECORDS TITLE: "LIGHT UP THE NIGHT" REEL #3 M

ARTIST: BROTHERS JOHNSON DATE: 10-31-79 MACHINE: STUDER A-800

PRODUCER: QUINCY JONES ENGINEER: BRUCE SWEDIEN STUDIO: KENDUN D

NAB ☒ CCIR ☐ 30 IPS

DOLBY ☒ DBX ☐ 24 TRK

MASTER REEL

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24
LEAD VOCAL	LEAD VOCAL	SYNTHESIZER OBERHEIM		PERCUSSION BLOCKS + SHAKER		PERCUSSION CONGA		STRINGS		HORNS BASIC		HORNS STACK		LEAD VOCAL GEORGE TAKE 1	LEAD VOCAL TAKE 2	LEAD VOCAL OK TAKE 3	BACKGROUND VOCALS CUE TRACKS		BASS DRUMS GUITARS		CLICK TRACK 11.6 FR CL	–	SMPTE –

LABEL: A+M RECORDS TITLE: "LIGHT UP THE NIGHT" REEL #3 WT

ARTIST: BROTHERS JOHNSON DATE: 11-5-79 MACHINE: STUDER A-800

PRODUCER: QUINCY JONES ENGINEER: BRUCE SWEDIEN STUDIO: KENDUN D

NAB ☒ CCIR ☐ 30 IPS

DOLBY ☒ DBX ☐ 24 TRK

WORK TAPE - 'A'

singers before they got to the studio. The final product is note-perfect, flawlessly performed and sounds as though they had been in rehearsal for six months.

After the vocal B.G.s, the horns and strings are added. These parts are arranged and orchestrated by Jerry Hey from the group Seawind. The horns include two trumpets, one trombone and two saxophones. I have recorded many horn ensembles over the years—big bands, symphonies, soloists, etc. I have to say here that these guys are the best; I have never worked with better. They play with passion! Real heart, a quality rarely found in musicians that spend so much time in the recording studio, is very evident in this marvelous group. (The strings are eight violins, four violas and four cellos. This string section is made up of the best of L.A. studio players.)

Next, final vocal leads are recorded and instrumental solos and so on. The last step is to add percussion and synthesizers. The synthesizers on the album are programmed by Steve Porcaro from the rock group Toto. Steve's equipment comes to the studio in a moving van. The men who set up Steve's equipment arrive two hours before session time and are usually still carrying something into the studio when we roll the tape. On this project Steve's synthesizers actually lined all the walls of the studio and filled the control room. We ended up using them all at some time or other, too.

The synthesizers are performed by three musicians: Gregg Philingains (Gregg also performed all other keyboards on this album); Larry Williams (Larry also plays saxophone, flute and piccolo on this album; he is a member of Seawind); and Steve Porcaro.

Final Steps

Everything is finally on the multitrack tapes. Next step, mix forty-two tracks of audio on two 24-track machines. More on a couple of the tunes. Fortunately, I did my homework right and it goes quite smoothly (this time, anyway!).

I mixed the album onto a Studer A-80 two-track at 30 ips using 3M 250 tape and a reference level of 250 nWb/m at 1 kHz. This makes an extremely clean, clear two-track master tape with very respectable signal-to-noise ratio. It also, I feel, eliminates the need for noise-reduction equipment at this stage. That is a big plus as far as I'm concerned.

The last phase that I can be personally involved in is the mastering of my two-track tapes to the disc master. This will eventually become the stamper from which records can be mass-produced. For this I spend a couple of days with Bernie Grundman, the man in charge of disc mastering for A & M Records. Bernie is a real artist. His reputation in the industry is gold-plated, indeed it is platinum-plated! In my experience in the industry I have found very few individuals who can do what Bernie can. We sit for hours with the tape and resultant lacquer reference discs and make sure that the volume level and frequency curve for each song in the album is exactly right. After this final check, the record is out on its own and its fate is in the hands of the listener. But the Brothers Johnson have never worked harder on an album, so we know *Light Up the Night* can only meet with success.

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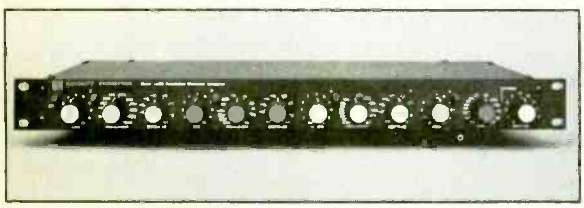
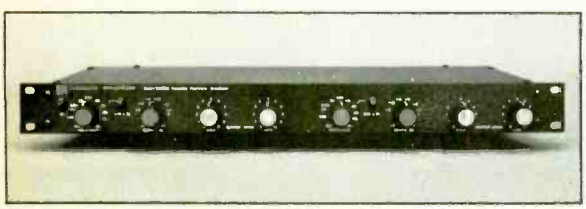
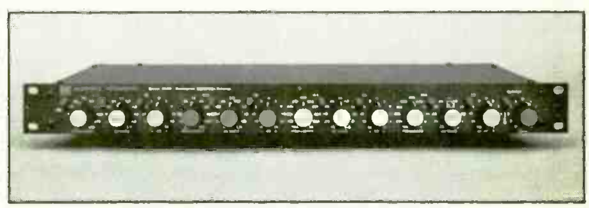


Model 1400 Parametric Electronic Crossover

Model 1500 Feedback Suppressor

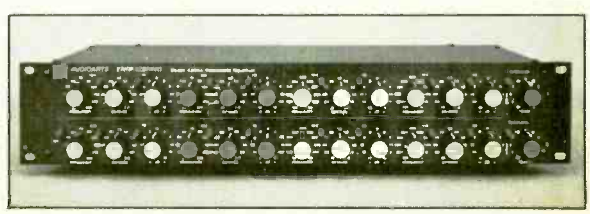


Model 2100A Tuneable Electronic Crossover



Model 4100 Parametric Equalizer – Preamp

Model 5200A Stereo Mixer/Preamplifier



Model 4200A Parametric Equalizer

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Profile:

JOAN ARMATRADING

By Jeff Tamarkin

Joan Armatrading recently released her seventh American album, Me, Myself, I *on A&M Records. She has been critically acclaimed on both sides of the Atlantic, although she has yet to "break" commercially in the U.S. Armatrading considers herself a songwriter first and a performer next; however, her distinctive style, which mixes elements of R&B, reggae, jazz, folk and rock music, places her in a category which is really no category at all. Joan Armatrading simply sounds like no one else.*

During a recent stop in New York, between the releases of a four-song EP entitled How Cruel *and her new album, Armatrading spoke with* Modern Recording & Music's *Jeff Tamarkin about her experiences in the studio and in concert. She can be described as a reserved person, and is not nearly as interested in talking about her personal life as she is in talking about her music. Having worked with top producers such as Glyn Johns, Gus Dudgeon and Richard Gottehrer, who produced the new album, as well as having produced herself, Armatrading has gained a wealth of experience in the studio during the past several years.*

Modern Recording & Music: First, how did you get started as a singer? What kind of background do you have?

Joan Armatrading: [Long pause]. Let's just start with the first album I made, which was in 1972. Then I made another one after that and another one after that. *[Laughs].* Talking about that is boring.

MR&M: O.K. then, what was your first experience in the recording studio like? Were you frightened by the sight of all that equipment surrounding you?

JA: No, the first time was really good, actually, because the first album I made was with Gus Dudgeon. I made the first album in France. We went into the studio in England first, and we did three songs as a demo. That was just his way of introducing me to the studio. It wasn't a big awesome place or one of the most used studios. It was pretty comfortable for me to get used to. We just did bass, drums and me playing the piano. We did two piano songs and one guitar. That worked out really well. Then we went over to France, and that was quite impressive because it was a live-in place. The room that I had was a huge room with lots of fancy stuff in it. The studio was great, but every day, maybe every two hours, something would break down. That wasn't too good.

MR&M: When you look back to your first recordings now, do you see them as a stepping stone to where you are now?

JA: I'm pleased with all the albums I've done. Obviously, you end up liking one more than the other. I usually like the one I'm about to make.

MR&M: Had you ever worked "live," with a band, before recording?

JA: No.

MR&M: How did you find the adjustment to performing on stage after having been a studio artist?

JA: The weirdest part about that wasn't so much getting the "live" performance together, it was getting a band together. Because the band I had after making the first album wasn't the same one that was on the album. That was the strangest part, because I had to audition all the people. And all of the people who were coming to audition were obviously much more experienced and so much better than I was. They knew much more about everything than I did. It was weird to be telling all these people what I wanted. It was different than making an album, because there I had Gus to help me. But (when I auditioned people) it was just me telling them what I wanted. Then came the rehearsals together, and trying to make sure that it all worked out. That was all hard work. And, obviously, being on stage was just something else. That was totally new.

MR&M: Who were some of the musicians you worked with at that time?

JA: The only ones you'd know are Davey Johnstone and Ray Cooper (of Elton John's band).

MR&M: How was Glyn Johns as a producer?

JA: Glyn? Oh, great. I learned a lot from everybody.

MR&M: You produced the *How Cruel* EP by yourself, along with Henry Lewy, and now you're going back to using a producer (Richard Gottehrer). Do you prefer producing yourself or having someone produce your records?

JA: It's a weird thing when you talk about producers. It depends on the artist. With some people, the producers *obviously* produce. With other artists, the producer *helps* to produce. Do you know what I mean? Because, I like to go in prepared, and I like to go in and know what I want, and what I want the musicians to play. I try to get that across first. Then the producer helps me to get what I'm trying to say. Then he puts in some of his ideas as well. It's not a matter of me coming in with a vague song and seeing what happens with it.

MR&M: Then, you always work out the material before you enter the studio? Some artists prefer to arrange the material once they're in the studio.

JA: When I write a song, I like to know that I've written it. And when I hear it, I like to know that it's my song. I don't want to hear little bits in it and think, "Oh, I had nothing to do with that."

MR&M: Has a producer ever taken one of your songs and suggested to you that you completely rearrange it?

JA: No.

MR&M: So, actually, in a way, you've co-produced all of your records. Why then does your name appear as a co-producer only on *How Cruel?* Did you have the same amount of production input on your other albums?

JA: It's the same amount, yes, The thing with Glyn, for example, is that Glyn engineers as well, and engineers are a very important part of making a record. He's the one who's got to get the sounds together. The general sound of the record is really important.

MR&M: Do you work closely with the engineers?

JA: Um. . .I do, but then it gets too "fussy." So, whatever I'm trying to get across, I tell to the producer or the musicians. I might say one thing to him [the engineer] and then the producer might decide that it would sound better like this. So if I say it to the producer and then he relays it to the engineer, the engineer won't get too confused. I get my bit first, with the musicians, and if there are any changes, I'll know about it as well.

MR&M: Then, do you prefer to just enter the studio and record, or do you like to keep your eye on what the engineer is doing?

JA: If you've got a good engineer, you don't have to worry. If you've got someone you don't feel is too capable, you might want to be in there and listen to everything he's doing. But I've worked with such good engineers that I don't worry what's going on. And the producer is in there, so he's keeping an eye on everything. It's enough to be sitting with the musicians, trying to get the music together.

• • •

MR&M: When you perform in concert, do you like to check out the hall before playing in it?

JA: I like to spend as much time in the place as I can, just getting the atmosphere. But my sound man will get down there early and check it all out.

MR&M: Have you ever played a place where the sound was just truly horrendous?

JA: *Ha-ha-ha!* How many do you want to know about? Yeah, that happens, and if it's a really bad place, there's nothing you can do about it. You just have to play and hope that it comes across. The worst places to play are the places where they say it's acoustically perfect. As soon as they say that, you know it's going to be awful. Those are the places where they have symphony orchestras and operas. Then, it's acoustically perfect. But, for electric instruments, it doesn't really work.

MR&M: What do you demand of a producer? Do you give him specific instructions before you begin work?

JA: That he doesn't take over my

work. *[Laughs].* You might have somebody who's just too eager to put his stamp on something. I couldn't stand someone who'd say "That song sounds great, but only if you had that verse there, and that chorus there." By the time you've finished, he's totally rearranged it and it's not your song. So, the first thing I say is try to remember what's mine. And then, just that he can be able to talk to the musicians. Producers who have been doing it a long time have a way of knowing how to talk to the musicians, and the engineers. And they know how to find good players for what you want. I like to work with different people. Unless there is somebody who I know that I want, I'll usually leave it up to the producer to find the people that we need. I find that different musicians work better on different songs. Sometimes you find that the people you work with on the road won't be able to adapt if you want to change in the studio. They're used to you saying this is how I want it, and when you go into the studio, it's hard [to change]. I've had that happen.

MR&M: Do you prefer "live" or studio work now?

JA: I prefer the studio. I don't prefer the singing, but I prefer getting the music done. It's the first time I'm hearing the song the way I wanted to hear it, and that's nice.

• • •

MR&M: In an earlier interview, you said that you preferred writing to performing. Does that still hold true?

JA: That hasn't changed. I really enjoy writing songs. But I enjoy performing a lot more now. I really enjoy myself on stage these days.

MR&M: You've played to some huge audiences in England, at outdoor festivals to thousands of people. What is the difference to you between playing a show like that and doing a club or theater concert?

JA: Big audiences can be weird. When we did Blackbush, with Bob Dylan, there were 250,000 people there. But there might as well have been two people there. Because who do you play to? You just try to have as good a time on stage as you can. I went outside and there's nothing to see. You just have to enjoy the person who's sitting next to you. In a theater, with 2,000, to 5,000 or 7,000, it's really nice because you can still feel as though everyone can see you and everyone is pretty near. I don't like clubs, really, because then I feel too closed in. It's also really uncomfortable on stage for the musicians, because the stages are usually too small. By the time you've got the drums set up, you're practically sitting in the audience.

MR&M: Have you ever run into the problem of a musician whose ego was bigger than it should have been, who just *had* to get in that solo even though you didn't want it there?

JA: Never in the studio. The only time that ever happened to me was when I was playing in England, and a guy on saxophone, who wasn't really in the band but had played on the record, obviously thought a lot of himself, and did a great solo. He got paid for the gig, of course, but the audience kept wanting encores. We did the one, and then he said, "The only way I'm going to go back out there is if you pay me ten more pounds." *[Laughs].* So I said who needs that? That's definitely ego.

MR&M: So, I guess he didn't do the encore.

JA: Oh, he did it, but he didn't get his ten pounds.

MR&M: What models of guitars and other equipment do you use?

JA: I used to use a Yamaha guitar at the beginning, in '72. I used to use a Barcus Berry pickup, which kept falling off, and the glue kept sticking to my hand. Then I went into a shop looking for a new guitar. I was trying to find a Martin and the shop introduced

SELECTED DISCOGRAPHY

Whatever's For Us	(1973)	A&M 4382
Back to the Night	(1975)	A&M 4525
Joan Armatrading	(1976)	A&M 4588
Show Some Emotion	(1977)	A&M 4663
To The Limit	(1978)	A&M 4732
How Cruel (EP)	(1979)	A&M 3302
Me, Myself, I	(1980)	A&M 4789

me to Ovation. At that time, there was hardly anybody using them. That's pretty much all I use. They [Ovation guitars] work really well. I use Bose speakers and a little amp. I used to use an HH but I don't know if I still use it. I've got a guy who looks after my stuff and he just swaps things around and I never know what he's doing. But it gets a good sound. It stays pretty acoustic-sounding.

MR&M: When you play a big hall, do you find that the natural sound of the Ovation projects, even to the rear of the hall?

JA: Oh yeah. It's a really good system, and it has a volume control on it. And I bought a stereo Ovation as well.

• • •

MR&M: Do you have any theories as to why the United States hasn't been as supportive of your music as England?

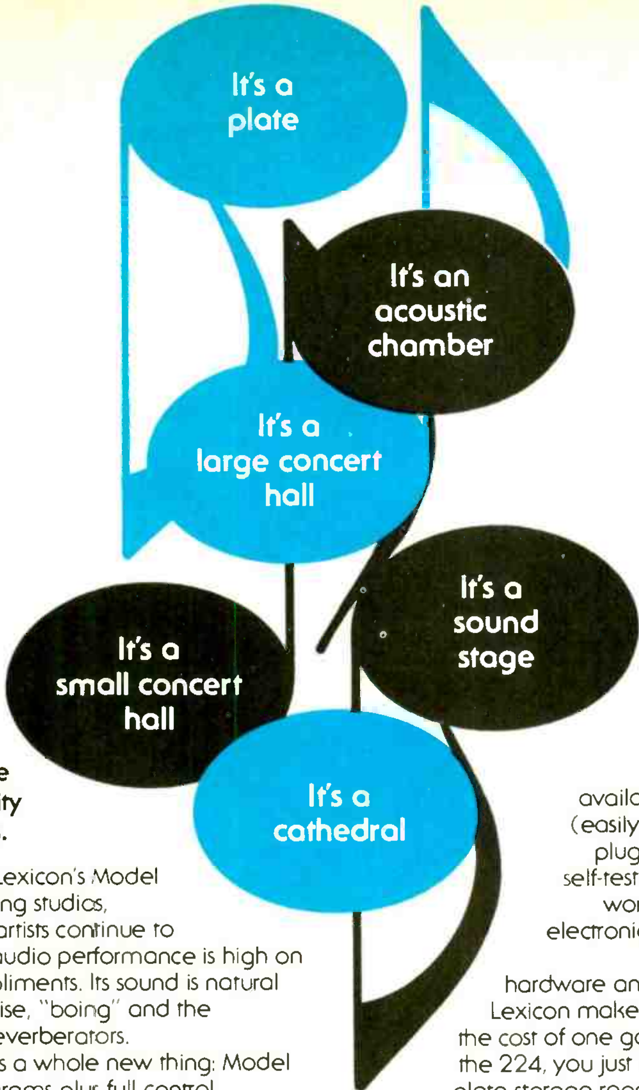
JA: I don't know. *[Shrugs].*

MR&M: What was the reasoning behind releasing the *How Cruel* EP? Why didn't you wait and release that material on your album, *Me, Myself, I?*

JA: I did two songs and Jerry Moss (of A&M Records) heard them and said, "Why don't we do another two and put out an EP?" I thought it was a good idea, and I still think it was a good idea. The only thing is that in the U.S., people don't seem to know very much about EPs. They got really confused. And the other thing that confused them was that all of the songs were on one side, and the other side was empty. And it was all on a 12-inch record, so the size confused them. Disc jockeys didn't know what it was, either. They thought it was a sampler for the next album.

MR&M: Is there a central theme which runs through your songs?

JA: *[Long pause, then laughter].* I was gonna say life. . .*[Laughs].* I don't know; what do I know!



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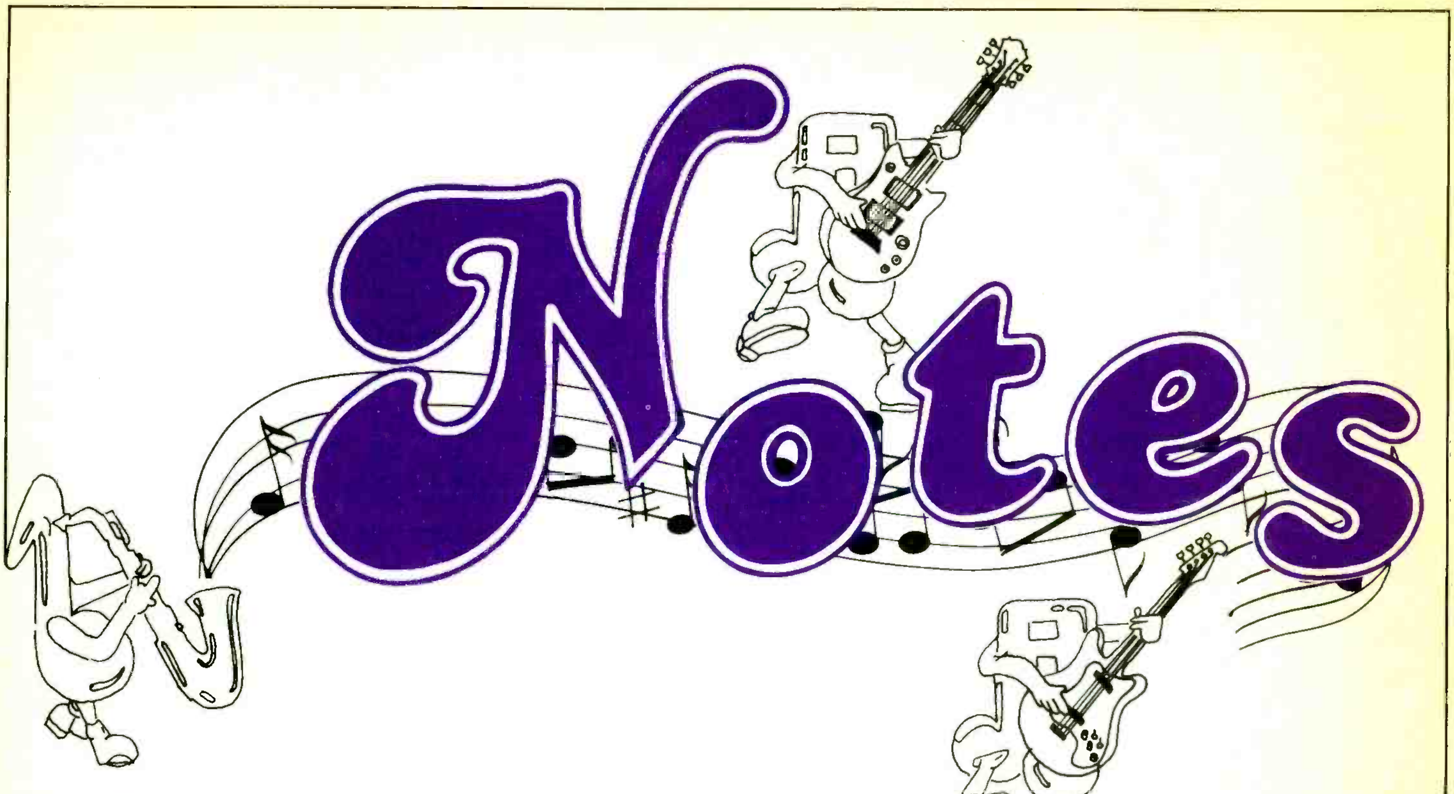


MXR Pitch Transposer

By Craig Anderton

Over the past few years, pitch transposing (or harmonizing) devices have undergone subtle, and not so subtle, improvements. Whereas the first generation of these devices was very expensive and marred by audible glitches that detracted from the audio quality of the transposed signal, the newest generation of transposing devices are less costly, more natural sounding and easier to use. One of the current entries in the pitch transposing sweepstakes is by MXR, a company which is mostly known for its effects boxes but which also manufactures a number of studio-oriented "high-end" pieces of equipment. The Pitch Transposer (or PT for short) is the newest of this line.

WHAT IS IT? The PT is an attractively packaged, rack-mounted unit that has been designed for both studio and "live" use. It adds one of four user-selectable harmony lines to any input signal; this harmony can vary continuously from slightly more than an octave below the original signal to slightly more than an octave above, and may be mixed back with the original input signal in any desired proportion. Also note that the PT can accept chords as well as single notes; in fact, I feel that the PT's effect is most dramatic when used with complex audio signals . . . but more on that later.

Figure 1 shows a simplified block diagram of the unit. There are basically four modules, which are:

1.) ***Input Module***—In the PT, the input module is very flexible and can accommodate a *wide* range of unbalanced line signals. There are two independent sets of jacks for inputs and outputs. One set is located on the front panel and is intended for low-level instruments such as guitar, electric piano, and the signals associated with −10 dB studios. The other set is on the rear of the unit, and is designed to accommodate line levels. Both sets of jacks may be used simultaneously, and the front panel set has an associated high/low switch to match levels even more precisely. A level-setting LED located directly over the front panel high/low switch simplifies level-setting; you just use whichever jack pair and high/low switch settings cause the LED to flash on the peaks of your playing.

One interesting note about the high/low switch is that the overall level does not change as you change the switch setting. If you add more gain (the *low* position, corresponding to *low* level signal input), you simultaneously add more output attenuation to compensate. This helps keep noise down while preserving unity gain through the PT.

2.) ***Pitch Shift Module***—This section actually creates the transposed line. There is a regeneration control for adding special effects to the shifted pitch.

3.) ***Preset Controls***—These are four highly interesting knobs that allow you to preset four different pitch shift intervals. I say "highly interesting" because each knob has a black faceplate which, when touched, activates that preset and lights an associated LED. If you grab the knob to turn it, your finger comes into contact with the knob's faceplate and selects that preset. This is a nice touch (sorry about the pun, but I couldn't resist) and is one of the PT's many "bells and whistles." However, being left-handed I couldn't help but notice one problem: when reaching for the mix knob, my left hand would often brush against the preset 4 knob. I don't think this would be a problem for right-handers, however.

These preset controls also have a priority system, with position 1 having the highest priority and position 4 having the lowest priority. What this means is that if you touch knob 3 continuously to select that preset and then touch knob 1 *at the same time,* 1 will be selected since it has priority. On the other hand, if you touch 4 at the same time as 3, 3 will stay selected because it has priority. Got that? To explain why this is useful, we need to investigate the next module . . ..

4.) ***Preset Footswitch Selector***—This is not included with the basic PT package; it costs an additional $50, and consists of a floor effects size box with four preset footswitches and a bypass footswitch. Thanks to the priority system described above, when doing harmony lines you can hold the preset 2 button down with the heel of your foot, then use the front of your foot to select preset 1. Even though you have both switches pushed, 1 will assume priority. When you've finished using that interval, rock your foot back so that you're just pushing down preset 2 again. Since in many harmony line situations you want to alternate between something like a major 3rd and a minor 3rd, the preset footswitch selector allows you to do just that—and easily, as well.

5.) ***Output Module***—This has a mix control that is continuously variable from dry signal only at the extreme counterclockwise position, to effect signal only at the extreme clockwise position. A bypass touch switch (and associated LED to show when the effect is happening) resides towards the left-hand side of the front panel. This bypass switch is not noiseless, however, and does contribute a small audible click when activated or de-activated.

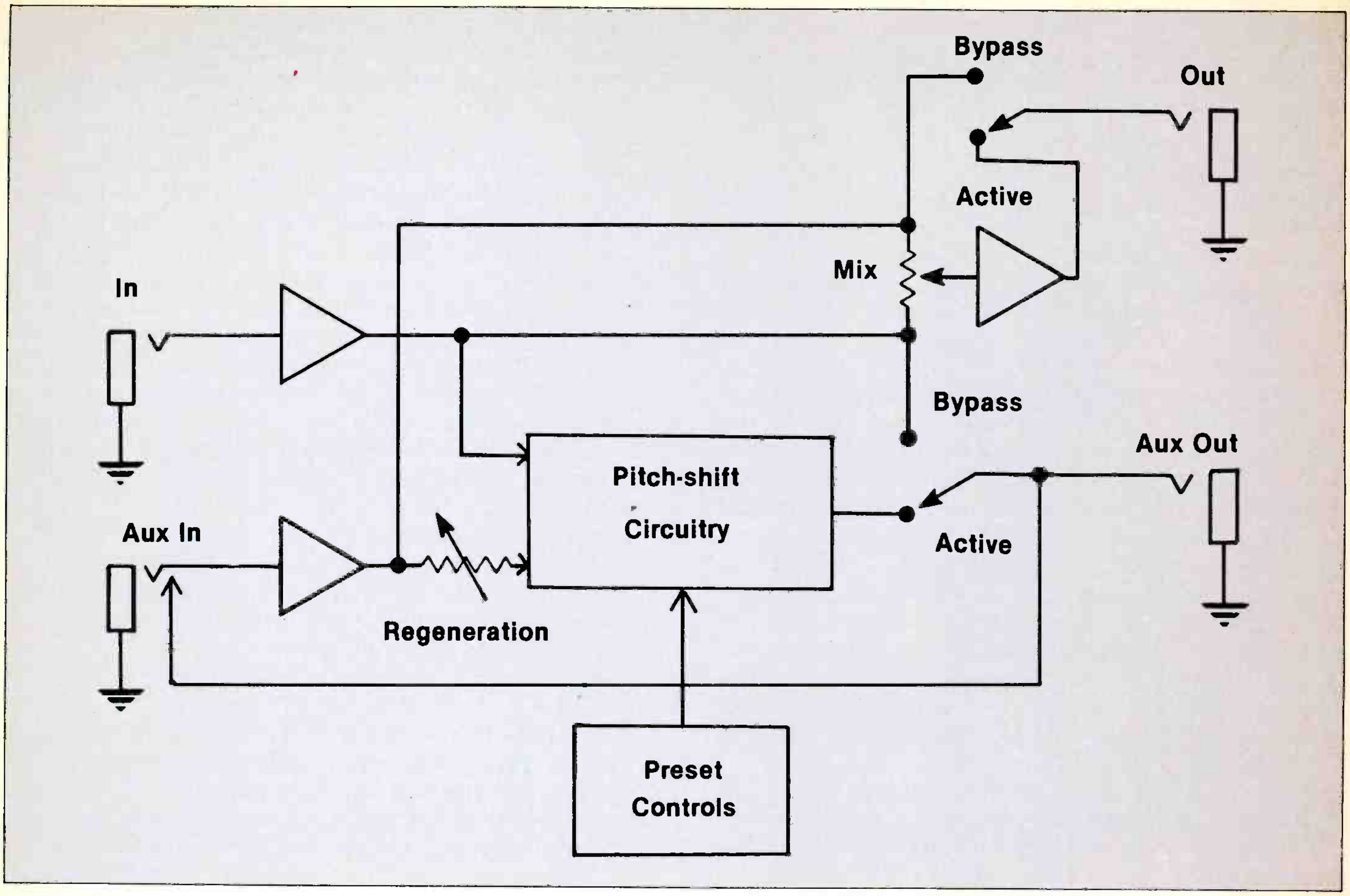
6.) ***Pitch Transposer Display Module***—This is another option for which you pay an additional $300. At first, I thought it was an overpriced frill . . . now I'm not so sure. The display allows you to read out either the mathematical interval, or the number of half-steps, separating the transposed pitch from the original signal. While not necessary for operation of the PT (which is no doubt why it's offered as an accessory), it does decrease setup time in the studio, and allows for "perfect" tuning—even by someone who is tone deaf. So, if you've got a big studio and some bucks, the display is probably worth owning if only for the mystical prestige that flashing seven-segment LED readouts can give a piece of equipment.

The display is also useful "live," since you can make rapid changes without having to listen to see if everything is properly in tune. However, the fact that you have four available presets for instant selection seems to make the display unit less of a necessity, unless you need to be able to select more than four presets in a very short period of time. The more impoverished among us can always tune the PT by ear if the display crimps your budget.

All in all, for ease of use, logical operation and clean layout, you'd have a hard time improving on the PT; MXR has obviously put a lot of thought into making this device easy to use. Its adaptibility to either studio or "live" use is another strong point in its favor. But then again, there's the price to consider: at $800 list price, the PT represents quite an investment, and you should get more than intelligent packaging for those kinds of bucks. So, let's see what it's like to sit down and evaluate its performance.

PRE-FLIGHT for the PITCH TRANSPOSER. If you have the display unit, hook it up first, then turn on power. Next, decide whether you're putting out an instrument level or line-level signal, and plug into the appropriate pair of jacks. Play your instrument, tape track or what have you for a while and observe the level-setting LED. If you're plugged into the front panel jacks and the LED glows too often, change the level switch to the *high* position. If it doesn't glow, change the level switch to *low.* If you're plugged into the line input jack and the LED does not flash at all, try the instrument jack instead. Conversely, if you can't avoid distortion when plugged into the instrument jacks, plug into the line-level jacks on the back.

With the regeneration control all the way off, and the mix all the way to dry, you should hear your straight instrument sound. Now select preset 1 by touching its associated knob, and put the mix knob at the halfway position. You should hear a harmony line that changes as you vary the



preset 1 knob; if not, chances are you're in the bypass mode. Touch the bypass switch, and its LED should light up to show that the effect has been selected.

EVALUATING the HARMONY SOUND. Let's ignore the various preset options and regeneration at first to keep things simple, and concentrate on listening to the harmony line. As you might expect, since this harmony line is being synthesized from the original signal there are some audible splicing sounds and differences when compared to the original signal. The transposed signal has the following anomalies:

It is noisier than the dry signal. A linear noise gating circuit (it sounds similar to compansion-style noise reduction) helps keep this under control, but unfortunately, the gating action is not truly linear. A guitar string, for example, will decay faster than the noise gate decay curve, leaving the noise unmasked by a higher signal level. This is when the noise is most noticeable. Any noise pretty much disappears when playing chords and the like; it's most audible on single notes that decay slowly.

There is a slapback echo effect on the transposed line. As you increase the pitch of the transposed signal, the delay time appears to increase somewhat, producing a longer echo (this might be a subjective opinion) effect; the echo time appears to be much shorter at decreased pitch settings). In any event, with high harmony lines (such as an octave above), there is always a kind of tight slapback echo, whether you want it or not.

There is a tremolo kind of effect. When the pitch selected is the same as the original pitch, the "tremolo" rate is very slow. It increases in frequency as you go either up or down in pitch, until it is running at a fast sub-audio frequency. (In other words, it's fast enough so that you can't count the individual cycles, but it's also slow enough so that it doesn't become an audio tone.)

Now, all of the above might sound discouraging; but while these glitches are indeed audible, they are really not bad and do not impair the musical usefulness of the PT to any great degree. For instances where these glitches do get on your nerves (and it can happen), simply setting the mix control for less effect signal usually solves the problem. It's amazing how little of the transposed line you need to mix in with the dry signal in order to get the full effects of pitch harmonization. For example, the octave above sound is pretty much the worst case condition for the PT; but even when mixed at a low level, the effect is still noticeable. So, while a perfectionist might quibble with the above-mentioned glitches, they are not really objectionable in many, if not most, applications. Electronically speaking, it's amazing that something like a pitch transposing unit works at all—and the MXR unit does work well.

ADDING REGENERATION. For special effects use, the regeneration control adds great amounts of color and interest. With the transposed pitch set very slightly higher than the normal pitch, advancing the regeneration control initiates a stepped, upward glissando effect (where each

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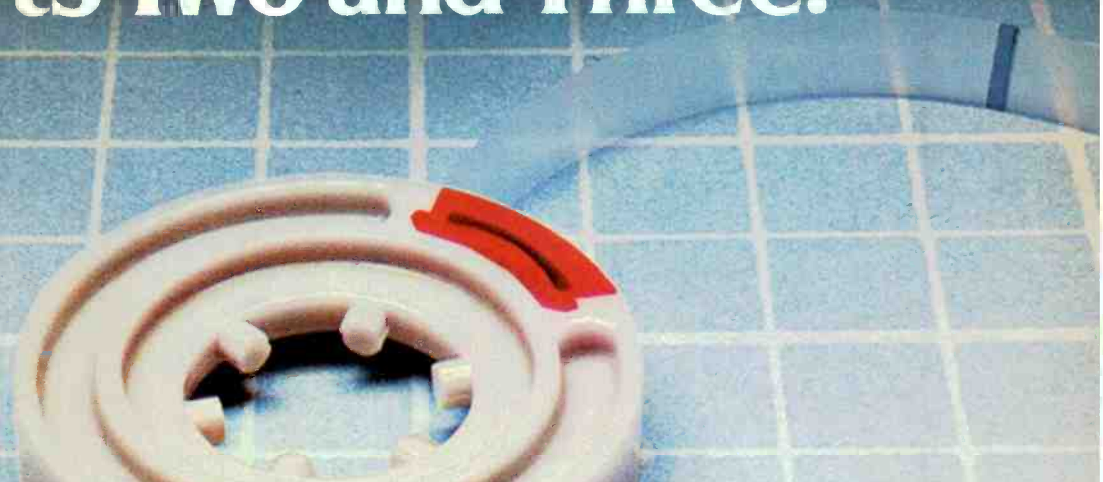
Parts Two and Three.



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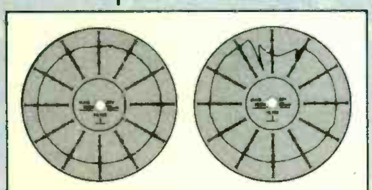

Circularity comparison diagrams.

These imperfections are exaggerated by second-rate clamps. Some manufacturers jam a pin into a notch to secure the tape. The result is a dip that's passed on through successive tape layers. A hub/clamp assembly off by as little as the thickness of this paper can multiply problems in a dozen layers of tape.

TDK uses computer-designed molding equipment and the very best materials to produce a unique 45° "W" double clamp with a special double purpose. The inner surface secures the leader flawlessly to the hub. The outer clamp section completes the circle, which is then checked for roughness and circularity on a machine that enlarges 100 to 10,000 times. As a result winding is always precise. Tape is off to a smooth start. The colored clamp acts as a moving tape indicator. You'll see tape direction and running stability at a glance.

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The Amazing Music Machine.

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step interval is the difference between the transposed and initial signals). Setting the transposed pitch slightly lower than the original signal produces a similar downward glissando. These effects are truly striking; I particularly like using the downward glissando in conjunction with single note solos. Thus, as each note ends, it sounds like it's spiralling downwards—most interesting. Holding a single note with the regeneration advanced, and then varying one of the pitch controls, produces a wide variety of interesting effects: from pseudo "bell trees" (seemingly endless upwards or downwards spiralling) to complex flanging-with-pitch-change effects. Regeneration changes the scope of the PT from being simply a pitch-transposing device into filling the role of an excellent special effects generator.

You can expand the limits of the regeneration option through the use of two additional jacks on the back, labelled "Aux In" and "Aux Out." These interrupt the regeneration path and allow you to add signal processing to the regenerated signal; what's more, there is a switch for matching the level of the regeneration path to either "instrument level" processors or line-level processors. If regeneration is indeed the key to getting great special effects out of the PT, then the Aux In/Out jacks are the key to making maximum use of the regeneration option. For example, by delaying the regeneration signal through an echo unit, you can slow down the speed of the glissando mentioned above. With a delay of, say, 0.5 seconds, and a pitch setting that is a half-step above the original signal, hitting a note will result in an upwards glissando that goes up by one half-step every 0.5 seconds. One setting I particularly like is adjusting the pitch control for a signal that's a fifth above the original signal, and adding some delayed regeneration. When you hit a chord, you hear a fifth above that chord a fraction of a second later (depending on the amount of delay), then a fifth above the second chord, a fifth above the third chord and so on until the regeneration dies out completely. Note that you only have to hit one chord to initiate this chain of chords—the PT takes care of the rest.

Another favorite patch of mine is to set the transposed pitch an octave above the original signal, with a slight amount of regeneration. This creates an almost string-synthesizer type of sound, even when added to an instrument like guitar. When applied to a true string synthesizer, the string simulation becomes even more life-like.

USING the PITCH TRANSPOSER with VOICE. Many people who are initially excited by the idea of pitch transposition devices want to use these to add harmony lines to vocals. However, this presents some problems since the timbre of the voice changes along with the pitch. For example, if you set the pitch for an octave above the original signal, the voice will sound as if you had recorded it at 7.5 ips on a tape recorder and played it back at 15 ips. So, unless your idea of a vocal harmony is to have Alvin and the Chipmunks singing along with you, the PT is of limited usefulness. The PT is excellent for adding doubling/flanging type effects by offsetting the pitch by a very small amount; however, a doubler/flanger will do the job just as well for less money, so the PT's doubling capabilities are more of a nice added feature than a primary reason for looking into the PT.

A less obvious use of the PT, but one which would be of great interest to salespeople, students, interviewers and other people who need to rapidly process large amounts of spoken material, is as a speech time-compression device. In this application, you can feed a microphone (or spoken program material) into the PT with the pitch set for one octave *below* the original signal, and then record this on a tape recorder running at, say, 7.5 ips. By then playing back the tape recorder at 15 ips—or exactly double the speed—the timbre of the speech goes back to normal, but the *rate* of speech is doubled.

OVERALL EVALUATION. Pitch transposition devices are musically valid and provide a wide range of dramatic effects; it is unfortunate that the uniformly high price tags of all harmonizing devices has prevented the average musician from being able to take advantage of these new effects. While the MXR unit doesn't exactly slash the price of pitch transposition, it does offer a certain price advantage compared to other units on the market. This should allow smaller studios, P.A. companies, and well-off musicians a chance at discovering some of the neat effects offered by pitch transposing devices.

As noted earlier, the MXR Pitch Transposer is very easy to use, and if you can afford the display, is equally easy to set up . . . maybe foolproof would be a more accurate word. The quality of the sound is quite good, there are a number of convenience features and the unit seems to have been designed for flexibility as well as "set-and-forget" types of applications. The inclusion of the Aux jacks is a strong point, since it opens up the PT to unusual applications and experimentation.

I feel that overall, MXR has produced a high quality device at a fair price; the emphasis is definitely on cost-effectiveness and ease of use. If you thought you couldn't afford a pitch transposing device, the Pitch Transposer just might change your mind—and open up a whole new texture of sonic effects in the process.

BASIC SPECIFICATIONS

Maximum input before clipping, transposed channel, regeneration off: Front panel jacks: 3 V p-p (high setting); 1 V p-p (low setting).
Rear panel jacks: 10 V p-p.

Phase, dry channel: non-inverting

Phase, effect channel: inverting

Front panel input jack impedance: approx. 350 K

Rear panel input jack impedance: approx. 50 K

Output impedance: under 800 ohms

Dry frequency response: essentially flat from 30 Hz-100 kHz

Effect frequency response: 1 octave up, down −6 dB at approx. 3.5 kHz*
Same octave, down −6 dB at approx. 7 kHz*
1 octave down, down −6 dB at approx. 8 kHz*

*Distortion occurs at a somewhat lower frequency. Input signals higher than these frequencies start to produce "aliasing" (a mixture of the input signal with signals generated internally by the Pitch Transposer).

Ambient Sound

By Len Feldman

The Sound That Refuses to Die

It has been variously described as a "warmer" sound, a "softer" sound, and even as a "glowing" sound. Color it red (instead of blue), but however you describe it or characterize it, the music performers who treasure and preserve their old, battered tube amplifiers, and the audiophiles who will pay just about anything for a vintage Marantz tube power amplifier have one thing in common. They both believe that tube sound is to be preferred over transistor sound. Notice that I didn't say that they believe that tube sound is more accurate than transistor sound—only that it is *preferred* by some listeners.

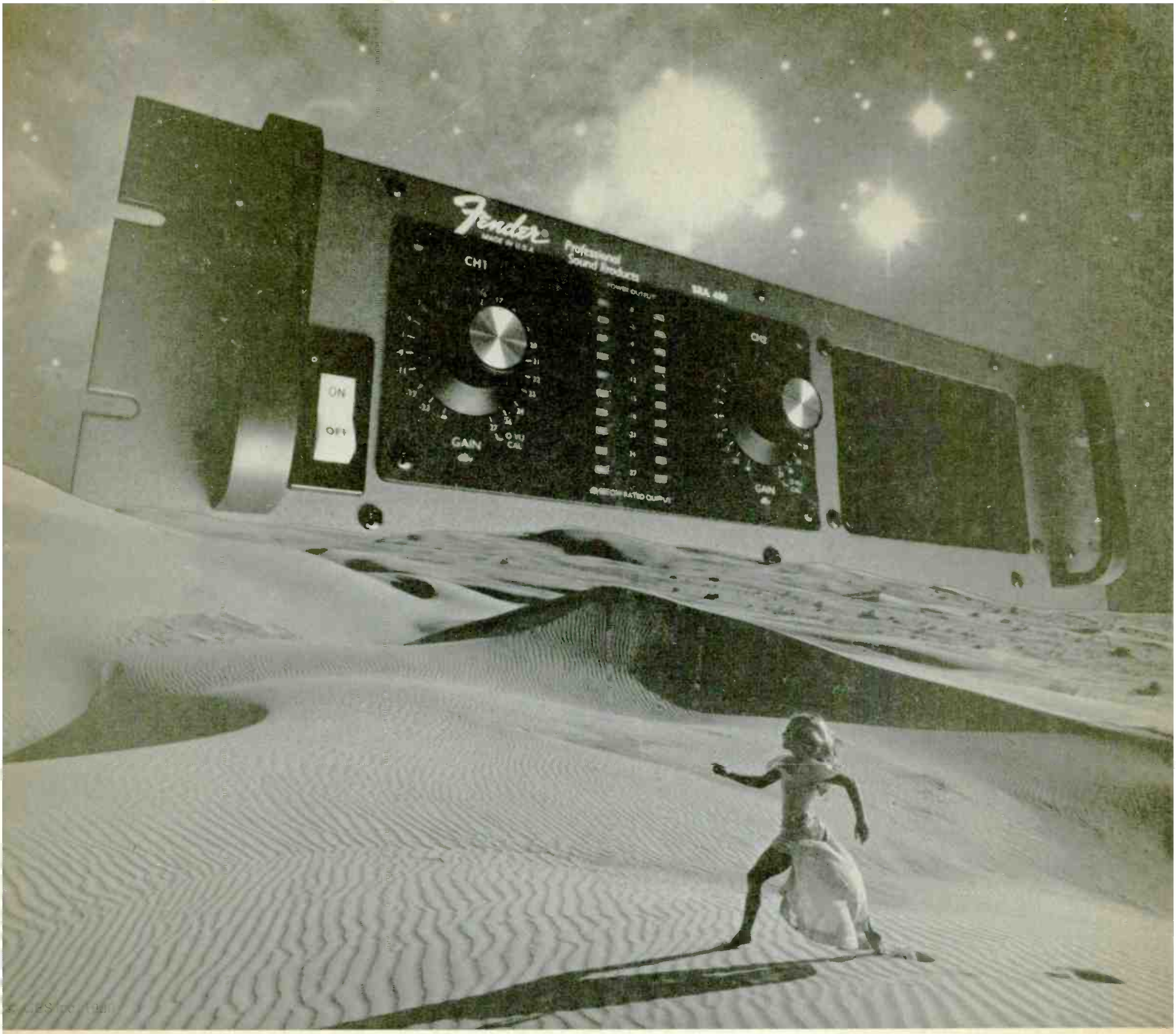
There was a time, not too long ago, when the more cynical devotees and critics of audio insisted that the traditional tube-sound lovers were talking themselves into something that really couldn't be heard. More recently, however, the theorists have provided us with a fair amount of scientific and measurable backup which supports the "tube sound" theory. What evolves from all of this, however, is the realization that it is not the tubes in and of themselves which provide inherently "better" or "warmer" sound, nor is it the transistors which, of themselves, produce a more brittle or, if you will, "cold" sound. Rather, it is the way in which these devices have traditionally been used.

Negative Feedback Not A Cure-All

Way back in the early 1930s, the concept of negative feedback was first used in an audio amplifier. The idea was fairly simple in concept. If you took a fraction of the output waveform of an amplifier and fed it back to the input, but out-of-phase in polarity, any distortion components in the output waveform would be re-amplified through the amplifier, but in an out-of-phase relationship with the distortion that the amplifier itself was creating. This would tend to *cancel* the overall distortion, or at least reduce it significantly. Of course, negative feedback also reduces the overall *gain* of the amplifier, and so you either have to add more internal amplification stages or settle for lower overall gain. The amount of feedback applied to an amplifier is measured in decibels, or dB, the number of dB of feedback corresponding to the amount by which the gain of the amplifier is lowered when feedback is applied. The reduction of harmonic distortion brought about by negative feedback is directly proportional to the amount of feedback applied.

Thus, if you have an amplifier that, without feedback (open-loop condition), delivers a distortion of 1.0%, applying 6 dB of negative feedback will reduce that distortion to 0.5%, while applying 20 dB of feedback will reduce the THD to 0.1%. (20 dB of feedback, by the way, was not an unusual amount to be found in the old tube amplifiers.) Tubes, it seems, can be made to operate fairly linearly over wide voltage swings, which meant that even without feedback applied, the distortion levels of tube amplifiers were not all that horrible. Furthermore, in those early days we were not engaged in the kinds of specsmanship contests that exist today. If a tube amplifier, less feedback, produced 3% total harmonic distortion and applying 20 dB of feedback brought that THD level down to 0.3%, that was considered to be just fine and probably more than adequate. Today, many would thumb their noses at a power amp that had a rated distortion level of 0.3%. (Whether or not you can *hear* 0.3% THD when listening to a music signal seems to be a secondary consideration.)

Even in the tube amplifier days, we all knew that excessive feedback could create problems. Too much negative feedback, for example, could lead to instability at high frequencies. The amplifier that had too much feedback could take off and become one fine oscillator at a super-audible frequency that quickly destroyed tweeters. Because of phase-shifts within the amplifier (which could also be thought of as time delays), what starts out as *negative* feedback at audio frequencies becomes *positive* feedback at out-of-band frequencies and that leads to oscillation. Highly reactive loads



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Fender's SRA 400 Stereo Power Amp is the rugged, new power source for your stage monitors, house PAs and side fill speakers. Fender engineers designed it to do any job where you need 400 ultra-clean watts—and to keep on delivering as long as your band can play.

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Total control and protection. The Fender SRA 400 lets you repeat volume settings with dual 22-step calibrated attenuators. Keep your output under control with logarithmic LED power level ladders. Maintain cool, reliable performance with forced air movement through a tunneled heatsink.

Fender's attention to detail is so accute that the on/off switch is also a current and voltage-limiting circuit breaker to protect against power line surges. This engineering concern with matters great and small, backed by months of on-the-job testing, pays off in long-term reliability.

We've put it in writing. The Fender SRA 400 written specs are all the proof you need:

Slew rate: 40 volts/micro second.
TIM distortion: Extremely low; typically less than 0.05%.
IM distortion: Typically less than 0.03%.
THD: Less than 0.09%.
Full complementary symmetry design.

At last, the dependability and versatility you've always wanted in a power amp are available in one 5¼" package. Turn on an SRA 400 at your authorized Fender dealer.

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(instead of the idealized resistive loads with which labs often measure amp performance) could accelerate the process, so that an amplifier that seemed stable on the lab bench often took off in the real world and was unstable. So, generally speaking, designers of tube amplifiers kept the feedback down, and depended upon the good linearity of tubes to take care of things.

Another point: When you drove a tube into overload, instead of clipping sharply and turning the waveshape into a square wave when severe clipping occurred, tube circuitry tended to overload in a slower or more gradual manner. There were no sharp corners to the waveform, just a rounding of the peaks and a gradual departure from true sinusoidal shape. Both the tube's characteristics and the moderate use of feedback contributed to this "soft clipping" effect.

The Transistor Never Meant to Be an Amplifier

Transistors make great switches. The easiest thing you can make a transistor do is turn on or turn off. But in between, when it is partially conducting (semi-conductors, remember?), it isn't anywhere as linear in its basic characteristics as were the good old tubes. At least that was so when the first solid-state amplifiers were designed, and, to some extent, it remains so today. And, as we said, non-linearity results in harmonic distortion. Well, the most apparent cure for harmonic distortion in an amplifier was, you guessed it, lots of negative feedback. If 20 dB of feedback was typical for tube amps, 60 to 80 dB became the norm for transistorized or solid-state power amps. To be sure, the harmonic distortion numbers came down—dramatically. We started seeing THD specs of 0.1%, then 0.01% and, finally, even 0.005%. But, as static forms of distortion were lowered, dynamic forms of distortion (the kind you can actually detect when trying to reproduce *music* signals) seemed to get worse and worse. The best known of these, of course, was TIM (Transient Intermodulation Distortion), but there were other effects, too, not the least of which was the "hard" sound we talked about earlier and the "hard clipping" which is closely related to it.

Overuse of feedback permitted designs which, without feedback applied, were narrow in bandwidth (poor frequency response) and high in harmonic distortion. Slap enough feedback from output to input and the bandwidth extends way out beyond the audible range; the THD goes down to near unmeasureable levels; and all *seems* well, at least on the test bench using static fixed tone signals. But when it came to sound quality, the experts would pick the old tube sets every time.

Gradually, designers began to take another look at all this, which brings us to some of the present day thinking concerning amplifier design. Many manufacturers have been concentrating on their open-loop (before feedback) amplifier designs, trying to provide these designs with flat response to beyond the audio range and a high level of linearity or low harmonic distortion. With advanced circuit techniques as well as low-level and output transistors that can be made to exhibit better linearity, such designs become practical. Once these goals are achieved, it then becomes possible to finish the design by applying only moderate amounts of overall negative feedback—perhaps as little as was used in the days of tube amplifiers. These measures, together with circuit techniques that have recently been developed, permit the amplifiers to exhibit overload characteristics that are virtually indistinguishable from those of tube amplifiers. And, as you might expect, the "tube sound" which everyone always talks about suddenly is heard from loudspeaker systems being driven by solid-state transistorized amplifiers.

To be sure, there are negative aspects (forgive the pun) to decreased overall negative feedback in an amplifier. One of these has to do with damping factor, or the ability of the amplifier to approach zero internal "looking back" impedance, as seen by the connected loudspeaker. A high damping factor helps to suppress unwanted low-frequency cone motion that is often triggered by record warp or turntable rumble. Large infrasonic cone excursions, while of themselves inaudible, cause intermodulation distortion products to be generated *within* the audible frequency range. This is especially a problem with today's so-called DC amplifiers that have uniform response literally down to "zero Hz."

Several manufacturers have come up with solutions to these problems. One approach is to apply two separate feedback loops to an amplifier. The first of these is the lower-feedback amount discussed earlier, which is applied throughout the audio frequency range. A second feedback loop, active only at sub-sonic frequencies, applied increasingly larger amounts of negative feedback down to zero Hz (from about 5 Hz downward). The lower quantity of feedback provides us with the "warm" tube sound we all crave and with much reduced TIM, while the secondary sub-sonic feedback loop not only "tightens" the speaker cone at those rumble and warp frequencies but, since increased feedback at these lower sub-audible frequencies means lower gain, it also acts as a sub-sonic filter as well, providing a steadily increasing roll-off of infrasonic frequencies down to zero Hz.

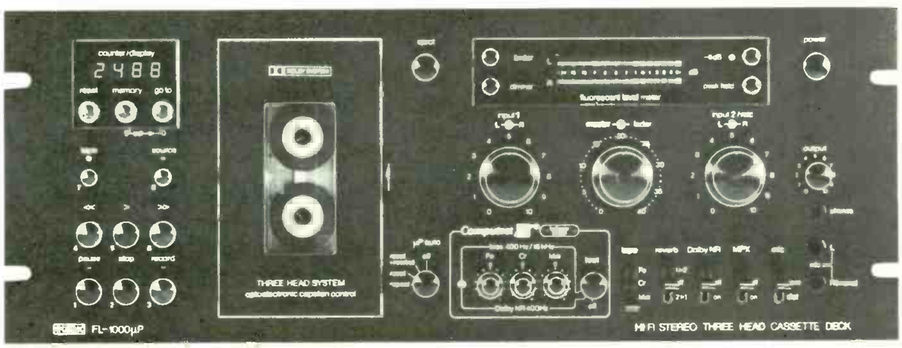
If you haven't done some serious A-B testing lately between older, tube-type amplifiers and the new crop of solid-state amplifiers that incorporate the kinds of design innovations we have been discussing, you may be very surprised to find that in a double-blindfold test you probably will no longer be able to pick out which amplifier really uses tubes and which uses transistors.

MODERN RECORDING & MUSIC

LAB REPORT

NORMAN EISENBERG AND LEN FELDMAN

Eumig FL-1000 Cassette Recorder



General Description: The Eumig FL-1000 is a three-head cassette deck of exceptional versatility. All transport functions as well as such electronic functions as alignment and optimizing for different tapes are controlled by a built-in microprocessor. In addition, the FL-1000 may be directly interfaced with any 8-bit computer for expanded versatility and applications (see LF's comments below). The FL-1000 also is metal-tape capable, it provides on-the-panel input mixing, as well as a built-in facility for introducing reverb into a cassette being recorded.

The transport uses an optical sensor and servo system that eliminates solenoids as well as belts, flywheels and pulleys. Tape motion is imparted via a step-motor and gear assembly. Logic-control enables fast-buttoning between the play, fast-forward and rewind modes, although run-in recording is not possible. However, the fast-forward and rewind speeds are variable, and may be slowed down to help locate (by reference to the counter) a specific recorded portion of the tape.

The microprocessor permits storing in its memory any point on the tape; the digital readout display is used to pinpoint the spot for storing as well as to identify it. In addition, the transport buttons are numbered so that the pause button is 1, the stop button is 2, the record button is 3, the rewind button is 4, the play button is 5 and the fast-forward button is 6. Two additional controls—the tape and source monitor buttons—are numbered 7 and 8. A memory button on the tape counter display is numbered 9, and a "Go to" button is numbered 0. These numbers are keyed to the microprocessor in the deck and may be used to activate its "Go to" function—that is, by pressing the appropriate combination of four numbers, you have instructed the transport to advance or return to that numbered spot on the tape. To get the transport to do this, you then press the "Go to" button. When that spot on the tape is reached, the transport stops—unless you also decide to instruct the deck to go into the play mode at that spot, or into pause-and-record which it also can be programmed to do. The "Go to" option also may be used to get the deck to record from any given point on the tape to any other.

It also is possible to engage an automatic counter reset and rewind, by means of which the tape is rewound to the beginning when a cassette is loaded into the machine. You also can get the counter to automatically reset itself to zero by merely opening the door to the cassette compartment. A play-repeat option also is provided, and unattended operation in record or play is possible with the use of an external timer.

In addition to a three-position tape selector switch (ferric oxide or normal bias; Cr or high-bias tapes; MET for metal tapes), there's an adjacent special "Computest" panel that permits both Dolby level adjustments and fine-bias adjustments for each tape. A pair of LEDs, associated with these controls, indicates correct control settings.

The front panel is "busy looking" but logically planned. At the left is the tape-counter panel with its four-digit indicator, plus the buttons for reset, memory and "go to." Directly below it are the two buttons for tape or source monitor, and below them are six transport buttons for rewind, play, fast-forward, pause, stop and record. As mentioned above, these buttons also are numbered for use in the access-and-program option via the microprocessor.

The cassette well is positioned vertically, and the door over it swings open from the right when the eject button is pressed. Near the right-hand bottom of the cassette well, just to its right, is the microprocessor-automatic control for rewind and counter-reset, repeat and timer operations.

The signal metering panel near the top of the panel uses the fluorescent bar-graph type of display, and is calibrated from −20 to +6. A button permits increasing the numbers shown on the metering scale by 6 dB; according to the manufacturer this enables recording at higher levels (as for instance with metal tape) than would otherwise be shown on the meters (e.g., with this button engaged, an actual recording level of +8 dB would appear on the the meter as +2 dB). Other buttons on the metering panel permit adjusting the intensity of the illumination, introducing a peak-hold indication and switching in a limiter. Just to the right of the meter panel is the AC power off/on switch.

There are three sets of signal controls. One dual-concentric pair handle input 1 signals (left and right channels independently or simultaneously, as desired) which are the normal line signals. There also is a similar control arrangement for input 2 signals. Microphones or high-level sources may be used on "input 2." Between the two pairs of controls is a third pair designated as master and fader. The master control is used for setting overall level after the individual inputs are adjusted. The fader may be used for mixing and to fade-in or fade-out either input.

Below these controls are the "Computest" controls. The general tape selector—a three-position toggle switch—is to the right, followed by the switch for adding reverb (from input 1 to input 2 or vice versa), the Dolby NR switch, a MPX filter switch and a near/distant microphone switch (attenuator). The input jacks for microphones are at the extreme right. Above them is a stereo headphone output jack with its own level control. This level control also adjusts optional variable output jacks at the rear, although there also are fixed-level output jacks. Inputs at the rear include the line 1 and line 2 pairs, plus a European-DIN socket. Also at the rear are a grounding terminal, a power-line voltage adjustor, a remote-control socket that is used for the interface with a computer and the unit's cord.

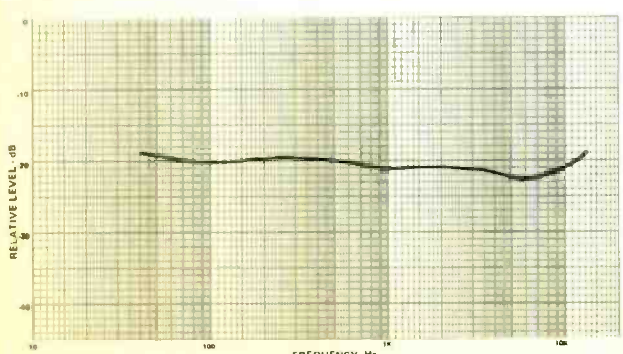


Fig. 1: Eumig FL-1000: Playback-only response, 120 usec EQ, using TDK AC-337 test tape.

Test Results: The Eumig FL-1000 met its frequency response specifications with room to spare for all three classes of tape with which it was tested (normal bias, high bias and metal bias). With the latter two tapes it actually exceeded the "20 Hz to 20 kHz" bandwidth at normal recording levels. Oddly enough, Eumig does not have a spec for distortion, and the test results for THD—while not too high—do not seem as low as one might have expected in a deck of this general class.

Playback-only response, using a 120-microsecond test tape (ferric oxide) is shown in the graph of *Fig.1*. Since the FL-1000 is a three-head machine, we were able to use our spectrum analyzer to obtain real-time sweep plots of record/play response for the three major classes of tape (we used TDK type OD for the normal-bias ferric-oxide sample; TDK SA, a ferric-cobalt type for the high-bias test; 3M Metafine for the metal tape test). Results for 0 dB and for −20 dB record levels are shown in the 'scope photos of *Figs. 2, 3 and 4.*

While the metal-particle sample results in our "Vital Statistics" chart may not seem all that impressive at first glance, if you examine the 0-dB curve of *Fig.4* and compare it with the 0-dB curve of either *Fig. 2* or *Fig. 3,* you can see that high-frequency headroom (before saturation) is, indeed, much better for the metal tape. For some reason—which probably has to do with the way the machine was set up at the factory—midrange headroom using the metal tape sample was disappointing, as was the overall signal-to-noise ratio of the metal tape. In the case of our sample machine, TDK SA actually yielded a better "single number" S/N ratio than did the Metafine. We do not make too much of this, however, since—initially at least—there were a great many variations in metal tape characteristics, and Eumig may well have adjusted our sample for a different brand of metal tape.

Wow and flutter were extremely low, and fast-wind time was exceptionally fast. The deck was put through its various operational modes and microprocessor-controlled options, and performed all of them very much "as claimed."

General Info: Dimensions are 19 inches wide (unit may be rack-mounted); 7 inches high; 13 inches deep. Weight is 26.5 pounds. Price is $1550.

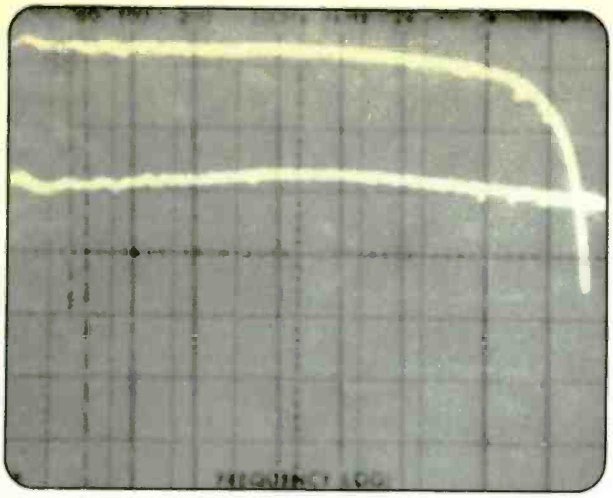


Fig. 2: Eumig FL-1000: Record/play response at 0 dB and −20 dB with TDK OD tape.

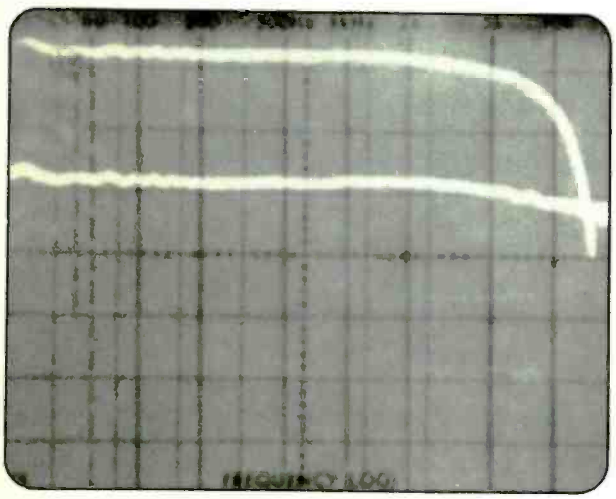


Fig. 3: Eumig FL-1000: Record/play response at 0 dB and −20 dB with TDK SA tape.

Individual Comment by L.F.: In terms of its features and performance, the Eumig FL-1000 is in every sense a fine cassette deck. But, as I learned from some research and a lot of talking to some of the people involved with it, it turns out to be more than that.

Each of its many functions is controlled by a microprocessor. And because all of the functions—such as tape movement, operating mode and even alignment/optimization (which Eumig calls "Computest")—are micro-processor controlled, the FL-1000 can be directly connected to any 8-bit computer such as a Commodore Pet, an Apple II, TRS-80, etc. Computer interface is via a 10-line cable connected to the 8-bit parallel bus or user port of the given computer. The other end of this cable fits into a 10-pin socket on the deck. A single computer can control up to sixteen separate FL-1000 decks, and control here means all operating modes as well as direct access location to any point on a given tape cassette. A computer command such as "go to" will send the machine to a designated location on the tape at which point the machine can be commanded to pause, play or record.

Given this sort of computer interface and the Eumig software program, a form can be displayed on the computer CRT which allows up to fifteen selections to be indexed on each side of each cassette. The index contains a starting location, ending location, and can also include the artist's name, song title, record company, album number, etc. Once the index has been typed on the computer keyboard, a command is given to digitally "write" it on the audio cassette, which gives you a permanent index of what is on each tape.

Once a tape has been indexed in this way, playback functions become extremely versatile. For example, if you wanted to play selection number 3 at location 1650, you might hit P and the number 3. The tape will automatically read the starting number 1650 from the index and go to that location and begin to play—all in a matter of a few seconds (note the speed of fast forward and rewind in this machine, much faster than most). A second way to reach the desired location would be to simply "say" P 1650. The tape goes to that location and plays whatever it finds there. A third method known as "search" allows you to type in any sequence of letters and numbers. The computer will search the index for this sequence and when it finds it the sequence will appear in reverse on the screen. To play that selection, you need then only hit "P" and "Return" and the computer will send the transport to that location and begin play.

If you own many cassettes it would be possible to store all the indexes from all of your cassettes on one floppy disc. If you then wanted to search your collection for a given artist or group, the computer would find all musical selections by that artist or group contained in your entire collection. These would be printed out on the screen along with cassette number and location within the cassette.

I was told that the FL-1000 can even be used as a search device for a home computer system. You could digitally read/write programs, data, files and so forth directly to the FL-1000. Its band rate is many times higher than the detachable cassette players that come with many home computers.

Whether you consider the purchase of a Eumig FL-1000 strictly for use as a high quality cassette deck or have ideas about how you might want to interface it with a computer in the future, the microprocessor control built into this deck is awesome. At an earlier demonstration at a recent AES show, I saw a dozen or so of these machines set up with a computer and arranged to provide total automation of an AM radio station! My, how the lowly cassette deck of the 1960s and early 1970s has progressed!

Individual Comment by N.E.: There is, obviously, much to admire in the FL-1000 and also some things to wonder about. As a feat of engineering, combining

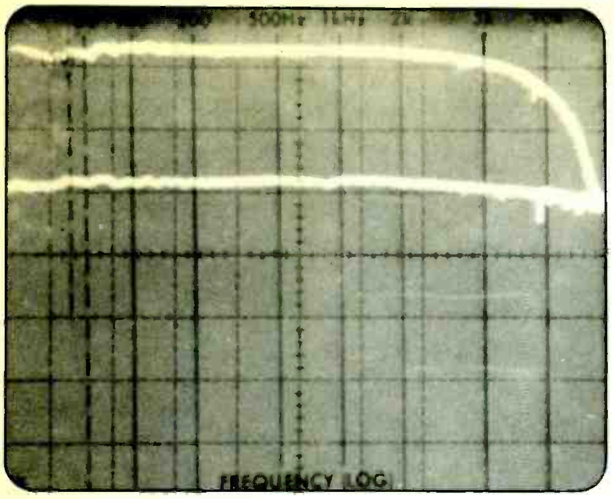


Fig. 4: Eumig FL-1000: Record/play response at 0 dB and – 20 dB with 3M "Metafine" tape.

state-of-the-art microprocessing with the cassette format, the FL-1000 is something of a tour-de-force. Its facility for interfacing with a "home" computer and the possibilities inherent in such a mating for a fairly sophisticated information storage and retrieval system lend an application-dimension to the FL-1000 not yet seen in the cassette format.

It is probably this aspect of the FL-1000 that makes it noteworthy, more so than its actual audio performance as a cassette recorder. Not that its performance is anything to "sneeze at." However, on final count, there are cassette decks that have lower distortion or better S/N figures, or complete fast-buttoning (in the FL-1000 you cannot go directly into record unless you hit the stop button first). On the other hand, the FL-1000 does have three heads; it offers a built-in reverb option that can enhance your recording efforts; it has an easy-to-use set of bias and Dolby level adjustments, a very useful metering system, a good on-the-panel input mixing facility, extra inputs and outputs, a welcome level control for headphones and a really smooth and very agile transport.

All that notwithstanding, the cost of the FL-1000 does reflect its "computerism" design and embellishments, and that is what the prospective buyer should consider, since—if that computeristic design does not interest you—there are cassette decks that *sound* at least as good but do cost less.

In common with a few other high-end cassette decks that have come along lately, the Eumig raises another question: Is the cassette format as a format reaching close to the overload point in terms of embellishments and features that have more to do with automated muscularity than with actual audio performance? One answer, obviously, would be that a unit such as this simply is "not for everyone" and that it is directed only at that portion of the market that really wants that automation and versatility and is willing to pay for it. But another answer could be less salubrious, and that would be that the cassette deck manufacturers have embarked on some kind of one-upmanship race in which ingenuity and the urge to come out with something different are seen as primary marketing necessities—a philosophy that is, to say the least, highly debatable in long-range terms.

One thing for sure: Demonstrated in all its glory by a nimble-fingered operator, the Eumig FL-1000 is bound to be a show-stopper.

EUMIG FL-1000 CASSETTE RECORDER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Frequency response, standard tape	± 3 dB, 30 Hz to 18 kHz	± 3 dB, 17 Hz to 18.5 kHz
high-bias tape	± 3 dB, 30 Hz to 20 kHz	± 3 dB, 17 Hz to 21 kHz
metal tape	± 3 dB, 20 Hz to 20 kHz	± 3 dB, 17 Hz to 20 kHz
THD at 0 dB		
standard; high-bias; metal	NA; NA; NA	1.9%; 1.2%; 1.3%
Record level for 3% THD		
standard; high-bias; metal	NA; NA; NA	+ 5; + 6; + 1.5 dB
S/N ratio, Dolby off		
standard; high-bias; metal	58; 59; 62 dB	58.5; 61.5; 59 dB
S/N ratio, Dolby on		
standard; high-bias; metal	66; 67; 70 dB	67; 69; 67 dB
Wow-and-flutter (WRMS)	0.035%	0.035%
Speed accuracy	NA	"0%" (perfect)
Microphone input sensitivity	0.2 mV	0.21 mV
Line input sensitivity	100 mV	95 mV
Line output level	775 mV	800 mV
Headphone output level	NA	390 mV/8 ohms
Fast-wind time, C-60	35 seconds	37 seconds
Bias frequency	100 kHz	100 kHz
Power consumption	50 watts	43 watts

AB Systems
Model 1200A Amplifier

General Description: The model 1200A by AB Systems Design, Inc. is a two-channel (stereo) power amplifier conservatively rated for an output of 300 watts per channel into 8-ohm loads, or 500 watts per channel into 4-ohm loads. It also may be bridged for a rated output in monophonic service of 1000 watts into an 8-ohm load. Totally modular in design, the 1200A employs independent and interchangeable output sections, each with its own fan. The complete amplifier consists of a main frame (model 1200) plus two each of the model 1201A amplifying modules and a bridge-mode switch.

The front panel reflects the twin-module design of this amplifier, with each channel independently mounted in the main-frame, and independently controlled. For channel 1, there are the power off/on switch, a level control, a power-on indicator and a clipping level indicator. For channel 2, the same items are repeated. The main-frame has air vents on its front and sides, handles, and provison for standard rack-mounting.

Inside the left-hand area of the main-frame (as viewed from the front) is a switch to select 4-8 ohm operation or 2-4 ohm operation. The former position applies to this model; the latter position is used with an optional alternate module. Inside the right-hand area of the main-frame is the bridge-mode switch to select stereo or mono operation.

At the rear, each channel has two kinds of inputs. One is for quarter-inch phone plugs; the other for XLR connectors. Balanced or unbalanced options are up to the user. Speaker outputs are standard binding posts. Also at the rear are fuse-holders, a low-high speed switch for the fans, the AC power cord and additional vents.

Test Results: The AB Systems model 1200A, in our tests, easily met its power and distortion specifications with room to spare. All power measurements exceeded the manufacturer's claims. Distortion was consistently lower than stated. Frequency response went well beyond the rated 20 Hz to 20 kHz band.

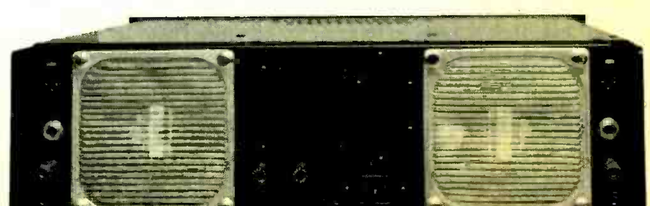
In addition to this high level of performance, we also were impressed with the overall professionalism of this amplifier in terms of its design, construction and—as far as we were able to judge—its reliability. A typical plug-in module is shown in *Fig. 1.* This is the "standard" module (4 to 8 ohms) supplied with our test samples. The switch inside the main-frame permits the user to go to 2 or 4-ohm operation with an alternate module (not tested) which is rated to deliver 400 watts per channel into 4 ohms, 600 watts into 2 ohms or—in bridged mono mode—1200 watts into 4 ohms.

The built-in fans are inaudible unless one listens very closely to the amplifier for their sound.

General Info.: Dimensions are 19 inches wide; 5½ inches high; 17½ inches deep. Weight is 72 pounds. Price: $1200.

Individual Comment by L.F.: The AB Systems 1200A is very much a "professional" amplifier. It is built to assure long-term "workhorse reliability," but there is nothing of the brute-force approach in its design. On the contrary, it is a highly sophisticated piece of audio architecture that is, according to its designers, the first "totally modular" power amplifier.

While I have no doubt as to its suitability in professional applications, there is the question of whether the 1200A would be equally suited for a home stereo system (assuming that system needed the kind of power this "King of the Jungle" amplifier can deliver). These days,



AB Systems 1200A: Rear panel of unit.

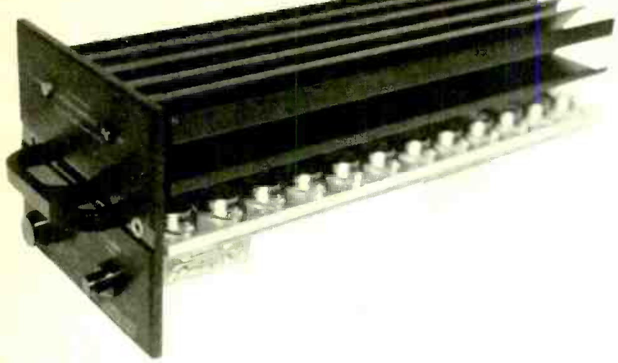


Fig. 1: AB Systems 1200A: One module (amplifier channel), shown separately, removed from 1200A mainframe.

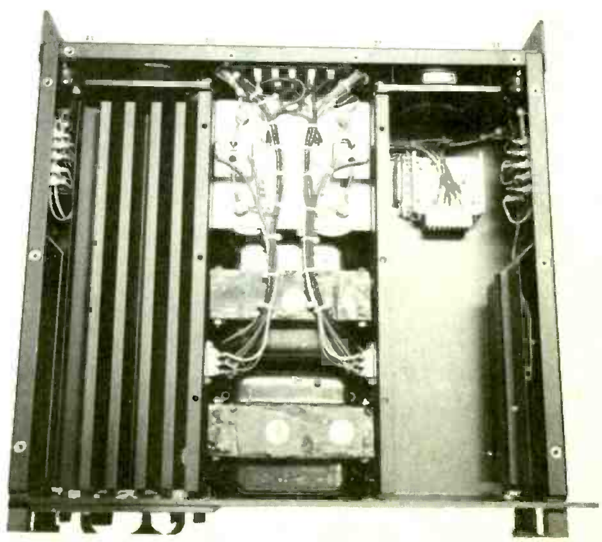
it seems as if no right-thinking audiophile would dream of buying a high-powered amp that has a rated harmonic distortion of 0.2 percent—not with the major hi-fi manufacturers touting an increasing number of zeros after the decimal point in their distortion specs. The fact that the 1200A measured only 0.1 percent distortion at its rated output would not alter the "true audiophile" view—even if it were proven (as it easily can be) that no one can *hear* harmonic distortion levels of either 0.2 or 0.1 percent. Happily, professional users of audio amplifiers are not easily influenced by such numbers races—which is all to the good, since the AB Systems amplifier is a great sounding unit that utilizes a minimum amount of loop feedback in a quasi-complementary Class AB-2 design that sounds great to these ears.

For short-circuit protection, the amp uses "VI type" energy limiters. Thanks to the large safe operating area of the output stage (12 RCA multiple-emitter power transistors with a total power dissipation capability of 1800 watts are used), the limiters do not operate until well below a fully reactive load or resistive load of 2.8 ohms at full power, using the 1201A module. Solid-state circuitry protects the speaker system and monitors DC at the output. In the event of a DC latch of either polarity in excess of one second, a control circuit defeats the entire system by turning off the primary AC circuitry.

Unless you bend right down to the amplifier itself you will not hear the fans running. Their speed can be changed to a higher one should it become necessary to install the amplifier in a confined location with inadequate air circulation. Again, I doubt that a dyed-in-the-wool home audiophile would consider owning an amplifier that had any mechanical noise even if that noise were below the level of the room's ambient noise.

I think you can see what I'm getting at. Here is a really superb and reliable amplifier that is not likely to find its way into stereo hi-fi shops. For professional users, that's probably good news. It means that AB Systems will be able to turn out enough of these fine units to meet professional demands without having to speed up production or to compromise quality. Who was it said: "Some things just can't be hurried"? Conceptually, there is no reason why AB Systems could not offer an equivalent model for the home user, but with much of the cost of this high-powered unit dedicated to long-term reliability and little to "specmanship," the home user might just not feel as strongly or as positively about this amplifier as I do.

Individual Comment by N.E.: There is little I can add to the above since I pretty much agree with it all. Readers of these reports may recall the name of Ross Tane who runs the sound shop called "The Sounds of Music" in Lenox, Mass. where I rent studio space. When a unit as "heavy" (metaphorically as well as physically) as the AB Systems 1200A amplifier comes along, Ross likes to get into the act not only to help lift the beast but also to listen and maybe offer some comment. Ross's comments, while those of a retailer interested in making sales, are perhaps more than typically informed since the man is after all a graduate engineer—one of the relatively few in commercial audio who knows what happens behind the panel and how to fix it when things don't happen that should, or that do happen that should not. The inevitable question came up of a retailer's attitude toward an amplifier that had a distortion spec of 0.2 percent as opposed to complete receivers that cost less and boasted distortion specs of a hundred or more times less distortion. I let Ross speak for himself: "Those astronomically small distortion numbers are academic nonsense. They appeal to people whose real understanding of audio and whose musical taste are in inverse proportion to their desire for status. I would be more enthusiastic selling a unit like this with its reliability and ruggedness than one in which those qualities were doubtful no matter how seemingly impressive the latter's specs were."

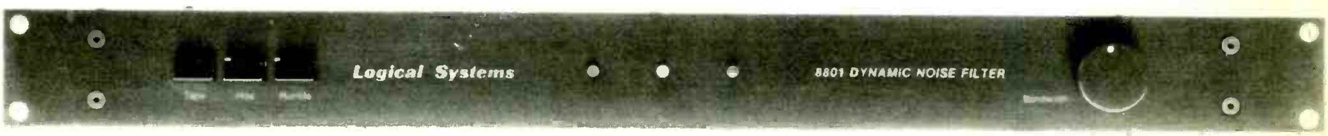


AB Systems 1200A: Empty module compartment at right shows how amp channels plug in without requiring any cable disconnection from associated equipment.

AB SYSTEMS 1200A POWER AMPLIFIER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Continuous power for rated THD,		
8 ohms, 1 kHz	300 watts	331 watts
4 ohms, 1 kHz	500 watts	522 watts
FTC rated power (20 Hz to 20 kHz)	300 watts	327 watts
THD at rated output, 1 kHz, 8 ohms	0.2%	0.1%
1 kHz, 4 ohms	0.2%	0.1%
20 Hz, 8 ohms	0.2%	0.06%
20 kHz, 8 ohms	0.2%	0.1%
IM distortion, rated output, SMPTE	0.2%	0.6%
CCIF	NA	0.05%
IHF	NA	0.1%
Frequency response at 1 watt	20 Hz to 20 kHz ± 0.25 dB	13 Hz to 44 kHz ± 1 dB
S/N ratio re:1 W, "A" wtd, IHF	NA	81 dB
re:rated output "A" wtd	103 dB	105 dB
Dynamic headroom, IHF	NA	1/4 dB
Damping factor at 50 Hz	NA	125
Input sensitivity, IHF	NA	0.072 volt
re:rated output	0.75 volt	1.2 volts
Power consumption, idling; maximum	NA; NA	140 watts; 1400 watts

Logical Systems Model 8801 Dynamic Noise Filter



General Description: The model 8801 from Logical Systems is a two-channel (stereo) device for removing noises from source material during recording or playback. It operates essentially on the variable-bandpass filter principle—which is to say, the device's bandwidth changes continually in accordance with how its circuitry analyzes the input signal in terms of amplitude, frequency and duration ("persistence" as the owner's manual puts it). Both high ("hiss") and low ("rumble") filterings are offered; either may be selected or bypassed via front-panel switches. A bandwidth control permits adjusting the threshold of the high filter. Associated with this front-panel knob is the unit's power off/on switch. A rear-panel slotted-shaft adjustment permits setting the rumble filter for low filtering action. Signal jacks at the rear are pin-type phono connectors, with provision for patching into the tape-monitor loop of a stereo system. To replace the tape-monitor loop thus preempted in this hookup, the rear contains additional tape in and tape out jacks, while the front panel contains another tape-monitor switch. Three LEDs on the panel indicate the "break frequencies" for the high filter (red for 3 kHz; yellow for between 3 and 10 kHz; green for above 10 kHz).

The model 8801 may be fitted into 1¾ inches of standard 19 inch rack-mount.

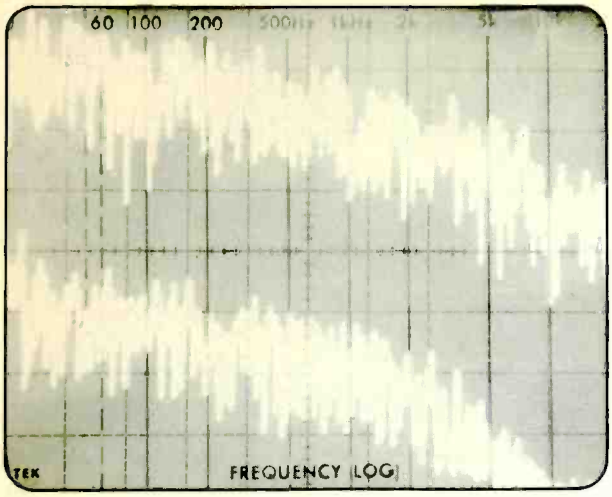


Fig. 1: Logical Systems 8801: Action of noise filter when fed with pink noise at 30 dB level difference.

Test Results: In *MR&M's* lab tests, the model 8801 met some but not all of its published specifications. For instance, signal-to-noise ratio was measured as a little better than claimed; reduction of rumble was about 3 dB shy of the claimed 20 dB. Distortion varied at mid-frequencies, and was generally higher than claimed at the low and high extremes of frequency. These discrepancies aside, the model 8801 does perform its intended functions reasonably well.

Fig. 1 attempts to illustrate the action of the high-cut dynamic filter, using a pink-noise signal source and a spectrum analyzer sweep and display from 20 Hz to 20 kHz. The upper noise spectrum slopes downward at the usual 3 dB-per-octave rate, which is characteristic of pink noise. In other words, at this loud level, the filter is wide open and passes all frequencies. It "thinks" that such "loud" highs must be program material (as, indeed, they normally would be).

But when the overall input level to the device is reduced by 30 dB (lower trace), it may be seen that—up to about 2 kHz—the display runs parallel to the previous upper display. However, beyond that frequency, the filter action causes a faster roll-off of the noise than was the case for the former sweep. No controls on the model 8801 were changed between the time of the first and second sweeps.

Fig. 2 shows the sweep changed from logarithmic to linear (to better show the action of the rumble filter). Zero Hz is at the left, and the scale is 50 Hz per linear division across the face of the screen (highest frequency here, then, is 500 Hz). As specified, the turnover point for the rumble filter always is at or near 150 Hz, but the slope of the low-frequency filtering is what varies as you adjust the rear-panel "bass calibrate" control.

To study what happens with various settings of the front-panel threshold control, we reverted to single-tone sweep of frequencies from 20 Hz to 20 kHz, and to a logarithmic frequency display on the spectrum analyzer (*Fig. 3.*) This time, 20 Hz is at the left; 20 kHz at the right. At the high end of the sweep you may notice what happens to the highs as the threshold control is varied from one extreme, through mid-position, to the other extreme. All three curves were taken at the same loudness level of the sweeping tone, which makes it clear that it is quite important to follow the calibration procedure when setting the threshold control. Fortunately, the three colored LEDs on the front panel make the setup fairly easy to perform, but the procedure should be repeated for different source material if optimum results are expected.

General Info: Dimensions are 19 inches wide; 1¾ inches high; 7 inches deep. Price: $289.

Individual Comment by L.F.: As Logical Systems puts it in its somewhat simplistic treatise on noise (which is packed in along with the owner's manual and the noise filter itself): "A dynamic noise filter is a device that decides what is music and what is noise, and tailors the bandwidth instantaneously to let only the music through. It constantly analyzes and tracks program material and gradually rolls off those frequencies that are not needed for the music. Dynamic filtering can be set up for any frequency, and so can be used to eliminate high and low frequency noise, not just tape hiss."

All well and true, but a subsequent statement in the booklet can easily be misinterpreted. It says: "Dynamic filters remove noise from existing program material." That, I think, is a bit of an overstatement.

The fact of the matter is that dynamic noise filters (and this is not the first one we have checked out) have pretty critical threshold settings. The ideal situation would be for the dynamic filter to "close down" and become a low-pass filter when there is no high-frequency energy in the music program at any level. Indeed, if

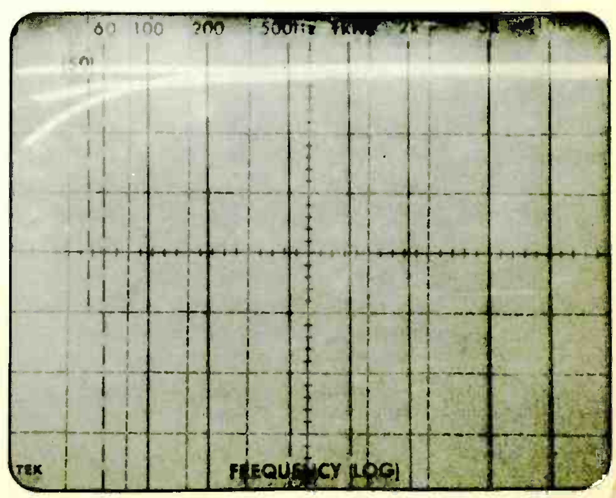


Fig. 2: Logical Systems 8801: Range of control of rumble reducing circuitry.

things are set up correctly, the 8801 works pretty well in this manner. But what about high frequency musical material that is at a very *low* level; as low a level as the residual hiss or noise you are trying to eliminate? Let's not kid anybody! Under those circumstances, even the very best designed dynamic noise filter is going to wipe out both the noise and the low-level high frequency content of the music. There is *no* way that it can actually tell the difference.

Note, too, that Logical Systems tells us that the attack and release time of the filter is "program dependent," and indeed it is. With some kinds of program material it works splendidly, while with other types, one can distinctly hear the unit breathing and pumping.

A dynamic noise filter, of course, should not be mistaken for any kind of two-step noise-reduction system (such as Dolby or dbx). Actually, though, that is its virtue: It can *help* to remove noise from program sources that already contain such noise (Dolby and dbx, on the other hand, cannot).

Individual Comment by N.E.: I think the model 8801 is likely to prove most effective when the noise content of a given program source is fairly steady and uniform, such as FM background hiss, or hissy cassettes that were recorded without benefit of Dolby and must be played back the same way, or a really dirty grumbly turntable. It is far less effective on noises of a random or sporadic nature such as pops and ticks on disc recordings.

In any event, I do not agree with the statement made in the owner's booklet: "A dynamic noise filter is a device that decides what is music and what is noise . . . ." Only your ears can decide that. The fact is, you cannot filter noise out of a signal, especially high-frequency noise below a certain amplitude level, without also knocking out some of the high frequency signal content. It's a compromise, at best, and the real solution is to improve the program source itself so that no "audio aspirins" are needed.

In light of the discussions included in the manual of other noise-reduction systems, including those by Dolby and by dbx, it may be necessary to repeat here that neither Dolby nor dbx is designed to "clean up" dirty source material. Rather, they go after the noise inherent in a signal handling format such as disc cutting, tape recording, FM broadcasting and so on—and in so doing they actually expand the clean usable dynamic range. A noise filter is essentially a lopping-off action and while admittedly the model 8801 does a good job of lopping, it is not—as implied—the be-all and end-all for all musical applications.

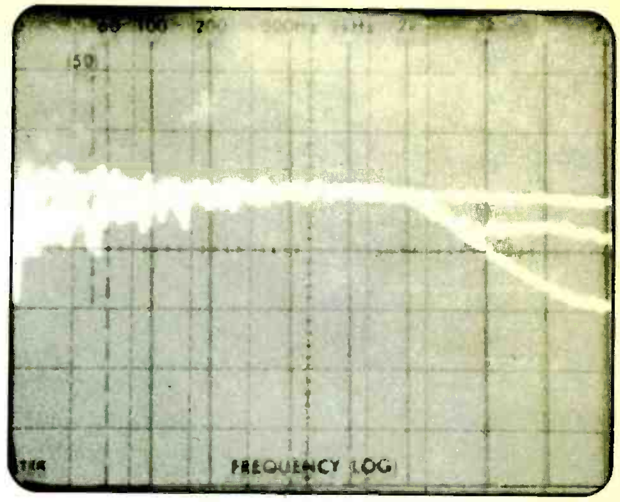
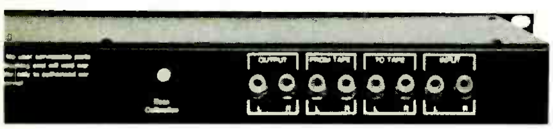


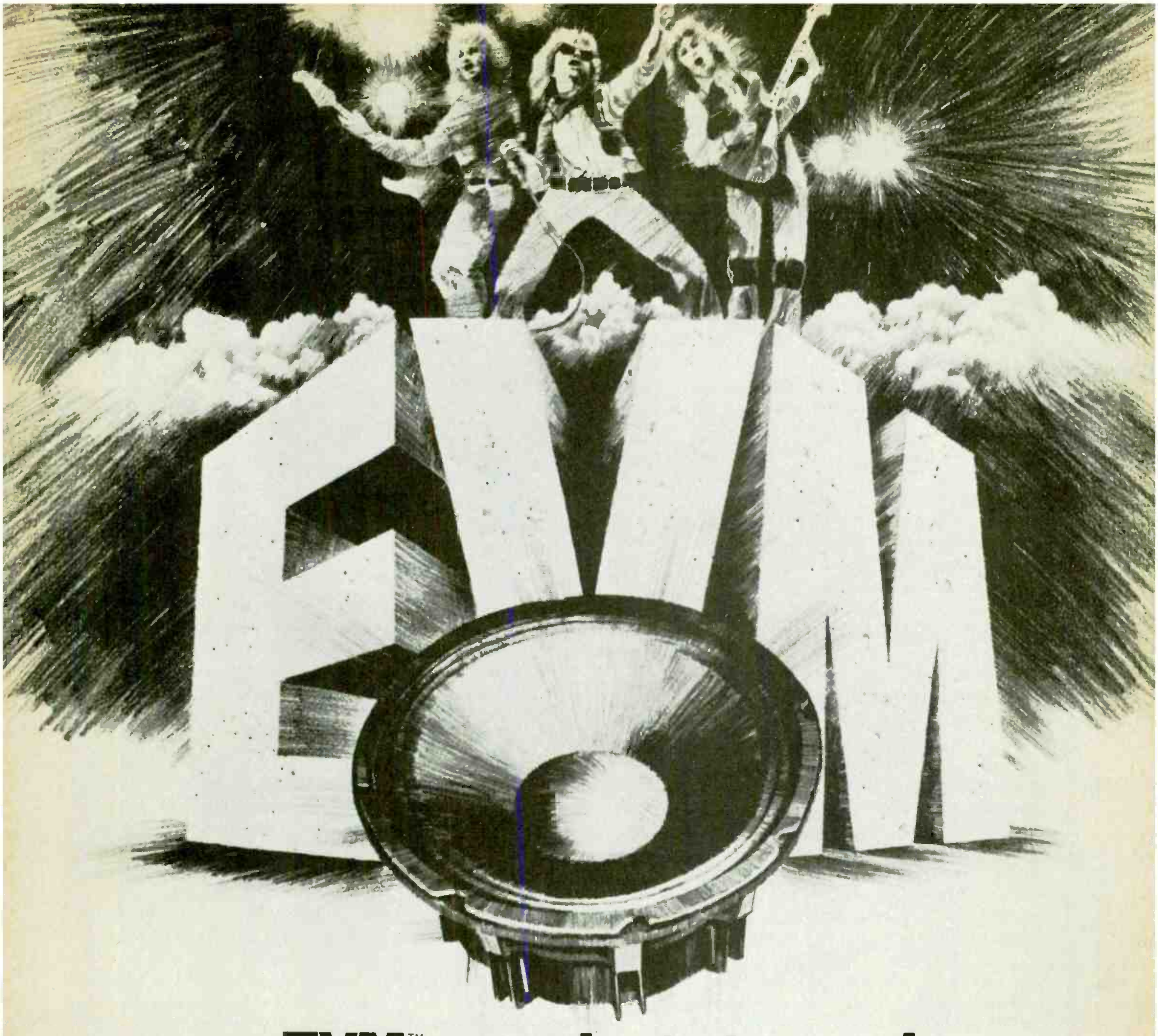
Fig. 3: Logical Systems 8801: Combined frequency sweep (20 Hz to 20 kHz) plus pink noise reveals action of 8801 at various settings of threshold control.



Logical Systems 8801: Section of rear panel.

LOGICAL SYSTEMS 8801 DYNAMIC NOISE FILTER: Vital Statistics

PERFORMANCE CHARACTERISTIC	MANUFACTURER'S SPEC	LAB MEASUREMENT
Hiss reduction at 10 kHz	15 dB	14.5 dB
Rumble reduction at 10 Hz	20 dB	17.0 dB
Maximum filter slope	9 dB/oct.	8 dB/oct.
Mono bass crossover	150 Hz	Confirmed
Mono bass slope	6 dB/octave	6 dB/octave
S/N ratio, re: 2.0 V	75 dB	78 dB
THD: (re: rated output)	0.1%	Varies from 0.0057 to 0.3 @ 1 kHz; 0.35 @ 20 Hz; from 0.08 to 1.8% @ 20 kHz
I M distortion	0.1%	0.3%
Max output	10 V into 10 K	Confirmed
Input impedance	47 K ohms	Confirmed
Output impedance	600 ohms	Confirmed
Power consumption	8 W	7 W



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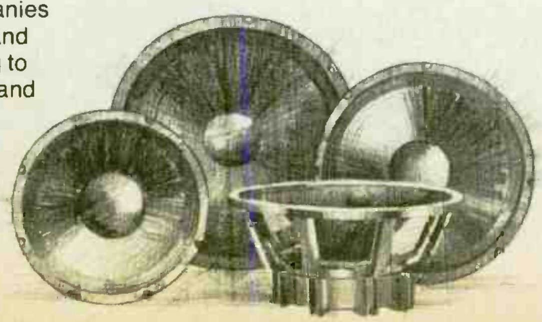

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An Overview of Crossovers

By John Murphy and Jim Ford

This month we are dedicating the "Hands-On-Report" to the subject of loudspeaker crossovers. In upcoming months, we will be reviewing a number of currently available electronic crossovers, and we felt that it would be appropriate to first give our readers a discussion of the factors important to the selection and use of crossovers. We also feel an obligation to point out that many of the currently available units share a fundamental design flaw which prevents the complete loudspeaker system from delivering a flat frequency response through the crossover region.

The discussion will begin with crossover basics where we'll explain the difference between active and passive crossovers and look at the advantages of biamplification. Then we'll examine the problems that arise when you try to recombine the frequency spectrum after the crossover has divided it. Having pointed out the flaws in the most popular designs, we'll then talk about some well-behaved crossover types and ways of employing the "problem" crossovers to get the least frequency coloration.

With this information under your cap, you should be able to select a good crossover, establish an appropriate crossover frequency and interconnect the components of the loudspeaker system in such a way as to obtain the best performance of which they are capable.

Crossover Basics

The frequency range of human hearing spans from about 20 Hz to 20,000 Hz (20 kHz); likewise, the music we produce and enjoy contains audio information over approximately the same range. Now, in order to accurately reproduce this musical information it is necessary for the loudspeaker system (and the other components in the signal path) to reproduce the full spectrum.

Unfortunately, most loudspeaker drivers are incapable of accurately reproducing the entire 20 Hz to 20 kHz spectrum. This is because the characteristics that make a driver good for reproducing one frequency extreme make it unsuitable at the other extreme. For example, good low-frequency drivers need to be rather large in order to move a lot of air; but high-frequency drivers need to be small to maintain a wide radiation pattern at the highest frequencies. This means that good low-frequency reproducers tend to perform poorly on highs and good high-frequency reproducers tend to perform poorly on lows. For high quality audio reproduction we are left with hardly any choice other than to use separate drivers to reproduce the frequency extremes.

In combining several loudspeaker drivers into a system, our goal is to obtain an accurate (that is, "flat") response across the complete audio spectrum. Although it is possible to use any number of drivers in a loudspeaker system, the most popular designs employ two, three or sometimes four drivers to make what are called two-way, three-way or four-way systems, respectively. Once it is known how to combine two drivers to produce a wide range response it is a simple matter to repeat the procedure and combine this two-way system with one more driver to produce a three-way system and so on. Considering that more complex systems are just an extension of the techniques used to create a two-way system, we will restrict our discussion to two-way systems. Now let's see how two drivers can be combined into a system.

The simplest multiple driver system would employ two drivers: a high-frequency driver (referred to as a "tweeter") and a low-frequency driver ("woofer"). Our first inclination might be to apply the full spectrum signal from the amplifier to both of the drivers by merely wiring them in parallel, but this approach would have some serious problems. First, in the range where the frequency response of the two drivers overlaps, there would be too much output from the system. That is, the system would have a "peak" in the overlap region. A second problem is that, in general,

the drivers will have different sensitivities. This means that one driver would play louder than the other. Since high-frequency drivers tend to be more sensitive than low-frequency drivers, the tweeter would probably play louder than the woofer. To make matters worse, the full-spectrum power-handling capability of most tweeters is quite limited. This is because the lower frequencies drive the tweeter's "cone" through very large displacements and can damage the unit at any significant power levels. Based on the power handling consideration alone it would not be acceptable to drive the tweeter with the full-spectrum signal. The only acceptable solution is to filter out the low frequencies from the signal applied to the tweeter. The loudspeaker crossover performs this function while controlling the overlap in response of the two drivers.

Multiamplification

There are two distinct ways to implement a loudspeaker crossover. The simplest and most often used approach is to perform the crossover function between the power amplifier and the loudspeaker drivers as shown in *Figure 1.* This type of crossover employs only passive components (resistors, capacitors and inductors) and acts directly on the speaker level signal from the power amplifier. Its main advantage is that it allows one power amplifier to drive a complete full-range loudspeaker system.

The alternative to a passive crossover is multiamplification (also referred to as biamplification, triamplification, etc.). In a multiamplified loudspeaker system there is one power amplifier for each driver and the crossover filtering is performed between the preamp (or mixer) and the power amplifiers as shown in *Figure 2.*

The crossover consists of a pair of electrical filters which modify the frequency response of the signals applied to the drivers. The full-spectrum signal passes through a "high-pass filter" on the way to the tweeter while the signal to the woofer passes through a complementary "low-pass filter." The high-pass filter passes information above a selected frequency, referred to as the "crossover frequency," and attenuates the components of the signal which fall below the crossover frequency. Similarly, the low-pass filter passes information below the crossover frequency and attenuates frequency components above the crossover frequency.

An easy way to understand the crossover is to imagine an ideal pair of filters which direct all the musical information above the crossover frequency, say 1 kHz, to the tweeter, and directs all the information below 1 kHz to the woofer. When full-spectrum music is applied to the system, the woofer would reproduce everything below 1 kHz and the tweeter would reproduce everything above 1 kHz. Because the overlap in the response of the drivers has been eliminated, the system will exhibit a smooth transition from the woofer to the tweeter. The difference between real crossovers and this ideal crossover is that a real high-pass filter passes some of the information below the crossover frequency to the tweeter. Likewise, a real low-pass filter will pass some of the information above the crossover frequency to the woofer. The further a frequency is from the crossover point the less information is passed to the "wrong" driver.

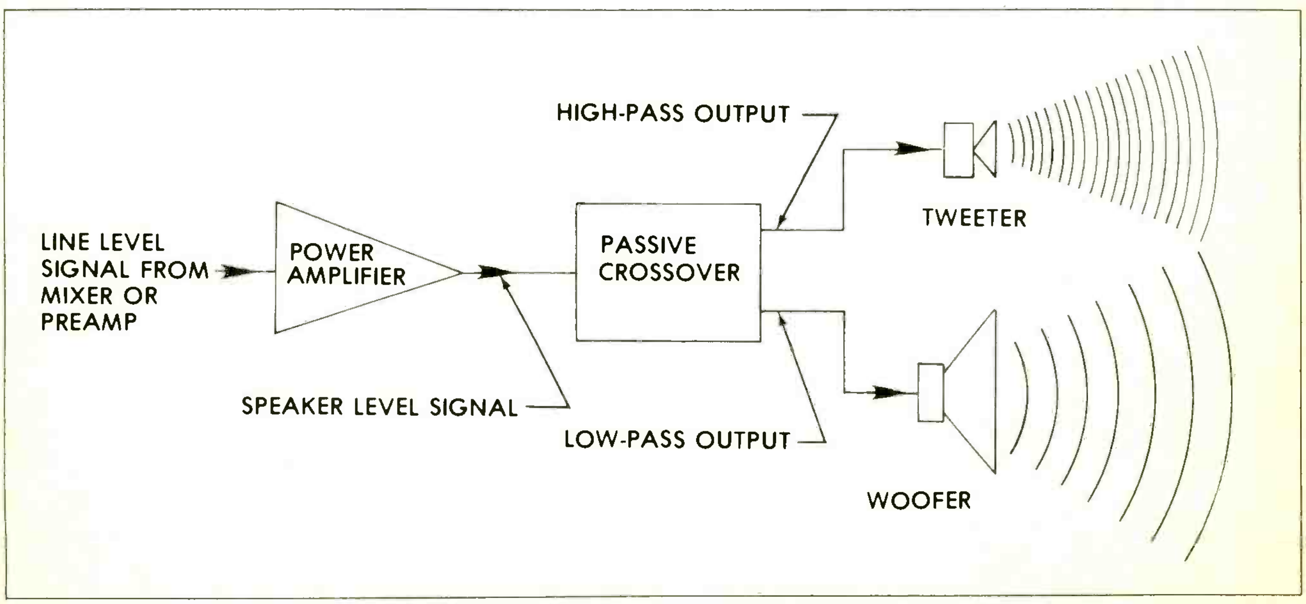
Crossover filters have the characteristic that for each octave we go in frequency beyond the crossover point the response of the filter is reduced by some fixed amount. This characteristic is referred to as the *slope* of the filter. Typical crossovers have slopes of 6, 12 or 18 dB per octave. In other words the filters reduce response by 6, 12 or 18 dB with each octave change in frequency beyond the crossover point. (Remember that either a doubling or halving of frequency represents one octave.) As an example, consider the response of a pair of filters with a crossover at 1 kHz and filter slopes of 18 dB per octave. At the crossover point the response of each filter is down 3 dB. Above 1 kHz the crossover will reduce the response of the woofer at a rate of 18 dB per octave. This means that at 2 kHz the response of the woofer will be reduced 18 dB and at 4 kHz the response will be reduced 36 dB compared to the response at the crossover frequency (−3 dB). Likewise, the response of the tweeter will be reduced 18 dB at 500 Hz and 36 dB at 250 Hz. In comparison, a 6-dB-per-octave crossover would reduce the response of the woofer only 12 dB at 4 kHz and the response of the tweeter would be down only 12 dB at 250 Hz. Because of tweeter power handling considerations, the sharper crossover slope of 18 dB per octave would allow the use of a lower crossover frequency than either 6- or 12-dB-per-octave filter slopes. An alternative view is that for a given crossover frequency, the greater the filter slope, the greater the tweeter power handling capability.

Multiamplification offers many advantages over systems employing passive crossovers. One very serious problem with passive crossovers is the fact that the filters are terminated by loudspeakers rather than simple resistors. This means that the driver is an integral part of the passive filter and that the driver's electrical characteristics strongly affect the response of the filter. This is not the case with a multiamplified system where the active filter crossover terminates its own filters. The response of the active filters can be established quite precisely and is totally independent of the loudspeaker driver's electrical characteristics.

Just as the loudspeaker driver affects the response of the passive crossover filter, the passive filter has an effect on the response of the driver. Consider a woofer sub-system which is powered through a passive low-pass filter. Ideally, the filter would only serve to attenuate the high-frequency response of the woofer above the crossover frequency. However, because of the series resistance of the low-pass filter the low-frequency response of the woofer will also be affected. The unwanted series resistance, even though it might be rather small (maybe 1 ohm), will effectively reduce the damping factor of the amplifier and therefore raise the total "Q" of the woofer sub-system. Unless the loudspeaker designer has made an allowance for the additional resistance in the initial design the result could be a peak in the woofer's low-frequency response. Also, some of the amplifier power will be wasted in the crossover resistance. In a multiamplified system the drivers are connected directly to the amplifiers and these problems are avoided.

Some of the other benefits of multiamplification are the result of using separate amplifiers for each driver. For example, high-frequency distortion products from the woofer amplifier are not reproduced by the tweeter. The woofer system can actually be driven into significant amounts of distortion and, as long as the tweeter system is not overdriven, listeners will perceive clean sound (up to a point at least). This characteristic of multiamped systems makes them especially attractive when high sound pressure levels are required (such as for concert sound systems, P.A. systems or studio monitoring). Also, because each of the amplifiers handle only a portion of the complete frequency spectrum, the amplifiers intermodulation (IM) distortion will be reduced.

A proposed advantage of multiamplified systems that is not well agreed upon among audio professionals concerns a power advantage gained over single amplifier systems using the same total amount of power. One group examines the power available to



ig. 1: A speaker system employing a passive crossover. (Note that the system is driven by only one power amplifier.)

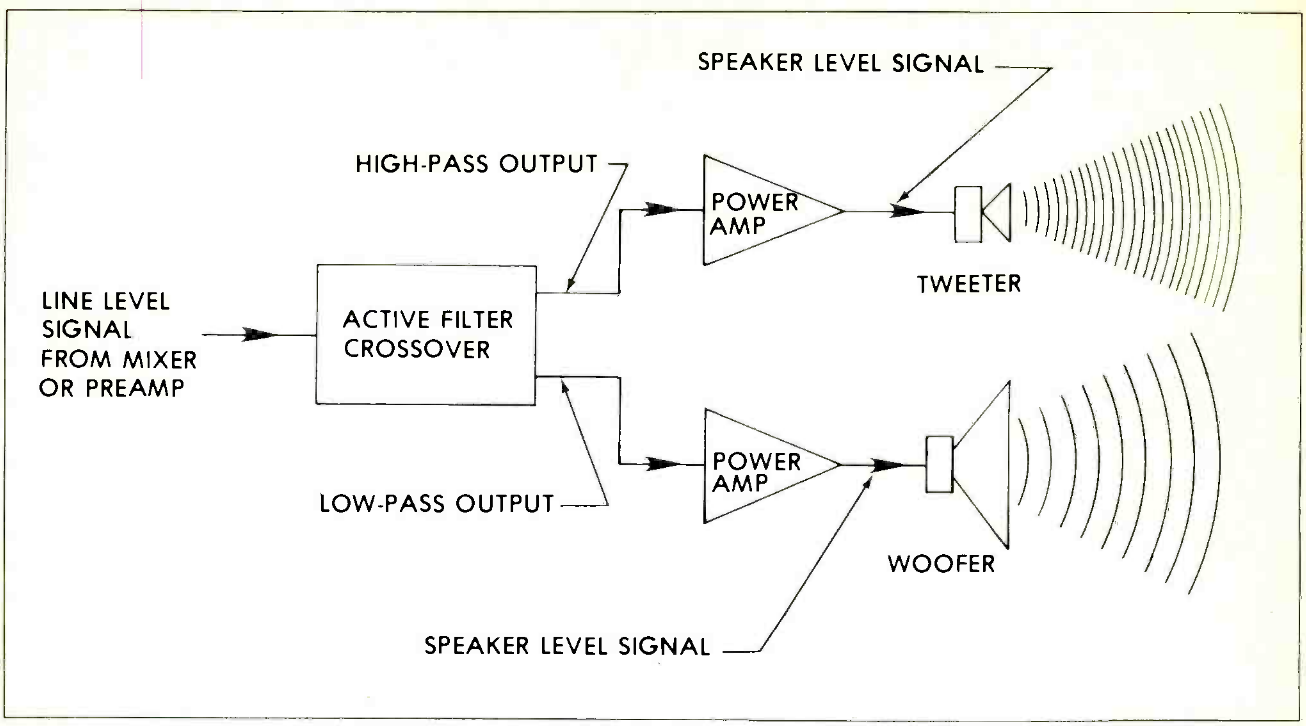


Fig. 2: A multiamplified speaker system. (Note that there is only one power amplifier for each driver.)

reproduce signal peaks and declares a definite power advantage for the multiamplified system; another group performs an analysis based on average signal levels and says there is no power advantage. Until someone performs a rigorous analysis of the two approaches taking into consideration the dynamic nature of music the question will remain unanswered. Considering that music is quite dynamic with a high ratio of peak levels to average levels, we are most inclined to think that there is some *effective* power advantage with multiamplified systems. The one point that there seems to be little disagreement on is that multiamplified systems sound better.

NOMINAL DRIVER DIAMETER	CONSERVATIVE UPPER LIMIT	HIGHEST RECOMMENDED FREQUENCY
18"	576 Hz	1.14 KHz
15"	720 Hz	1.48 K
12"	863 Hz	1.73 K
10"	1079 Hz	2.16 K
8"	1.23 K	2.46 K
6"	1.73 K	3.46 K
5"	2.16 K	4.32 K
4"	2.88 K	5.76 K

Table 1: Highest recommended crossover frequencies for loudspeaker drivers of various diameters.

Crossover Frequency Selection

In order for the outputs of two drivers to be combined to produce a flat wide-range response it is first necessary for each driver to have a smooth response and that there be a generous amount of overlap in the responses of the two drivers. For the best results each driver should have a smooth response for at least one octave beyond the crossover frequency and for two octaves if possible. Less overlap than this will result in a rough response through the crossover region. Based on overlap considerations it is best to set the crossover frequency at the center of the overlap region. It is also necessary to consult the manufacturer's "lowest recommended crossover frequency" for the tweeter, as operation below this frequency will probably give poor results.

Because loudspeaker drivers start to become directional with increasing frequency it is necessary to establish an upper usable frequency for woofers and midrange drivers. In Table 1 we have listed "conservative" and "highest recommended" upper frequency limits for various driver diameters. Using most drivers above these limits will result in rather narrow radiation patterns (i.e., "beaming").

Now, after introducing you to several design problems and techniques and the above-mentioned table of parameters, we will leave you—until next month—to mull over the crossover information. Next month we will describe some of the problems with present crossover designs and their possible solutions.

GROOVE VIEWS

Reviewed By:
ROBERT HENSCHEN
NAT HENTOFF
JOE KLEE
ALLAN KOZINN
STAN SOOCHER

POPULAR

SUE SAAD AND THE NEXT: ***Sue Saad And The Next.*** [Richard Perry and James Lance, producers; Gabriel Veltri, engineer; recorded at Studio 55 in Los Angeles, Ca.] Planet P-4.

Performance: **New Wave chic**
Recording: **Richard Perry chic**

This band seems packaged for success in the changing rock marketplace. Foxy Sue Saad is a little naive, a little nasty, and can belt her basic rock & roll with convincing urgency. The boys in the band have that same kind of teen appeal: French cut t-shirts, satin pants, and brash good looks. Together, they rock near the fringes of New Wave and show moments of both promise and fulfillment on this Planet debut.

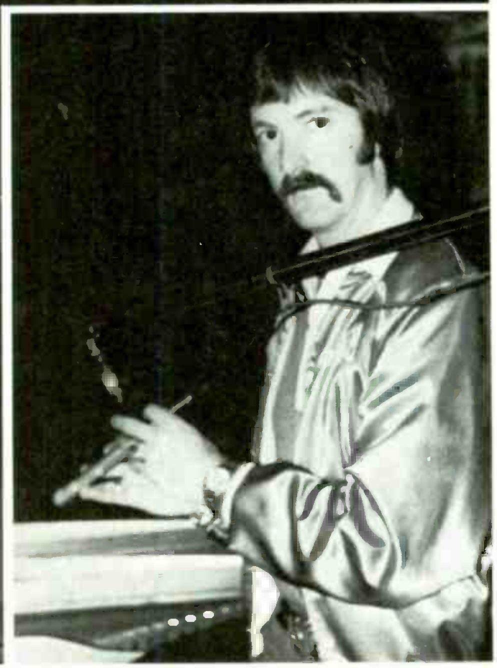
There has been an explosion of female-led New Rock units. Sue Saad And The Next may not quite have the stunning originality of The Pretenders or The B-52's, but the flare is definitely there, the instrumentation is fairly hot, and most of the cuts are quite good. "It's Gotcha" and "Prisoner" contain hard rock hooks that may prove attractive to AOR programmers. "Young Girl" has a punky reggae funkiness that should be a show stopper too.

Saad's dusky voice functions well in the rock context. She sounds a lot like Olivia Newton-John did on her breakthrough *Totally Hot* album last year—and that's meant as a compliment, surprisingly enough. Behind this successful woman, there's a man, drummer James Lance, who co-produced the session, wrote on eight of ten songs, and provides a potent rhythmic launchpad for The Next. Up front are twin guitars and bass, all bridging New Rock to the traditional.

"Gimme Love/Gimme Pain" and "Your Lips-Hands-Kiss-Love" start each side of this disc with Sue Saad in sensual situations. Much of the group's music builds on that tempestuous old boy-girl relationship, and Ms. Saad sings her part aggressively. The group sound is invigorating even when the sometimes simple lyrics begin to wear thin, and the Richard Perry production shows an appropriate adaptability. While there are several exciting new albums that should be bought before this one, *Sue Saad And The Next* should be in the running. R.H.

SUE SAAD AND THE NEXT: Basic rock and roll: naive and nasty

fact:
"I listened to them all... and nine times out of ten, with our artists, the best microphone was the SM81"



Criteria Recording Studios,
Miami, Florida

Dennis Hetzendorfer

Dennis Hetzendorfer,
Staff Engineer

"The true sign of a really excellent microphone is that it can *maintain* its high performance, session after session after session. Here at Criteria, when the situation permits, several different microphones are set-up at each instrument, without the engineer knowing which mike is exactly where. We then fade from mike to mike and let our ears find out which is best for each application. Nine times out of ten, with our artists, the best microphone has been the SM81.

"The switchable bass rolloff and pad (a built-in 10 dB attenuator) gives the SM81 incredible versatility. We can use it with bass drums and cymbals, as well as with acoustic guitars. In fact, all the acoustic guitar segments on the Bee Gees' *Spirits Having Flown* album were recorded with the SM81.

"The SM81 really changed our minds about the ruggedness of condenser microphones. It's a precision piece of equipment, but it's durable. You don't always think about a studio microphone needing durability... after all, we don't have the rough handling problems encountered in concert recording. But, when you have a reputation as one of the most technically exacting studios in the country, you appreciate how *many* little things can subtly affect the sound of a delicate condenser microphone. The SM81 sounds good *every* time we use it... and, at Criteria, as in any good studio, we just can't afford to have a microphone we can't depend on.

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CIRCLE 118 ON READER SERVICE CARD

BOB SEGER AND THE SILVER BULLET BAND: *Against The Wind.* [Bob Seger, Punch, The Muscle Shoals Rhythm Section, and Bill Szymczyk, producers; John Arrias, Steve Melton, and Bill Szymczyk, engineers; recorded at Criteria Studios, Miami, Fla., Muscle Shoals Sound Studios, Sheffield, Ala., and Bayshore Studios, Coconut Grove, Fla.; mixed at Capitol Studios, Hollywood, Ca., Bayshore Studios, and Muscle Shoals Sound Studios.] Capitol SOO-12041.

Performance: **Hot rocking**
Recording: **Clean, but compliments Seger's gritty vocal style**

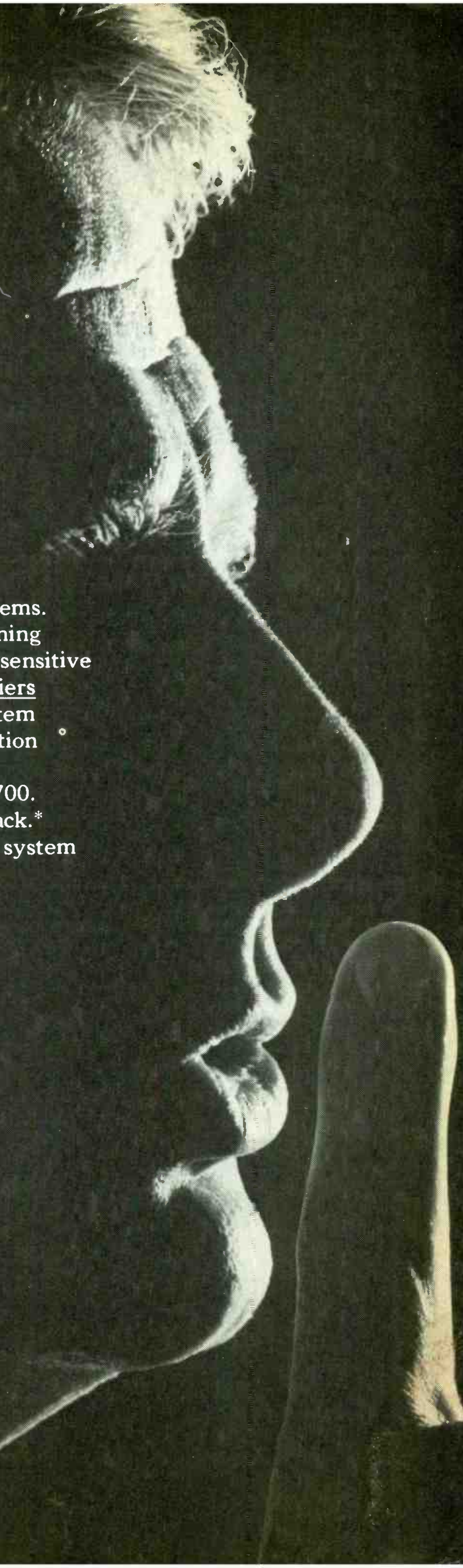
Who wants to play those eight
 and aces
Who wants a raise
Who wants a stake
Who wants to take that long
 shot gamble
And head out to Fire Lake
—from "Fire Lake" by Bob Seger

Fred Schruers, a writer I once worked with at a rock magazine and who now freelances for *Rolling Stone* and the *New York Daily News*, had a habit of walking on his hands. He could cross a room twice as fast this way, it seemed, even if he occasionally bumped into a wall. Then he would just as quickly disappear under a desk with a telephone he had once broken in half by casually (he thought) slamming it down on the receiver. He resurfaced only to slap another record on a rather motley turntable.

Why talk about Fred Schruers in a review about Bob Seger? The point is that both Schruers and Seger know that by butting your head against the wall and taking the knocks, you eventually find your way out of what might appear to be a hopeless trap. In his lyrics, Seger has always played the underdog who uses rock as a way of evening up the odds. This has made him, in a sense, like Bruce Springsteen, a rock reactionary reaffirming from album to album unextinguishable sexual posturings as relief for the proletarian struggles of his audience. Seger rose, after all, like a phoenix out of the ashes of a troubled Detroit in the late '60s and honed his message from *Ramblin' Gamblin' Man* through *Beautiful Loser* and *Night Moves* to, now, *Against The Wind.*

Seger's bluesy, growling vocal style is the perfect counterpoint to his lyrical ideas. And working with both his regular Silver Bullet Band and the Muscle Shoals Rhythm Section on *Against The Wind* (as he did on *Night Moves* and *Stranger In Town)* has provided him with a series of powerful backing tracks to wrap his images around. The grandiose keyboards of Paul Harris on the title tune and the New Orleans-style piano of Mac "Dr. John" Rebennack on "Horizontal Bop" are just as effective. Normally, dealing with several sets of producers on any project can present special problems, but the final product here doesn't sound piecemeal in any way.

"Horizontal Bop" opens the proceedings on *Against The Wind* with a salute to '50s rock and roll which resurfaces later in "Betty Lou's Gettin' Out Tonight," a Little Richard tribute about a good-time girl who causes the local drug store to run out of prophylactics when her mother lets her out of the house. "Her Strut" contains an unusual grunting guitar, "No Man's



Land" and "You'll Accomp'ny Me" jump quite nicely from simple acoustic guitar introductions to full blown productions, and "Shinin' Brightly," Seger's "Oh, Happy Day," ends the album on a positive lyric note. "Good For Me," however, is not as strong a ballad as "We've Got Tonite" from *Stranger In Town.* Blame that on the fact that Seger hasn't stretched his style one bit on his new album, and so a song like "Long Twin Silver Line" touches on monotony. So maybe Bob Seger should start writing his songs while walking on his hands? S.S.

BILLY JOEL: ***Glass Houses.*** [Phil Ramone, producer; Jim Boyer with Bradshaw Leigh, engineers; recorded at A & R Studios, New York, N.Y.] Columbia FC 36384.

Performance: **Classic Joel professionalism**
Recording: **From lush to lean**

Imagine Frank Sinatra leaning over a piano in a dimly-lit bar, drink in one hand and a microphone in the other, putting his rendition of Billy Joel's "Just The Way You Are" (which he included on his latest release, *Trilogy: Past, Present, & Future*) through the paces. "I don't want clever . . . clever conversation," he croons. The pianist looks familiar to you, but you can't quite place his face because his features are hidden by a cloud of cigarette smoke. Just then the pianist knocks the drink out of Sinatra's hand, the glass breaks, and Billy Joel (!) launches into the tough rock tunes of *Glass Houses.*

Confusing? Maybe, but this is the way Joel wants it. He has based his success on bringing the emotional anonymity of lounge music to the defiant arena of rock, protecting himself from the loneliness of the former with the communalism of the latter. *Glass Houses* is not a wimp's dream by any means, however. It is every bit as threatening as Joel's last album, *52nd Street.* "You May Be Right" (the heir to "My Life" with a guitar riff straight out of the Beatles' *Revolver*), "Sometimes A Fantasy," "All For Leyna," and "Close To The Borderline" are all about living life on the edge of reality. "It's Still Rock And Roll To Me," a send-up of new wave bands like the Cars and the Knack, claims that the changing sounds and various genres of rock are really no more than variations on the same theme ("Don't waste your money on a new set of speakers/You get more mileage from a cheap pair of sneakers"). A series of sound effects add some ragtag realism to *Glass Houses* with a phone being dialed on "Sometimes A Fantasy" (compare to "Preface" from Robert Fripp's *Exposure*), breaking glass opening the album (a theme explored by Nick Lowe on *Pure Pop For Now People*), and a television sign-off starting "Sleeping With The Television On" (whose lyrics mention "white noise").

Still, we find Joel in his old cocktail haunts throughout *Glass Houses.* On the McCartneyesque "Don't Ask Me Why," he's in "your grand cafe." "I Don't Want To Be Alone" takes place in "The bar/At the Plaza Hotel." "C'Etait Toi (You Were The One)"—

perhaps the best utilization of French in a rock ballad since the Beatles' "Michelle"—finds Joel "in this smoky place/with my brandy eyes." A glass house may symbolize a fragile character, then, but Joel isn't afraid to admit in "Sleeping With The Television On": "I really wish I was less of a thinking man/And more a fool who's not afraid of rejection." Hey, my eyes are starting to sting from the smoke in this place. Is that Sammy Davis, Jr. over there with Sinatra, or is it really Garret Morris? S.S.

JAZZ

CHARLIE BYRD: *First Flight.* [Ozzie Cadena, producer; Bob Porter, reissue producer; Rudy Van Gelder, engineer; recorded Feb. 4, 1957 in Hackensack, N.J.] Savoy SJL 1131.

Performance: **Early Byrd gets the goodies**
Recording: **The best of Rudy Van Gelder**

Has it really been more than 20 years since Charlie Byrd showed up at Van Gelder's recording studio and made his first sides for Savoy? It really doesn't seem that long. For a realistic essay on the life and times of the acoustic guitar, I refer you to Michael Rozek's thorough and intelligent liner notes. For an essay on the wonders of the acoustic guitar, I refer you to Charlie Byrd's playing on this LP. Whether it is the ballad wizardry of Rodgers and Hart or Les Freres Gershwin or the nearly classical Spanish guitar work of Byrd's "Prelude" or "Interlude," Byrd's mastery tempts one to dub him the Segovia of the Jazz Guitar. That two examples of his electric playing are included is not really a terrible shame. Charlie is well aware that the electric guitar and the acoustic guitar are two completely different instruments operating in totally different ways. As such, his acoustic playing is different from his electric playing both in technique and concept.

While a lot of this LP is solo playing, some of it is accompanied by a capable flute player whose tenor sax work is far more than just capable. And who does it turn out to be but Doc Severenson's right hand man on the *Tonight Show,* Tommy Newsom! For an example of Newsom at his best, check out Byrd's "Homage to Charle Christian." Byrd's electric guitar and Newsom's tenor riff along in unison, sounding very much like a bopper's edition of the lines that Charlie Christian contributed to Benny Goodman's Sextet some years earlier.

Anyone who has been reading my reviews for any length of time knows my feeling that the major problem involved in recording acoustic guitar is the excess of pick noise and string slide that is picked up when an acoustic guitar is miked closely enough to give a full sound to the instrument. How Rudy Van Gelder has avoided this pitfall is one of his secrets which he'd do well to bottle up and sell to the rest of the industry.

Charlie Byrd's philosophy can best be summed up in the quote with which Mike Rozek brings his liner notes to a conclusion: "Why must the guitar be confined to single-line playing and rhythm accompaniment? If the guitar is going to take an important place as a solo instrument, it must be broadened! Leave the sax sound to the sax and play the guitar! Its capabilities were proven centuries ago." And had they not been proven centuries ago, Charlie Byrd's work on this recording would have rectified that situation nicely. J.K.

TOMMY DORSEY: *The Complete Tommy Dorsey, Vol. 1.* [Frank Driggs, reissue producer; Don Miller and Ed Begley, remastering engineers; original recordings made between September 26, 1935 and July 10, 1937, primarily in New York, N.Y.] Bluebird AXM 2 5521.
TOMMY DORSEY: *The Complete Tommy Dorsey, Vol. 2.* [Same production credits as above.] Bluebird AXM 2 5549.
TOMMY DORSEY: *The Complete Tommy Dorsey, Vol. 3.* [Same production credits as above.] Bluebird AXM 2 5560.
TOMMY DORSEY: *The Complete Tommy Dorsey, Vol. 4.* [Same production credits as above.] Bluebird AXM 2 5564.

Performance: **Vintage music for dancing and swinging and listening**
Recording: **Better than some of Victor's '30s products, but still not up to the best**

They called Tommy Dorsey the "Sentimental Gentleman of Swing" because

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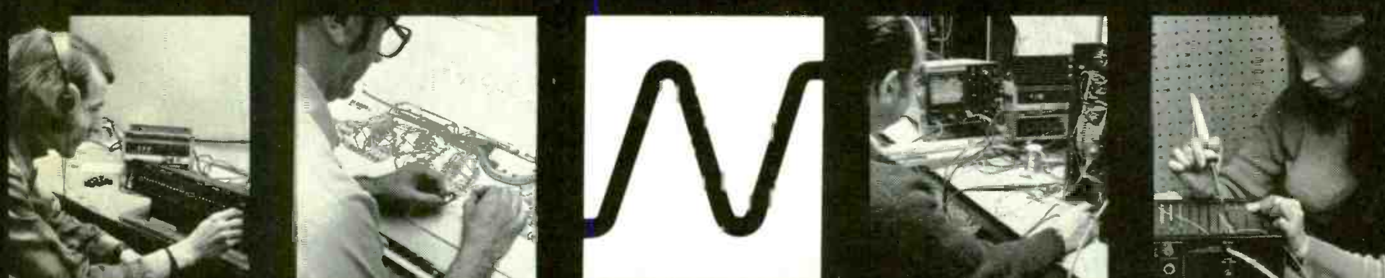

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The Return of the Legendary Fiddler and a Neglected Lady

By Nat Hentoff

Count Basie still talks of violinist Claude Williams with delight—and regret. Regret because Claude, who came East with the Basie band in 1936, didn't stay. He went back to Kansas City and so missed fame. Musicians kept talking about him though, and Claude became a kind of legend. ("You want to hear some really *hot* fiddle, there's your man!")

In 1972, however, the Canadian label, Sackville—a model of how jazz should be recorded in terms of artistic freedom and engineering quality—made a session with Williams and the magisterially swinging KC pianist-singer, Jay McShann. The fiddler wasn't just a legend any more; he was real. And since then, Claude Williams and McShann have played throughout Europe, with Williams enjoying a particular triumph at the 1979 Nice Festival when he more than held his own with Stephane Grappelli and Svend Asmussen.

Sackville has now made that 1972 date newly available. *The Man from Muskogee* (both Williams and McShann were born there) is a marvelously buoyant, timeless distillation of classic swing. Williams is indeed one of the hottest of all violinists in jazz history, and it's intriguing to speculate what the vintage Basie band would have sounded like with that added tangy play of string colors (Williams sometimes sounds like a whole, sizzling violin section). McShann is easefully powerful, and there is attentive support from bassist Don Thompson and Paul Gunther. The recorded sound is utterly, naturally clear and spacious.

Unlike Claude Williams, Ernestine Anderson has not been limited to only one particular regional scene for decades. She makes a lot of gigs in this country and in Europe, but she tends to be overlooked by the arbiters of jazz renown—the reviewers and critics. Yet, in *Sunshine* (Concord Jazz), Anderson has created one of the most satisfying *jazz* (not just jazz-influenced) song recitals in a long time.

With horn-like phrasing (that in no way distorts the vocal lines) and surely swinging time, she sings with warmth, wit, and an often jubilant authority that few singers can approximate. This is a woman who has found her very own sound and style, and now keeps challenging herself to see how much more she can discover about her capacities.

The firm, resilient background consists of bassist Ray Brown, pianist Monty Alexander, and drummer Jeff Hamilton. The repertory ranges from "God Bless The Child" (done in *her* way) and a strutting, crackling "I'm Walkin' " to "I've Got The World on a String" and a deeply sensuous "I Want A Little Boy."

As is customary in the collected works of Concord Jazz's president, Carl E. Jefferson, much care is taken with the engineering to make sure it keeps the music intact rather than becoming an instrustive part of it. As usual, the sound is first-class.

JAY McSHANN, CLAUDE WILLIAMS: *The Man From Muskogee.* [John Norris, producer; Phil Sheridan, engineer.] Sackville 3005.

ERNESTINE ANDERSON: *Sunshine.* [Carl Jefferson, producer; Phil Edwards, engineer.] Concord Jazz CJ-109.

he and his band could do both. Maybe their swing wasn't as hot as Benny Goodman's and maybe their sweet tunes weren't as smooth as those of Hal Kemp's band, but no band could do both as credibly as Dorsey's. On the hot side, he had soloists such as Pee Wee Erwin, Max Kaminsky, Sterling Bose, Johnny Mince and Bud Freeman to spice things up. When it came to the sweet side Tommy relied mainly on his trombone work—and how sweet it was!

On September 26, 1935, Tommy Dorsey (newly split up from his brother Jimmy with whom he had been co-leading the Dorsey Brothers Orchestra for some years) brought his young band (mostly taken over from Joe Haymes recently defunct crew) into Victor's New York studios. They cut three tunes. One of them, "Santa Claus Is Comin' To Town" was obviously geared for the Christmas trade and, backed with Benny Goodman's jazzed up version of "Jingle Bells" it must have enjoyed a healthy sale in the winter of 1935. There was a plug tune which made the western hit parade for awhile, "Take Me Back To My Boots And Saddle" featuring a dreadful vocal by trumpeter Cliff Weterau (Cliff Weston on the record) who didn't last long with the Dorsey band. The other tune waxed was the dixieland standard "Weary Blues" in a swing band arrangement by Spud Murphy. There you have the Dorsey success formula. He gave both sides of the dance floor the kind of tunes they wanted and whether they danced cheek to cheek or jitterbugged, they could count on Tommy Dorsey's Orchestra for the necessary music. "Weary Blues" featured some healthy attempts at hot playing by Sid Stoneburn on clarinet and, I would guess, Johnny Van Eps on tenor sax but by December the band could boast of a hot soloist like Sterling Bose on trumpet.

By Volume 2 we find two of the giants of Chicago jazz in the band—Dave Tough on drums and Bud Freeman on tenor sax. Bud's arrival was celebrated with a special introduction by Edythe Wright, the band's perky girl singer, on Dorsey's record of "At The Cod Fish Ball" and some remarkable solo work by Freeman on that side. Early 1936 also brought a superior male singer, Jack Leonard, into the band.

When Johnny Mince finally replaced Joe Dixon on clarinet, the Dorsey hot front line was in place. If my preference runs to the Clambake Seven sides over those of the full band, that's only my

personal love for the small dixieland format. You'd find as many dancers telling you that they preferred the way Tommy's big band played their latest favorites with those wonderful Jack Leonard vocals.

These first four double-LP sets in what, it is hoped, will be the eventual issue of all the known Dorsey 78 rpm records (at least in their original format, even if alternate masters aren't included), are an endangered species. If you danced to Tommy Dorsey's record of "Strangers In The Dark" or pranced to the Clambake Seven's version of "Posin'," and want to add them to your LP collection, now is the time before they are removed from the catalogue.

There is at least one error in the discographical notes that I'd like to clear up. According to the liner notes in Volume Four, Dave Tough was the drummer of the first three tunes recorded July 10, 1937 with Russ Isaacs coming in after the first (played straight) recording of "Am I Dreaming." This isn't so. Tough was not on the date and it was Isaacs all the way. This is according to Brian Rust's *Jazz Records 1897-1942*.

Also it is very interesting to hear the version of "Am I Dreaming" which closes Volume Four. Many collectors have treasured this one for years. It was originally issued anonymously backed with "Burglar's Revenge" by Bert Shefter's Orchestra. The label simply stated "Are All My Favorite Bands Playing Or Am I Dreaming" but those in the know knew that it was Tommy Dorsey's Clambake Seven taking out their musical revenge on such corny bands as Russ Morgan, Henry Busse, Shep Fields, etc. Only the names were changed to protect the players from being punched out. I sort of wish they hadn't reissued it because now that the secret is out, I won't have half as much fun trying to fool my friends who think they know as much as I do.

J.K.

CLASSICAL

[This month, in a departure from our usual multi-review format, we feature an in-depth, behind-the-scenes—and machines—glimpse at the digital recording sessions for the RCA Bartok release, Concerto for Orchestra. *P.H.]*

Philadelphia's Digital Bartok Experience

By Allen Kozinn

A couple of years ago, RCA quietly began investigating digital recording techniques. Using Sony and Technics videocassette machines modified for use as digital audio recorders, they made "safety" copies of normal sessions so that they could compare the quality of regular analog masters with tapes made with the new system. The works recorded were a Stravinsky *Firebird*, with Mata and Dallas, and some Ravel, also featuring Mata and Dallas. These discs, as of this writing, are about to be released. They have been withheld so long, according to the official story, "because we couldn't find a way of editing them."

That may have been a substantial *part* of the problem. But equally important, it seems, was that RCA did not want to simply bring out a series of digital recordings with no fanfare. Looking around the field, they saw small digital companies putting out standard or showpiece music by lesser known orchestras. RCA wanted to make more of a splash than that—which meant, essentially, that as flashy as the *Firebird* might be as a digital display, the Mata/Dallas combination was hardly luminous enough to attract proper attention. (RCA had also recorded a Mendelssohn Walpurgisnacht with Ormandy and Philadelphia—but would buyers scramble for this piece of music?)

The solution, naturally, was to shelve the experimental stuff, set up another Philadelphia recording session, and announce to the press that "for the first time, a major American label is recording a major American orchestra under a major conductor, playing a major work, using this shining new technology." The session was to take place in April 1979 and—according to the plan—the disc would hit the market in June 1979.

The repertoire? Here Tom Shepard, RCA's Vice President in charge of the Red Seal classical music division, decided to tempt the fates. Several years back, when Shepard was a producer at Columbia, he became an enthusiastic convert to quadraphonic sound, and he decided to show off *that* shining new technology by recording the New York Philharmonic, under the baton of Pierre Boulez, performing the Bartok *Concerto for Orchestra.* For these sessions, Boulez conducted from a podium in the center of the studio, with the various instrumental groups surrounding him. The result was exquisite, and Shepard is still proud enough of his effort to keep the cover of the LP (Columbia M 32132) on the wall of his RCA office. But quad never took hold, and Shepard, who says he listens to quad at home, felt that his great production feat was wasted.

Like quad, digital recording is said to bring a new kind of clarity to orchestral recording, but *unlike* quad, it was going to be made available (for now) in a format that anyone can play on a normal turntable. As before, Shepard turned to the Bartok *Concerto for Orchestra,* a dazzling essay in orchestral timbres and complex rhythms, to demonstrate the new recording technology.

The Search for Silence

And so it was that Shepard—now serving as a sort of overall supervisor—producer Jay Saks, project coordinator John Pfeiffer and recording engineer Paul Goodman went to Philadelphia on April 16, 1979. By this time, they had recorder—actually two small blue boxes labeled "BIX Edit Code Generator" and "Edit Code Reader"—and two conventional Ampex "safety" machines.

The Soundstream recorder, it was explained, recorded at 35 ips and could put about 25 minutes on a reel. One piece of information was captured for every 20 microseconds (20 millionths of a second)—an unbelieveably minute increment that allows for absolute pinpoint editing.

"The machine records," explained Bloomenthal, "by taking voltage measurements every 20 microseconds. When you hear a sound, you are responding to air pressure changes, and you interpret certain differences in pressure as variation in pitch. The machine works much the same way. It hears pitch variation and records it as changes of frequencies."

The Session

As the session began, producer Saks addressed the orchestra over a monitor, advising them that "nothing will be different from the way we normally record. The differences are all in the storage of the sound." To which conductor Ormandy (upon whose podium the RCA team had trained a closed-circuit TV camera) added, "I think this new system will allow us to get better balances and to capture all the details—even the quiet *pianissimos*—perfectly."

But as things got underway it became apparent that not all was well. During a run-through of the last movement (the first to be recorded) the yellow LED indicators on the Soundstream machine stopped registering, and it was a few minutes before anyone realized that something was wrong. Apparently, the close-quarters of the ladies'-room-cum-control-room had caused the unit to overheat. The solution turned out to be almost comical: to cool the Soundstream recorder, an ancient cast-iron electric fan had to be set up and aimed at the streamlined blue box.

There were some other limitations. According to Bloomenthal, Soundstream is continually experimenting with new computer programs to make the recording process easier, and the company is also working hard to build new and bigger recorders—a matter of pressing necessity, because the small firm is being hired to oversee an increasing number of digital sessions. The current Soundstream recorders can handle only two channels. At this session, two units were used, seemingly giving RCA a four-channel capacity. But not quite. According to Shepard and Bloomenthal, mixing

decided on a good, editable recording system, the widely used Soundstream, created by Dr. Thomas Stockham (whose name first graced RCA liners when he began using a computer process to clean up and equalize the old Caruso recordings). Representing the Utah-based Soundstream company at the session was a young engineer, Jules Bloomenthal.

"This is not about a search for a better way of recording sound," Shepard explained as the engineers prepared for the session. "This is about the search for silence. We believe that the job of recording an orchestra should remain in the hands of a capable producer. But we want a system that will not introduce noise and hiss. The recording system should have no personality of its own."

The session took place in a large auditorium of Philadelphia's Scottish Rite Cathedral, with the orchestra seated spaciously throughout the hall, rather than on the stage. A ladies' room at the back of the hall served as a control room, and was jam-packed with equipment: there were 4 KLH studio monitors, a 16-track console, the Soundstream two two-channel tapes as if they were a single four-channel tape would be difficult, if not impossible. Instead, RCA chose to make a pair of alternative stereo mixes. It is possible, says Shepard, to edit from one mix to the other, and it is possible to edit each *channel* separately. And in any case, the RCA men almost reluctantly admitted, having to do "live" mixes imposed a certain discipline. Of the two mixes, only one was constantly monitored throughout the session, although parts of the other were used in editing the final master.

Whatever the excitement in the control room, the feeling in the auditorium was, as Saks had promised, just like that of any other session. Movements were recorded generally in full takes, with patches of ten or twenty bars at a time where either Ormandy, Saks or one of the players felt something was wrong. The playing was magnificent, and from the balcony overlooking the orchestra it seemed impossible that any recording process could really capture the grandeur of the Philadelphia sound.

During the lunch break, there was a playback. As is often noted upon listening to digital recordings, the brass was bright and crisp, and the winds—particularly the flutes—were beautifully captured. It seemed, however, that the low end was somewhat boomy, although that may have had more to do with the characteristics of the KLH monitors than with imperfections in the recording process. In any case, Ormandy seemed delighted, the players impressed and the engineers relieved. The sessions continued after lunch, finishing the middle movements and patching a few spots where players had reservations, during the playback, about their performances. Despite the difficulty of the work, the early breakdown of the equipment and the obvious tension of working with a system that promised such clarity that all warts would be revealed—and which could not be radically remixed—the entire work was completed and the sessions wrapped up on time and with a few minutes to spare.

The Realization of Brilliant Sound

The sessions, though, were only the first step in the making of the Philadelphia Digital Bartok. Within a month, Shepard and Saks were on their way to Utah to edit the master tapes—and to learn how to use the Soundstream computer, which stores the "information" captured at a recording session on a series of small, quickly revolving discs. Each series of ten magnetic discs contains six minutes of music.

"It's just amazing," Tom Shepard says of the process. "You sit in front of an editing console and give instructions to a computer, which will extract the pieces of information you call for and put them in any sequence you want. When Jay edited the Ormandy Bartok disc, I was with him, and we had the time of our lives. You can choose one point from this take and one point from that take, and narrow down to the exact spot you want to edit. To help you do this, you can turn on a video screen which displays a waveform readout; or you can just use the meters, which display a running count of seconds, milliseconds and so on. When you hear the exact spot where you want to make an edit, you just punch it into the machine.

"One of the most attractive things about this is that the equivalent of splices are not splices at all, but computer instructions. The editing is so precise that you just don't *hear* anything. What you have is a series of discs with playback heads between them. And in editing, you instruct the computer to grab one instant from this disc and another from that, and that's how you program each six-minute sequence of the work. When you have given the computer all the information it needs for a full side, then it will copy everything, exactly as you want it, onto a digital master tape, which is much more convenient, for our purposes, than the series of small discs."

The master was then returned to New York and cut on a half-speed lathe, and then sent off for pressing. Because of the special nature of the release, and although the list price, at $9.98, is only a dollar more than conventional RCA product, Shepard insisted that the disc be pressed on special top grade vinyl. Apparently, the first few pressings were not up to Shepard's standards, though, and as test pressings were sent back and forth from Shepard's office to the pressing plant, it became obvious that the June release date would not be met. Nor would the July. Or the August. Meanwhile, newspapers and magazines were running stories about digital recording featuring promotional photos of the Bartok cover; and in some cases, RCA advertising featuring the LP made it to press.

Finally, in September 1979, RCA, Eugene Ormandy, Philadelphia and Bartok made their digital debuts. As a performance of the Bartok, this recording stands up well in the face of the competition, including the Boulez/New York. There are a few places where one detects a lack of backbone—something that comes through more in the recording than at the session. In some of the quieter and less frenetic moments the orchestra seems to have let some of the rhythmic propulsion go slack. But those moments are few, and where brilliant sound is called for the Philadelphia Orchestra produces it without fail. Nor is there anything tired-sounding about the way the 80-year old conductor (who had announced his impending retirement just a few weeks before the session) leads the orchestra through Bartok's energetic sonic maze.

One reason this work was chosen was that it contains not only loud, colorful *forte* passages, but quite a bit of nearly silent sections. Recordings using conventional tape methods invariably suffer from hiss at these points, and as loud as the listener tries to make the music, the unwanted tape noise, naturally, follows right along.

Here, it is a delight to turn the sound up high and hear mostly music. I say *mostly* because, particularly at high volumes, you do hear other things too. And this brings up some questions about this whole digital craze. Until new playback systems are introduced, even listeners with expensive, top-of-the-line equipment are bound to hear the result of our current friction-on-the-groove playback system. Quiet as this and many another digital disc may be, digitally recorded tapes pressed into analog discs are only partial improvements. They are prone to dust, friction, and all those other things that make record owners cringe and records crackle. And even in this case, where special care has been taken, they are prone to bad pressings. I've had the opportunity to listen to three copies of this disc, and while I found one of them absolutely flawless, the other two came with an annoying swooshing noise at the end of the first side. One interesting note, however, is that because the discs are pressed on clear/red plastic, it is fairly easy to find defects (especially correctable ones, like something stuck in a groove) by holding the disc up to the light. —Allan Kozinn

BARTOK: *Concerto for Orchestra.* Eugene Ormandy, cond. Philadelphia Orchestra. [Jay Saks, producer; Paul Goodman, engineer; recorded at the Scottish Rite Cathedral, Philadelphia, April 16, 1979.] RCA ARC1-3421.

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