NOVEMBER/DECEMBER 1987

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NOVEMBER/DECEMBER 1987 VOLUME 21 NO.6

the recording engineer

the sound contracting engineer

2 to 8 trk the smaller recording studio

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About the Cover

• In this our twentieth anniversary issue, no look back. The cover points to the future with products that you can buy now or will be able to buy in early 1988. Clockwise from the top left: Sony PCM 3402, Tascam PRO DAT, Waveframe's Audio Frame computer digital recording/editing system, and Sony's RDAT PCL unit. Now on to the future.

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Letters

Dear Jay,

I thought your TEK TEXT #103 concerning unwanted magnetization of tape recorder parts was very comprehensive. The several causes pointed out were quite interesting. Since you invited your readers to add more possible answers to these mysteries, I would like to share these.

LIGHTNING:

A strong bolt of lightning striking a well-grounded target spits out an awesome unidirectional magnetic pulse. Any magnetic material within one hundred feet of the strike is instantly affected if not protected by mu-metal or enclosed steel cabinets.

INADEQUATE DESIGN OF RECORD SYSTEM:

Though your text mentioned the home-built pre-amp, there have also been commercial units that threw a pulse into the record head upon switching to play mode leaving the head magnetized, but not every time, of course. (Why should things be this simple?) Then, when you went back into record mode, the bias automatically demagnetized it. Finding a problem like this is not easy because of its random nature but will usually expose itself within a dozen tries by using the pop and thump test on a clean, bulk-erased tape.

LEAKAGE OF THE HEAD DRIVER CAPACITORS: (...even new ones.)

If you want a pass response down to 20 Hz, some head impedances require a hefty capacitor. I'm not talking really bad noise but maintaining noise around the area of a bulk-erased tape and having this response, too, ...you're talking real purity. You can't measure any leakage but the head sure knows it. I solved this problem by installing a driver transformer since I never did find a capacitor with zero leakage.

HIGH VOLTAGE DISCHARGE (CORONA):

Now where is this gremlin found in a low voltage device like a tape recorder? Anywhere static can build. If you've ever rewound a tape in winter onto ungrounded metal reels you'll hear and see the arcs as they jump to the chassis. Get your hand near the reel if you need to be convinced. This condition is responsible for random clicks, but one time I ran into a very mysterious *timed* click condition. My precious masters were acquiring permanent clicks at specific intervals and they were increasing with every play so it was imperative that I find this gremlin. The most meticulous methods of demagnetizing made no difference. The problem was the erase head. An erase head causing a problem in the play mode? Unbelievable? For sure, but upon analysis there was a spec of area between the ferrite core and the grounded outer shell which was arcing due to static built up on the core. Of course with every arc there was a pulse of magnetism that permanently recorded itself onto the tape. Did you ever wonder why we have gray hair?

I hope this was of some benefit.

Sincerely,
J.K. Hodgkins
Ch. Engineer

Dear Mr. Hodgkins,

Thanks for your comments about my tape-recorder magnetization paper, and for your list of magnetization sources. I hope that you don't get hit by a lightning bolt too often, but I agree that all of the others are real practical problems. They are all "mysterious" until you figure them out the first time. But most cause the magnetization to occur again right away, and they are easy to identify this next time.

The real "stinkers" are the ones (like your pulse in head switching) that are intermittent: Every test you do in the lab says "this machine has no magnetization problems." Then you put that machine into practical use, and it comes back a few weeks or a few months later with a magnetized head or guide. Why, oh why?

Yours truly,
John G. McKnight, President
Magnetic Reference Laboratory

Correction: In the July/August 1987 issue, Bruce Bartlett's article contained an incorrect address for the Magnetic Reference Laboratory. The address is:

Magnetic Reference Laboratory
229 Polaris Ave., Suite 4
Mountain View, CA 94043

Dear Editor,

Cheers to Drew Daniels and his article entitled, "Motion Picture Sound 1987: Dawn of a New Era" (September/October 1987, pp. 40-44). Wonderful coverage, wonderful writing, wonderful subject!

Amidst the misinformed, undereducated, pseudo-technical people who permeate the world of professional audio, it is indeed a pleasure to read an informed, objective, intelligent opinion. The snake oil peddlers masquerading as audio experts need exposing from time to time. The only industry rivaling audio for marketing hyperbole ambiguities is wine!

Sincerely,
Dennis A. Bohn
V.P. Research and Development
Rane Corporation

Dear Editor,

Late in 1986 or early this year db Magazine ran a most informative article on vented speaker enclosures. I valued my copy so much I put it away in a safe place for future reference. So safe a place that no one has been able to locate it since. Can you identify the volume and tell me how I might obtain another copy.

Thank you very much for a valued publication that I have enjoyed for many years.

Yours truly,
Ed Breland
Radio Station WBSJ

Dear Mr. Breland,

The issue in question was July/August 1986, and the article, "Vented Loudspeaker Enclosure Design Made Easy" was written by Drew Daniels. You should have received your copy by now.

Thank you for your kind letter.

Calendar

• COMMTEX International '88 is an exposition for communications equipment, software and accessories for professional dealers and users. It will be held on January 16-18, 1988, at the New Orleans Convention Center in New Orleans, Louisiana, and is co-sponsored by International Communications Industries Association and the Association for Educational Communications and Technology. Video, computer, audio-visual and other communications equipment will be featured. Activities held in conjunction with COMMTEX include ICIA- and AECT-sponsored conferences and seminars designed to help participants develop sales and business management skills and provide technological updates on communications product developments. Additional COMMTEX exhibit information is available from Kay Hynson at the International Communications Industries Association, (703) 273-7200.

• The upcoming schedule for the SYNERGETIC AUDIO CONCEPTS two-day audio engineering seminars is as follows:

• Anaheim, CA- January 27-28

• Seattle, WA- February 17-18

• Orlando, FL- March 9-10

• "Master Loudspeaker Designer's Workshop," conducted by Dr. Eugene Patronis and staff, will be held in Atlanta, GA on March 17-19.

For more information contact:
Synergetic Audio Concepts
PO Box 1239
Bedford, IN 47421

• The United States Institute for Theatre Technology, Inc. (USITT) is presenting the annual conference and Stage Expo '88 commercial exhibit show. The event is being held at the Disneyland Hotel in Anaheim, California from March 23-26, 1988. Stage Expo attracts suppliers and manufacturers of those products needed for live performances.

For further information contact:
RJA Exposition Management
486 Fullerton Ct.
San Jose, CA 95111
(408) 225-6736

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HOW TO READ THE LABEL

The top line of your label contains information that enables us to find you on the computer. Without that we can't effect address changes or find your entry. The sample below is what most labels look like.

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1120 Old Country Rd
Plainview, NY 11803

The first sequence of characters is how we locate you. The next four numbers indicate the issue you have just received. Finally, the last four numbers represent the date of expiry *reversed*. Just read backwards: 9309 is Sept/Oct 1993.

Microphones

In the studio, over the air or up on stage, there's a Fostex RP mic specifically designed for the job at hand. RP stands for regulated phase, a transducer technology which has been awarded over 20 international patents to date. These mics have the warmth of condensers, the ruggedness of dynamics and a sound as transparent as it gets.

Headphones

These are more outstanding examples of RP Technology. Model T-20 has become almost legendary among studio musicians, producers and engineers. Its flat response at any listening level and its comfortable design help you listen longer without fatigue. And the sound is so clear and well-defined, critical listening is enjoyable.

Speaker Systems

You're up & going with Fostex PA systems. Modular designs let you control the sound according to the needs of the event. Stack them, gang them. From a simple portable PA to an entire rig, look to Fostex speaker systems to help you solve your sound problems.

Powered Mixers

Model MP-800 has 8 inputs and delivers 180 W per channel and Model MP-1200 has 12 inputs and delivers 250 W per channel. These rugged, road worthy stereo mixers have quiet running fans, digital echo, normalled connections at all critical patch points, stereo graphics on the mains and super monitoring flexibility. The best at any price.

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MARK E. BATTERSBY

PROFITING FROM BANK SERVICES

• The commercial bank is usually the first, longest lasting and most intimate capital supplier to any recording or sound operation. Unfortunately, few sound or recording engineers think of their banks as anything but a capital supplier or processor of checks, but there is much more to banking than simply checks and loans.

We are all aware that our bank offers checking and savings accounts. Thanks to this service it is possible to deposit the checks received from our efforts and have the amounts credited to our account. Similarly, we can write our own checks using the money in that account to pay our own bills. Those checks will generally be accepted by anyone regardless of where they conduct their banking business.

The bank may take forever to process those deposited checks and to credit the money to your account (at least until Congress eventually clamps down on this "float"); the whole process is relatively painless—unless, of course, one of those checks "bounces." A bounced check or one that is not honored for any reason may mean that the sound or recording studio should have utilized another service offered by most commercial banks—credit checks.

Yes, your friendly banker will usually perform a credit check that will enable the studio owner or operator to determine just how much credit to extend a customer, how reliable a supplier may be or even how financially solvent a job applicant may be. What's more, the ever-neutral banker will perform these credit checks or supply credit information on any firm or individual—customer of that bank or not.

To help make the decision of whether to add extra employees, that same bank can also provide local or regional economic data that is usually quite reliable because of the bank's unique position as a financial confidante and intermediary for a variety of local businesses. Just imagine utilizing this resource in the merger and acquisition field. Banking services usually encompass a merger and acquisitions specialist or department which can do anything from finding a suitable candidate for a merger or acquisition up to actually negotiating a mutually-satisfactory deal.

The majority of commercial banks routinely offers custodial accounts for the recording operation's negotiable instruments. And, surprisingly, that same bank can also act as a broker or dealer for the short-term investments of the studio—despite Federal laws that prevent commercial banks from entering the long-term capital markets.

Among the most common services a bank can offer, however, is the use of its money in the form of short-term capital. Of course, to most recording enterprises short-term capital simply means a loan. But it can also take other forms such as leasing, credit card or merchant charge plans, the factoring of accounts receivable, etc., all of which help free the studio's capital and all of which are usually offered by your friendly neighborhood commercial bank. And, don't overlook the largest short-term capital need of all—the payroll.

Banks have the computer resources, the skills and the money to take care of the sound or recording operation's payroll accounting. Depending upon the relationship, in fact, the same bank may advance the payroll funds needed to pay the employees and even file the necessary tax and information forms and reports.

Banks also offer a variety of other accounting services as well. Demand deposit accounting, installment loan accounting (and collection), mortgage loan servicing and accounting, etc.

A bank that offers revolving credit loans or term loans could be said to be entering the intermediate-term financial arena, closer to long-term financing. That is the trend today as banks press up against the Congressional limit on long-term financing and securities dealing. Banks continue to develop more and more services that utilize their computer and personnel resources and all of which are designed to attract more and more customers to that bank.

This volume of services obviously benefits the small or medium-sized sound or recording operation, or at least those who are aware of the amazing number of services that a bank is willing to perform. But banking benefits don't stop with the studio operation. Owners and key employees can also profit from the banking relationship.

Banks routinely offer preferential interest rates to the owners and key employees of its business customers. Plus, trust services, retirement plans, financial planning or even a free safe deposit box, all make desirable fringe benefits for attracting and retaining key employees—as well as for the operation's principal or owner.

Just as the so-called "prime rate" is merely a guide to interest rate levels and usually negotiable, the many

The first 500-Hz Driver that doesn't turn cymbals into trash can lids

Listen to most of today's HF drivers, including our leading competitor's, and you could logically conclude that "trashy" sound is an inescapable fact of life. Poor definition, inadequate output beyond 10 kHz, annoying breakups, and "ringing" are all too common.

EV engineers, rejecting the notion that poor high frequency sound is inevitable, created the DH1A, a driver that deals effectively with every one of these problems.

To boost high-frequency output we utilized a magnet with the greatest flux density available, plus an optimized, balanced magnetic circuit to "stiffen" the coupling between the amplifier and the diaphragm. The resulting increase in high-end response also solved the problem of definition and articulation, so the sound is cleaner and livelier, with better transient-handling capability. As a result, trashy instrumental and vocal sounds are consigned to the trash can, where they belong.

The 10 kHz breakup you've heard in our competitor's driver was eliminated by using a 3-inch diaphragm instead of the other guy's 4-inch component, moving the primary diaphragm breakup point all the way out to 16 kHz, well beyond fundamentals and first harmonics.

A field-replaceable diaphragm, we reasoned, could make the DH1A still more useful. So we made that a part of the package, too. Plus the option of 8- and 16-ohm impedance match. And our EV-exclusive PROTEF™ feature that guards against voice coil damage.

Talk, as they say, is cheap. So, we insist that you make us prove our claims. Audition a DH1A today and hear for yourself how easily you can bid a hasty goodby to trash-can cymbals and high-end distortion.

For more information, write Electro-Voice, Inc., 600 Cecil Street, Buchanan, MI 49107.

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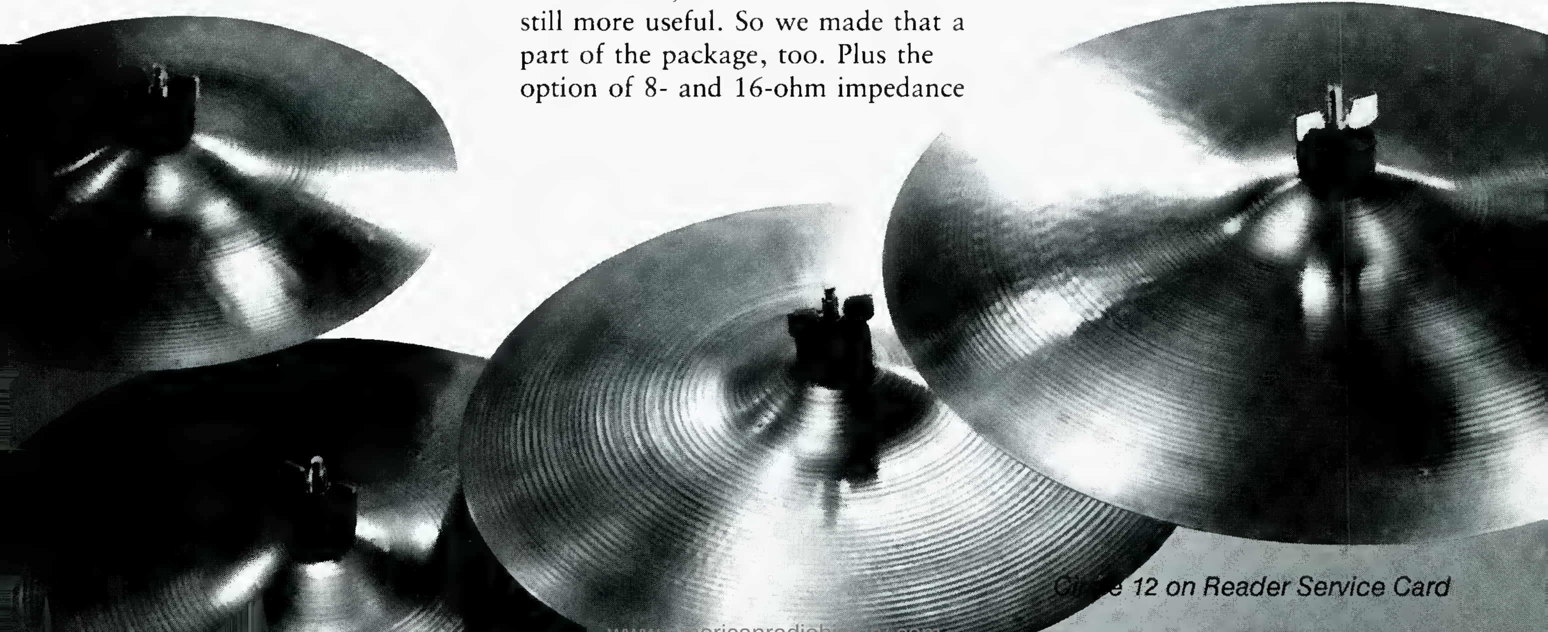

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services offered by the average bank are also offered by another bank just down the street. In other words, there is a competition which exists between banks all attempting to get your business. While it might help to be the largest recording studio in the area, most banks are more interested in the amount of money they can realize by doing business with a given sound or recording operation than the actual size of the operation.

Obviously, the more services which a bank performs for the studio operation, the more the bank will value the operation as a customer. Naturally, the better the customer, the less the bank will charge for its various services. And, no, those bank services are never free despite the bank's advertising. Fortunately, the amount paid for everything from a checking account to a loan is negotiable, if not at your present bank then at the next one you approach.

When it comes to paying for those banking services, there are two widely differing schools of thought. Some experts believe that a studio owner should know the exact amount being charged for each service utilized and pay accordingly. Other experts believe in trade-offs best illustrated by the everyday checking account.

Some banks charge a flat ten cents per check written, seven cents for every check processed, twenty-five cents per deposit and, perhaps, a bookkeeping fee of ten dollars per month. Other banks loudly proclaim that they charge no fee in return for the customer maintaining a minimum balance in the account at all times. Of course, with both methods, the bank usually fails to mention the profits it earns on the funds that it received for cleared checks but which isn't immediately available for the studio to use. This float is a significant amount as recent Congressional hearings have revealed.

While the float (the money which the bank earns on funds in its control but not used or available to the customer) is significant, the average sound and recording engineer or studio would be wiser to concentrate on how much the banking services actually cost it out-of-pocket.

For instance, if your bank offers free business checking in return for a minimum $1,000 account balance, the owner must compute how much the operation could realize if that $1,000 were invested. This, then, is what that "free" checking actually costs. Similarly, low interest rates offered in exchange for minimum balances requires this same computation.

To compare the cost of these "free" checking accounts with the cost of one which imposes flat fees for each deposit, each check drawn, etc., the average volume must be computed. Thus, weekly deposits, ten checks written each week, etc. costs x dollars opposed to the interest lost on the funds required to maintain the minimum balance required for a "free" checking account at a competing bank.

This bank comparison applies to every service offered by banks. What does each service actually cost in either hidden or up-front fees? The comparison not only tells the owner the actual price of his or her banking, but it also provides invaluable ammunition for negotiating with those banks which are competing for your business.

Bank A might offer an interest rate of X percent to anyone who applies. Customers might be offered an interest rate of X-1 percent. A customer who also uses—and pays for—the bank's payroll accounting service might be able to ask for an interest rate of X-2 pecent.

All banking services are negotiable. If your present bank won't discuss its rates, the one down the street will. If the studio is paying too much for banking services, it can usually be attributed to ignorance. Few of us actually know what we are paying for bank services—particularly in those hidden costs.

Shopping or negotiating for needed banking services has several other benefits. First, it helps demonstrate just how many services commercial banks actually offer. It also brings the owner, principal or manager closer to the operation's bank. And that banker is the key to a profitable banking relationship.

Banks offer many services that can benefit the studio. Finding the right bank for a particular operation is a matter of shopping. And benefiting or even profiting from those banking services means negotiating—both prices and number of services utilized. db

We challenge you to hear the $775 difference!

There's nothing better than a $1,000-plus condenser microphone to capture every performance detail. If you can afford it. But what if you can't?

Listen to some very talented musicians and mixers who recently tested the new ATM33R condenser cardioid from Audio-Technica. They told us the sound was almost identical to their big-bucks favorites. They liked the wide dynamic range and uniform off-axis response. The ability to use any standard phantom-power source from 9V to 52V, and the famed *Road Tough* construction were also definite plusses.

After comparing the ATM33R, several testers suggested they could now duplicate their studio sound on the road, where studio condensers were too expensive to risk. Others could see the advantage of four or more ATM33R microphones in a demo studio, at no more investment than one expensive condenser.

Compare the new ATM33R with any other condenser cardioid on the market. At ANY price. Check it for sound quality, ruggedness, and affordability. Whether you are MIDI sampling, cutting demos, or on stage every night, the ATM33R can make a big difference...for far less!

ATM33R Condenser Cardioid Microphone

audio-technica.

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Ad Ventures

BRIAN BATTLES

• Now back to our regularly scheduled column, already in progress (remember, last issue we took a break in our signal processing series to go over the the format mechanics of putting radio commercials on tape ["Ad Ventures," **db**, September/October 1987]).

REVERB

Almost every respectable recording studio has some tool for adding synthetic reverberation to a signal. In the production of radio commercials, however, reverb is hardly ever used, except as a special effect. Elderly readers (over 30) may recall the days of the old Top 40 AM stations that used to put reverb on all their jocks and announcers. (A handful of stations still do it.) This practice was used because in amplitude modulated broadcasting, the volume of the audio signal actually improves the radio signal strength a bit. Consequently, all other factors being equal, one can hear a louder station somewhat farther away than a quiet one. Higher volume also suppresses a certain amount of AM's inherent noise, and makes it sound as if the listener is getting better reception. (It's considerably easier for a station to compress its levels than to get FCC approval for a more powerful transmitter or a bigger tower.) In order to capitalize on this phenomenon, AM stations feed their audio through banks of compression devices and automatic gain controls that keep the average overall level up as high as possible without too much distortion.

As you may have guessed, extreme compression has some undesirable consequences. For example, every time the sound drops off below a preset bottom limit on the compressor's threshold, even in the milliseconds between spoken words, the compressor's ultra-short release time causes it to struggle to crank up the volume to fill the gap. With no proper signal available, the result is a bizarre and annoying effect commonly called *pumping*, a syncopated, frantic mixture of voice, breathing, hiss, and background sounds, as the compressor jacks up the low-level garbage in between the announcer's words. Since reverberation lengthens the time it takes for sound to decay, it can be used to fill the dead spots. The idea was that the reverb would keep the compressors from releasing too rapidly and therefore minimize the "pumping" sound. With the development of quieter audio equipment, better transmitters, and shifting listener tastes, the use of reverb has largely fallen out of favor. It has never been a consideration for FM broadcasters, since frequency modulated radio signals are virtually unaffected by the audio level.

EQUALIZATION

Most recording studios have access to equalizers, too. Most audio engineers are quick to point out that EQ can be tremendously helpful or it can be your worst enemy. Don't fool around with equalization unless you really need it. You might need to touch up an instrument or voice slightly, or dampen a bit of unwanted loudness in a particular frequency range, but if you have set up your microphones in the best configuration to make a crisp, clean original recording, you shouldn't need to touch those tricky EQ knobs. When mixing your commercial it is tempting to thicken up an announcer with a few extra dB at around 100-200 Hz or to brighten the cymbals at 10 kHz, but don't give in. I've told you before that stations run their audio through all kinds of processing gadgets, and a limiter might see the "hump" you've created at 150 Hz as an unwelcome overload, causing it to reduce the overall volume of your spot. Even if that doesn't happen, you'll probably wind up with an ad that sounds muddy or tinny compared with the other material the station airs.

I have found good uses for equalizers in creating special effects. Cutting out the top and bottom ends of a voice can produce a close approximation of a telephone caller, public address announcer, old-time radio broadcast, or police dispatcher. Use your imagination. Just remember that for your commercials you should almost always use equalization to reduce the level, rather than increase it. As a matter of fact, you might try an old trick: roll off everything below 75 Hz and above 15 kHz. I mean at least 10-12 dB per octave. Not only do you stifle superfluous audio debris, but you may find that broadcast equipment will actually boost your spot's volume a few tenths of a bel.

NOISE REDUCTION

What's the easiest and cheapest form of noise reduction for tape machines? Regular head cleaning and demagnetizing (degaussing). Manufacturers of studio tape recorders should somehow wire them such that the power switch is inoperable unless the heads are scrubbed and demagged first. Of course, knowing that this procedure is obligatory automatically makes doing it a pain in the hindquarters, but it's really not that big a deal. Get a pack of those long-stemmed cotton swabs (usually cheapest to buy if you order a box of a thousand or so from a medical supplies dealer). Then go to a pharmacy and ask for a bottle of real alcohol. It should be at least 91 to 98 percent pure--not the wimpy denatured rubbing alcohol in the health and beauty aids department at K-Mart. I've heard tell of folks using Graves or Smirnoff in an emergency, but my machines cost money, so I'm not experimenting. I suppose since it's also made out of your basic C_2H_5OH it's the same stuff, but consult your local chemist. (My hunch is that the flavorings and other impurities in booze could damage the heads, or at least leave residue. Why not save it for after the session? Otherwise it's a waste of good likker!)

Now, humming the theme from *Rocky*, dunk them swabs in the ethyl and swish 'em 'round. Two or three should be plenty. Massage the tape contact surfaces of the heads, then scour the capstan, guides, pins, and other miscellaneous widgets in the tape path. Finally, get a couple of dry swabs and polish up the places where you just applied the goofy juice. You see, the alcohol primarily loosens up the slime and goo, and your elbow grease cleans it off. If the dry swabs turn black or deep brown, you might want to run through the routine again. If this happens daily, think about purchasing a better grade of tape, you miser.

Don't let the stuff dribble down inside the mechanism (it rinses oil and grease off the parts that need to stay lubricated), and don't use alkeehawl on the rubber pinch roller. You can buy a rubber-conditioning cleaning fluid from your neighborhood pro audio dealer, or from many tape deck manufacturers. You should keep the pinch roller clean and pliable, but not gummy or sticky. If it starts to look out of round or takes on a strange color, replace it.

Shop around for a good degausser (ask to see the specs; choose one that puts out a healthy magnetic field). To use it safely, first clear the area of stuff like recorded tapes and computer diskettes within a six to ten foot radius. Make sure the tape recorder's power is off before you plug in the demagnetizer. Bring it to within a few 64ths of an inch of the erase head, slowly wave it around in front of the surface, then gradually draw it to a couple of feet away. Do the record head next and hit the reproduce head last. Before you unplug it, use the gizmo on your razor blades, steel tools, and any other objects that contact the tape. (Note: if you wear a Rolex, you might want to slip it off before you start. You never know what can happen when you're messing around in the invisible world of artificial electromagnetic force fields. If it really bothers you, maybe you should write to Mr. Wizard.)

Say, while we're off on this wacky tangent, note that you should never place recorded media (reels, cassettes, computer disks, etc.) near loudspeakers, utility company meters, electric motors, TV sets, metal detectors, radios, hydroelectric dams, AC cables, police radar guns etc. Back to noise reduction. Use it. Dolby, dbx, whatever. (I prefer dbx Type I.) But, hold on: never use noise reduction on the copies of the tapes you'll be sending out to stations. Most of them don't have the equipment to decode it properly. If you've ever listened to a non-decoded dbx-processed recording, you'll quickly understand the problem. Some people say you can use noise reduction if you check with the station first, but I wouldn't trust a stranger's ability to operate fancy outboard processors to decode my material accurately. If you've made a commercial that's relatively hiss-free from the start, one generation shouldn't add appreciable noise. As I said last time, just duplicate the stations' copies in real time on good quarter-inch tape at 7-1/2 in./sec. with the level up nice and high, and use as close to the full width of the tape as possible. Ideally, monaural recordings should go full track, stereo on half-track. Store and ship the reels tails out, and use plenty of leader at both ends.

Noise gates are valuable in avoiding buildup of clutter in a multi-track mix, and some can even be used on a plain voice recording, provided they do not act in an obvious or abrupt manner. With a reasonably quiet background a noise gate will entirely block the signal path during silences between words or notes much like a squelch control on a CB radio or police scanner defeats the speaker when there are no transmissions coming in above a set lower limit. Gating is all but compulsory in a number of recording situations.

STAR WARS

It's difficult to discuss today's high tech electronic devices built by Martians and designed to Excite, Harmonize, de-sibilate, fibrillate, twist, shift, mutate, electro-burp, phase-align, and otherwise adulterate innocent audio signals. First, they're usually one of a kind inventions, so there aren't many different brand names in any one category. Second, the beings that dream them up are inevitably secretive about just exactly what they do and how they do it. Or maybe even the manufacturers aren't quite sure. The point is, some of these units are kind of fun to have around, and a few even have practical uses. I'll pick a few at random:

The Eventide Harmonizer Model H969 is certainly an impressive thingamabob. Not only does it sport plenty of *2001*-type readouts and lights on an ominous black faceplate, it also has a wide range of features. There's adjustable digital delay (up to over 3 seconds), flanging, Doppler effects, electronic repeat, and the notorious pitch control section for which it is famous. With the latter you can shrink a 64-second ad into a :60 or stretch a 58-second spot to the same standard length, correct tuning variations up or down a full octave, and create unlimited contortions that would be the envy of Max Headroom. Not everybody needs one --at nearly five grand not everybody can afford one--but it can sure be handy in certain applications.

Barcus-Berry Electronics, a company once known for its instrument pickups, now brings us the BBE 802 (and other models). This little piece of hardware employs some method of frequency-sensitive delay and phase alignment yielding recordings that seem brighter and louder without increased noise or volume. Forget using it on an already hissy recording of solo instrument or voice because it tends to impart a noticeable yo-yo sound on the background noise. Yet it can give a muddy old cassette a pleasant sense of apparent clarity, and it helps midrange and upper-frequency material to bite through a mix. With no signal present it idles as quietly as a mausoleum. The earthlings employed by the alien cyborgs who secretly own BBE are delightful to deal with, too. The BBE is a good deal for around five hundred bucks.

The acclaimed Aphex Aural Exciter is familiar to most of us by now. Obviously created at the expense of the inventor's selling his soul to Satan, this audio processor somehow synthesizes peculiar harmonics from its input source and manages to impart an indefinable feeling of crisper sound without the harshness you might expect. Like the BBE, it only works when there's something for it to work on, so there aren't any spurious chirps or wheezes during silent passages. It's worth looking at for a couple hundred dollars.

A lot of firms make things called sibilance controllers, or de-essers, mystical boxes containing circuits or incubi that in some fashion are able to sift out swishiness from voices and certain instruments. The better ones have little effect on non-stridulant sounds.

There are, of course, dozens of other contraptions available to suit a variety of particular applications. Recording technology is at the point where one can find equipment to deal with virtually any conceivable situation. The status of recording science and technology scarcely imposes limitations; only your finances can regulate your ability to realize your creative visions.

OVERDUBS

In a previous column I stated that the current copyright laws and publishers' policies are such that a broadcast station could purchase blanket rights to the use of prerecorded material by paying annual fees to the appropriate sources (e.g., ASCAP, BMI, SESAC, etc.), and that this meant that a local radio station can incorporate portions of commercially-released recordings as background music in advertisements. Apparently, this is not the case, according to a letter I received from Mr. Eric B. Culp, Engineering Manager at WOWO/WIOE in Fort Wayne, Indiana. Mr. Culp refers to the September 17, 1987 issue of *Broadcasting And The Law* (L & S Publishing, Inc.) which clearly explains that stations' licensing agreements do not permit use of recorded works for inclusion in commercial or promotional advertisements. Separate permissions or "synchronization rights" must be negotiated directly from the music publisher for each song.

Yes, it is an utterly inane legal situation. After all, most radio stations have been using tracks from record albums as production music beds on many of their own commercials for ages, and it's clearly infeasible to enforce this rule. The most ludicrous aspect is that stations can acquiesce by purchasing a generic music library from one of dozens of production houses. About all this does for recording artists is deny them any additional publicity they might receive from having their tunes played on the air in commercials. For some performers this is the only means of achieving exposure on commercial stations. The music publishers continue to follow this preposterous and avaricious line of thinking begun in the 1920s when they first tried to have recordings of published material banned from the airwaves with the ignorant notion that it would diminish the sale of records.

Eventually, the pressure they applied permitted them to compel stations to pay royalties for the privilege of using records. Of course, it quickly became plain that without airplay, most records were doomed to fail. (Soon, record companies established wealthy promotion departments whose chief mission is to induce program directors to play their records.)

No one disputes the right of artists and composers to profit from their efforts, but instead of recognizing the detrimental effect of their misguided regulations, ASCAP, BMI, SESAC, et al assume that the few dollars they might extract from a handful of latest album by Norman and the No-Hits. They spend $100,000 on trade magazine ads, radio spots, billboards, and contests, then dispatch Jane Q. Promoter to WCHEAP with an armload of free No-Hits lp's, concert tickets, T-shirts, keychains, visors, baseball jackets, and posters. That afternoon John Q. Deejay obtains a purchase order to use *Waste O' Vinyl Jam* by the No-Hits in an ad for a health club or stereo shop. Right.

Ridiculous as it sounds, that's what the publishers expect, so that's the rule. I'd say that's great news for us jingle composers.

TALKBACK MIC

Thanks to Eric B. Culp of WOWO/WIOE for the helpful letter and info...Don Townsend of Colorado Media, I look forward to a mutually beneficial business relationship...Dennis Constantine and John Bradley of KBCO, Boulder, Colorado, sincere thanks for your consideration (better luck next time, right?)...I received a superb demo cassette from Mario Fencetta of NuAge Studios in Auburn, Maine. You're on the right track!...Congratulations to Steve and Robin Kiely of Vernon, Connecticut on the birth of their second son, Shane...Terrific ideas on a tape by Kyle Underwood of Second City Tape Factory of Schaumburg, Illinois...Ken and Diana Gershon, after whipping up the funniest male/female character spots I've heard since Nichols and May, have finally opened Gulf Farm Advertising as a spare-time enterprise in Panama City, Florida...whoever sent a black cassette with four sixty-second jingles (generic record shop, car dealer, arcade, and restaurant) postmarked High Point, North Carolina, you forgot to enclose a note or return address!...personal note: I'm now a full-fledged member of the International Television Association. I expect to make some valuable contacts for voice-overs and audio consulting. The ITVA people I've met so far are wonderful.

POSTSCRIPT

Don't let Washington ban part of the music scale-- oppose that dreadful anti-copying bill. (For readers who have been stranded on an island for the past few months, let's have a general summary of the crisis.) There is a new home audio taping format about to come to market called DAT (Digital Audio Tape). It's a cousin to conventional cassette recording, except that the sound is recorded digitally. This means that theoretically one could dub a Compact Disc onto a DAT cassette and reproduce a virtually identical copy. Sounds terrific, doesn't it? Like having a CD player that records as well as plays back.

Unfortunately, some moronic record companies disagree. They are laboring under the delusion that most of us are going to stop buying authorized recordings and start an enormous bootlegging industry that will put them out of business. To forestall this catastrophe, they're urging the United States Government to pass a law making it literally impossible to copy a CD onto DAT. (Funny, I recall hearing them whine about the same thing when analog cassettes came out. Too bad they couldn't get cassettes outlawed; surely the revenues from cassette sales really throw them into a panic at tax time.)

This news is not to be taken lightly, because they believe they've found the ideal solution. CBS has invented some insane system which requires the original master recordings to have a notch or gap 250 Hz wide so that a mandatory decoder inside every home DAT deck would refuse to record from one of the punctured originals. Okay, so there'll be no CD to DAT copying at home, so what? As long as they put the notch up around 20 kHz where nobody will miss it, right? Wrong! It will be around 3 kHz or so, right smack in the middle of the audio spectrum! The record industry and CBS imbeciles are trying to convince the lawmakers that consumers won't notice it. (These are people whose primary mode of communication is evidently AMESLAN.) If you think that's bad news, keep in mind that the chunk of music cut out would also be eradicated from all versions of the recording, including regular lp's and cassettes.

I hope this plan infuriates you, because there's only one way to prevent waking up one morning and finding out that this debacle has become law. It's simple. All you have to do is pick up your phone, call your local town hall or city clerk, and ask for the name and address of your representative in Congress. Then grab a piece of paper or a postcard and dash off a note stating that you are vehemently against this proposal (known in Congress as HR-1384). If you're really upset, you'll also write your Senator and say that you vigorously oppose passage of the bill. (The Senate designates it S-506.) Let's stop this outrage while we still can!

In order to receive further information on this subject, there is an organization to be contacted. The Home Recording Rights Coalition can be reached at 1-800-282-TAPE. They can provide an abundance of information, including names of Congressmen and Senators. db

Recording Techniques

BRUCE BARTLETT

A SHORT COURSE IN MICROPHONE TRAINING

• Here's a systematic series of experiments that can train you to achieve various sounds through recording techniques. We'll start with very simple methods and work up to complex ones. This route is the same as that taken by many professional engineers in the course of their careers.

RECORDING A SOLOIST WITH ONE MICROPHONE

First, place a microphone 1 foot from a musical instrument or voice and plug the mic into a tape recorder (as in *Figure 1*). Plug in headphones to monitor the recording and playback.

Although this set-up is simple, there's a lot to be learned from various experiments:

1. Make recordings at -20 VU, 0 VU, and pinning the meter. You'll hear how excessive levels cause distortion and too-low levels make tape hiss audible. Note that percussive instruments require lower recording levels for undistorted recordings than do non-percussive instruments.

2. Keeping the recording level at 0 VU, make recordings at 6 inches, 1 foot, 2 feet, 4 feet, and 10 feet. You'll hear how mic'ing distance affects the amount of room reverberation heard in the recording. Distant placement sounds distant; close placement sounds close.

3. At a mic'ing distance of 1 foot, make recordings with several different microphones (one at a time) at the same recording level. Listen to the tonal differences of various microphones. Write down what you heard for future reference. Repeat this experiment with various musical instruments.

4. Obtain a single-D cardioid microphone (such as a Shure SM-57 or SM-58). Hold the mic 2 feet from your mouth. While maintaining a constant recording level, talk into the mic and move it slowly toward your mouth until it's touching your lips. You'll hear the bass increase as the mic gets closer—a phenomenon called proximity effect.

Repeat this experiment with a multiple-D cardioid mic (such as an Electro-Voice RE-18) or an omnidirectional mic. The tonal balance will stay relatively constant with mic'ing distance.

5. Record a loud instrument (such as kick drum, guitar amp, or screaming voice) 1 inch from the mic at -8 VU

Figure 1. The setup for one-mic recording experiments.

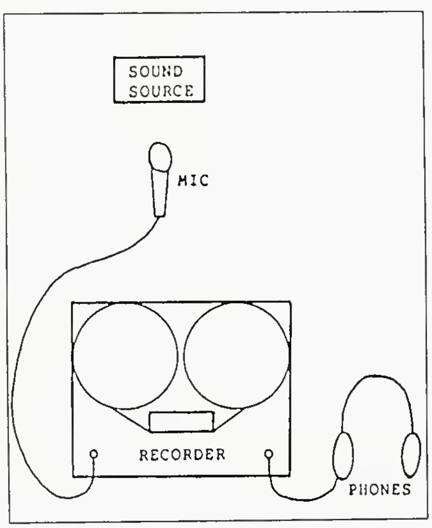
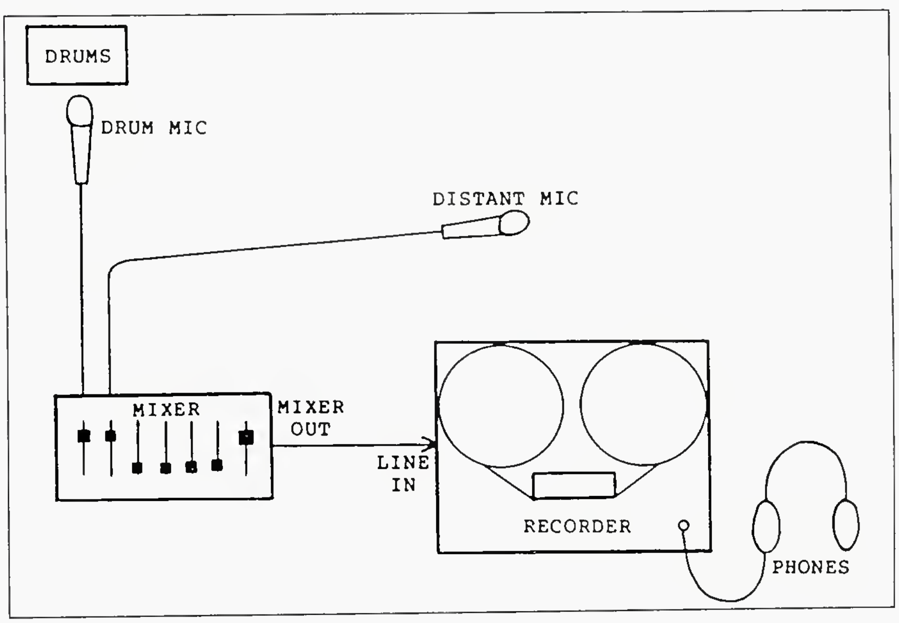


Figure 2. The setup for leakage experiments.



recording level. Play back the tape. Even though you recorded at a moderate level, you're likely to hear distortion caused by the microphone signal overloading the mic preamp in the tape recorder. Insert a pad in the microphone cable (such as a resistive voltage divider with 20 dB loss) and repeat the recording. The distortion should be eliminated.

6. Record the output of a cardioid microphone. Talk into the mic from all sides while maintaining a constant distance from it. Play back the tape. Your reproduced voice will be loudest in front of the mic (on-axis) and softest behind it. Note that off-axis rejection of sound is best in an open, non-reflective area. Off-axis rejection is poor when the front of the mic is aiming at a nearby wall, which reflects sound into the front.

7. Repeat Experiment 6 with an omnidirectional microphone. The overall level will stay constant when you walk around the mic, but the high frequencies ("ss" sounds) may become duller off-axis. You'll find that miniature microphones have very little change in sound off-axis.

8. Using one microphone, place the mic at various positions around the sound source, keeping a constant distance (say 1 foot) and record the result. You'll hear how microphone placement affects the tonal balance. Repeat this for various instruments and note the results. This is one way to become more proficient in microphone techniques.

9. Record an electric guitar with a direct box connected to (1) the guitar, (2) the output of the effects boxes, and (3) the amplifier speaker jack. Compare these recordings to one made with a microphone near the amplifier/speaker.

RECOGNIZING PHASE CANCELLATIONS

Now let's place the microphone near a hard reflective surface and make a recording. While saying "Sally's sister" into a microphone placed 1 foot above a table top, slowly lower the microphone down to the surface. You'll hear a filtered, swishy tonal change called comb filtering. It's caused by phase interference between the direct and reflected waves arriving at the microphone.

For comparison, record with a boundary microphone (such as a Crown PZM) placed on the table. Phase cancellations are eliminated.

RECOGNIZING LEAKAGE

Let's move on to another experiment involving two microphones and a mono mixer. Plug the mics into the mixer, and connect the mixer output to your recorder line input (as in *Figure 2*). Place one microphone close to a snare drum, and place the other mic several feet away. The distant mic can simulate a microphone intended for another instrument, such as an acoustic guitar.

Set the mixer master volume control 3/4 up. Turn up the volume control of the drum microphone and make a recording at -8 VU. You'll hear a clean, tight drum sound.

Now while recording the drum, gradually turn up the distant mic and mix it with the drum mic. You'll hear the clean drum sound become distant and muddy as you bring up the distant mic. That's caused by leakage of the drum sound into the distant microphone.

Next, turn up the distant mic just enough to make the drum sound muddy. To reduce leakage into the distant microphone, (1) aim the "dead" rear of the distant cardioid mic at the drum, (2) place the distant mic in a padded, acoustically dead area, (3) place a large, heavy wooden panel between the drum and distant mic, or (4) try placing the distant mic at various distances from the drum. Record these experiments and note

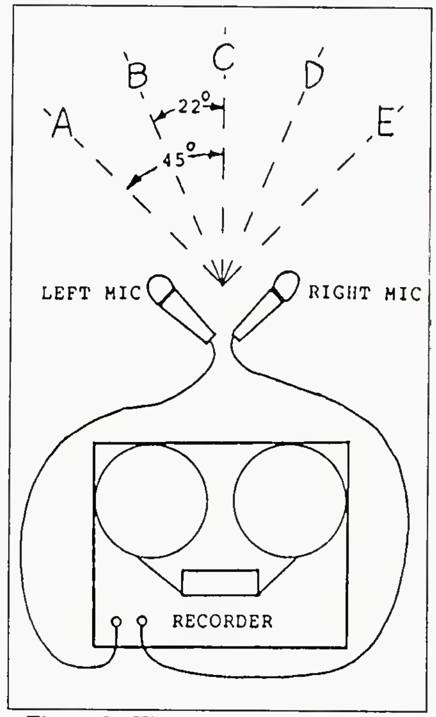
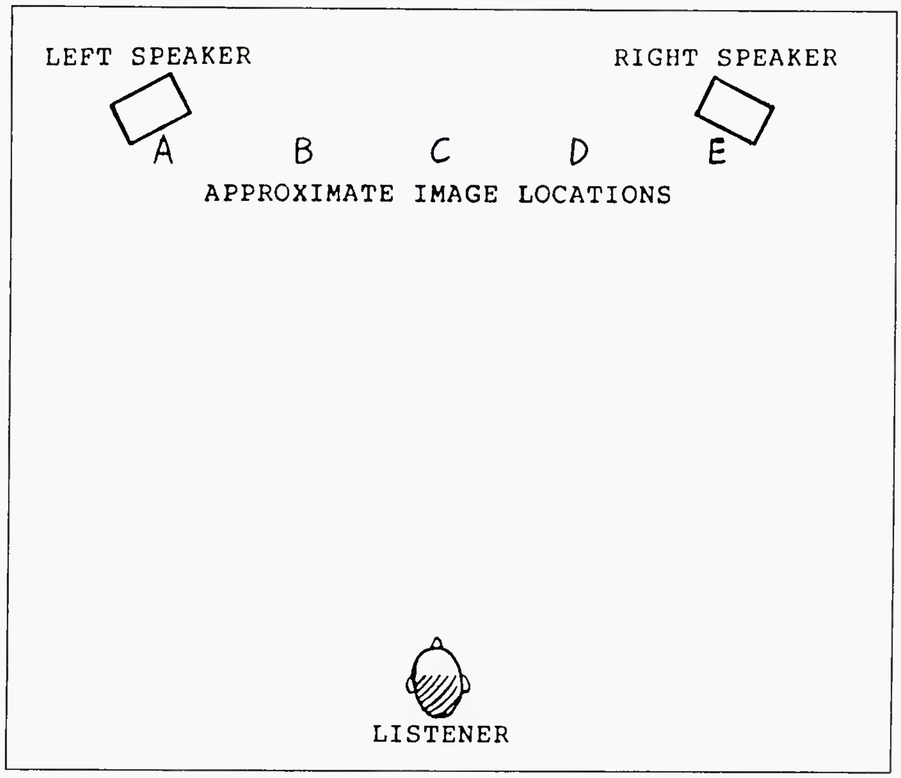


Figure 3. The recording setup for the stereo localization experiment.

Figure 4. The setup for the stereo localization experiment.



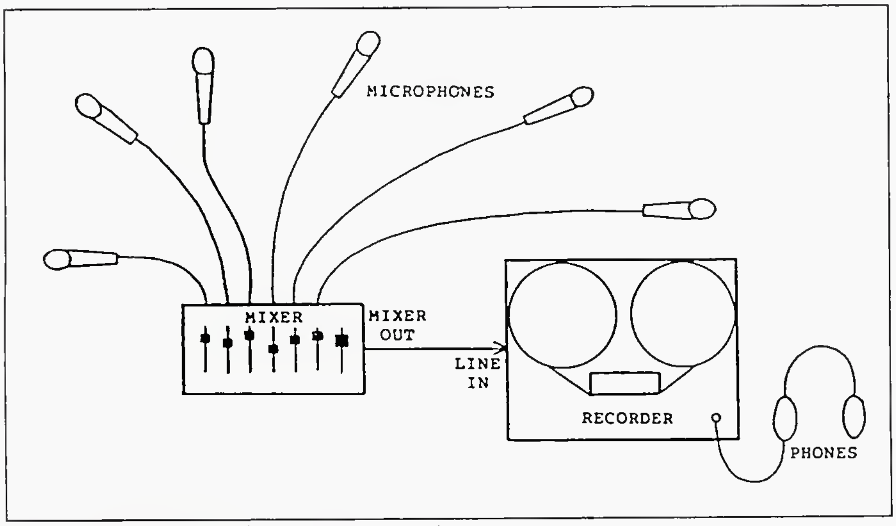


Figure 5. The setup for a live mono mix.

how the clarity of the drum sound changes each time.

HEARING STEREO EFFECTS

For this experiment, plug two identical cardioid microphones into a recorder and place them as shown in *Figure 3*. Record yourself speaking while you walk around the microphone pair at a constant distance. Say "I'm speaking in position A; I'm speaking in position B," and so on. Play back this stereo recording over two loudspeakers. Sit equidistant from the speakers, as far from them as they are apart, as in *Figure 4*. You'll hear yourself speaking from various positions between the speakers.

In making the recording, try these variations:

1. Angle the cardioid microphones 110 degrees apart (55 degrees to the left and right of center) and spaced 7 inches apart horizontally. You'll hear fairly accurate stereo localization and sharp imaging.

2. Try various anglings and spacings of the two cardioid mics, and record and play back the results. Try eliminating the spacing—put the grille of one mic directly over the grille of the other, and angle the mics apart. Note the results.

3. Place two omnidirectional mics 3 feet apart, aiming straight ahead, and record yourself as described above. In positions A or E, your voice should be reproduced from the left or right speaker. In position C, your voice should be reproduced midway between the two speakers (that is, straight ahead). In positions B or D, your voice should be approximately halfway off center, but without pinpoint imaging.

4. Repeat the above with two omni mics spaced 10 feet apart. Now you'll hear your voice in positions B and D coming from the left or right speakers—an exaggerated separation effect.

5. Repeat the above with two omni mics 6 inches apart. you'll hear very little stereo spread.

6. Listen to all the above recordings on headphones. Note the differences between headphone localization and speaker localization.

DOING A LIVE MONO MIX

Please refer to *Figure 5* for this next experiment. Plug several microphones into a mixer without equalization or effects. Use a mono mixer, or if your mixer is stereo, assign all the mics to one channel. Connect the mixer output to a recorder line input.

Place each microphone close to each instrument or voice in a position where your experiments yielded a natural sound.

Set the mixer master volume 3/4 up. Turn up the microphone volume controls. As the instruments are playing, adjust the pad or gain-trim control for each microphone input to prevent input-overload distortion. Many mixers have LEDs that flash when input levels are excessive; turn down the gain trim (increase the attenuation) just until the LEDs go out.

Do a live mono mix of the instruments and vocals, with the mixer meter and recorder meter both peaking around 0 VU. Is anything too loud or too quiet? Can you hear everything? Can you understand the lyrics? Adjust the volume controls for a pleasing balance, and record the mix. Play it back and evaluate it.

Try turning up the mixer level until the meter starts pinning, and reduce the tape deck's recording level back to 0 VU. Listen for the distortion caused by overloading the recorder input.

DOING A LIVE STEREO MIX

Let's complicate things a little. Use the same set-up as before, but with a stereo mixer. Assign instruments to channel 1 (left), channel 2 (right), or both. Use pan pots to position instruments in various locations. Normally the kick drum, bass, and lead vocal go to center, while keyboards and rhythm guitar are either in stereo or are split equally left and right.

Note the "unbalanced" effect of placing bass or drums in only one channel. Also note how it's easier to hear what similar instruments are playing if they are separated spatially.

Monitor the recording alternately in mono and stereo to see if the mix changes. Listen for center channel buildup: In mono, centered instruments and vocals sound louder relative to non-centered instruments than they do in stereo.

DOING A MULTI-TRACK RECORDING AND MIXDOWN

Now we're getting into professional techniques. Use a multi-channel mixing console and multi-track tape machine to record each instrument on a separate track, as shown in *Figure 6*. After doing the recording, plug the multi-track machine's outputs into the mixer's line inputs. Connect the mixer's channel 1 and 2 outputs to the line inputs of a 2-track recorder (or a cassette deck).

Play back the multi-track tape and mix the instruments with the mixer. Note how much easier it is to mix a tape than to mix live instruments. That's because

(1) it's easier to hear what you're doing with no live band playing, and

(2) you can play the multi-track tape over and over to practice the mix.

ADDING EQUALIZATION

Using the same set-up as before, add EQ to each instrument. Apply extreme boost or cut at various frequencies to hear the effect. Note that the same EQ has different effects on different instruments. For example, a boost at 12 kHz affects cymbals but not bass guitar.

Invent descriptive terms for these tonal changes, such as "bassy, warm, dull, crisp, present, edgy," etc. Then the next time someone asks you for a particular tonal balance on a certain instrument, you'll have an idea how to get it.

Listen to each instrument alone and set the EQ for the most natural tonal balance. Some instruments may need no EQ at all, depending on the microphone chosen and its placement. Then mix all the instruments together and note whether the EQ needs changing. Often, instruments mask or cover up each other's high frequencies, so extra EQ is needed when instruments are combined in a mix.

Compare recordings made with and without EQ. Ideally, the EQ'd recordings should sound much better tonally, unless your microphone techniques resulted in a great sound without EQ.

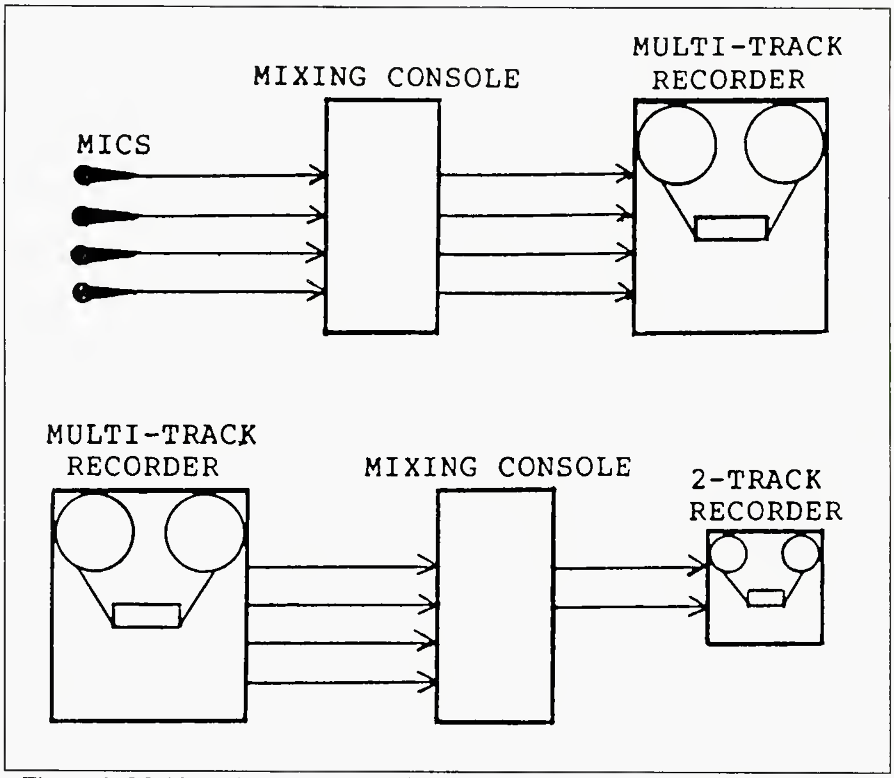


Figure 6. Multi-track recording (top) and mixdown to two track (bottom.)

ADDING REVERBERATION

Using the previous set-up, connect an external reverb unit between the echo send and receive jacks (or aux send and receive jacks) on your mixer. Next, set the echo-return level about halfway up. Now adjust the echo-send level on each instrument's channel for the desired amount of reverberation.

Usually, bass and kick drum get little or no reverb so that they retain their clarity. Note the effect of adding reverb to these instruments. Also try adding too much reverb to vocals and note the loss of clarity.

Compare mixes made with and without reverb. Those with reverb sound more spacious, as if they were recorded in a concert hall.

If possible, compare a mono reverb unit to a stereo one. Pan the mono reverb-return signal to center; pan the stereo reverb-return signals to left and right. Note the increase in spatial realism that stereo adds.

ADDING COMPRESSION

Since vocals have a wider dynamic range than the instrumental backup, vocals occasionally sound too loud or too quiet. Compressing the vocals keeps their level more constant relative to the instruments.

Plug a compressor between the vocal tape-track output and a console line input. Bring up the vocal volume control, play the tape, and listen to the compressed vocal track alone.

A typical threshold setting on the compressor is -10 dB; a typical compression ratio is 3:1. Starting with these settings, change them and listen to the effect. Too much compression sounds "squashed" or "forced."

Now mix in the rest of the instruments. The compression should be less audible, and the loudness of the vocals should be better controlled.

Listen to the effect of compression on kick drum, drums, bass, and lead guitar. Try to describe these effects in words so that you can communicate with others how compression sounds.

ADDING DELAY

Here's another signal processor to play with. Connect a digital delay to the echo send and receive jacks on your mixer. Set the mix control on the delay unit all the way to "delay." Play a drum track through the mixer, and turn up its echo send until you hear both the direct and delayed signals in equal proportions. Adjust the delay from 0 sec. to 1 sec., and note the effect of various delay settings.

Up to about 20 msec, you'll hear flanging or comb filtering. At about 50 to 100 msec, you'll hear a quick slap echo. Slow repetitions occur around .5 to 1 second.

Turn up the recirculation control until the echoes repeat. Add a repeating echo to vocals, and note the amount of recirculation needed for a tasteful result.

SUMMARY

To train your hearing, it helps to spend some time with each piece of recording equipment. Get to know the audible effects of different control settings and microphone placements. Compare the sound of various pieces of equipment. While doing this, record the results and announce on tape what each experiment was. Take notes on what you heard for future reference.

By trying these experiments, you can train yourself to hear things in recordings that you may not have noticed before. When you listen closely to commercial records, you'll start to perceive microphone techniques, mix balances, EQ, reverb, delay, and so on. You'll also learn how to achieve all sorts of sonic effects in your own recordings. db

How to put a

If you have some bright ideas about making music, we have a place you can develop them.

The new MT2X multitrack recorder/mixer.

With the MT2X, you can record separate parts on separate tracks, one part at a time. Then combine tracks to hear all the parts together as a complete musical work.

Track by track, you can put a whole band together. Try out new arrangements. Or experiment with different sound combinations.

It's like a notebook for your imagination.

One that comes with four recording tracks, two tape speeds, six input channels, and endless possibilities.

For example, you can "ping pong" or transfer the music from several recorded tracks onto just one. Which clears those tracks to record still more. In this way, you can easily create more than a dozen tracks (voice, instrument or special effects) from just one four-track recorder.

And the dbx® noise reduction means you'll hear more music and less tape hiss. It's the same system that professional studios use.

The MT2X's built-in mixer opens even more creative possibilities. With six inputs, you can mix up to six different parts simultaneously (for example, background vocals, rhythm section or horns) onto a single track. Leaving plenty of room on the remaining tracks to add lead vocal, instrumental solos and special effects.

band together.

And speaking of effects, you can also plug in digital effects units like our SPX90II. And add dramatic stereo echo, reverb, chorus, etc.

The MT2X is also designed to accept FSK sync from a drum machine, music computer or MIDI sequencer. And even if your sequencer can't do tape sync, our optional YMC2 MIDI FSK converter will get you on track.

Each channel has Hi and Lo EQ controls for shaping each instrument's tonal qualities, and LED meters indicate the optimum input level.

For the pro, an MT2X is perfect for taping rehearsals and making polished audition tapes.

But for anyone, it's a complete, affordable home studio. A creative garden where ideas can grow.

Your Yamaha Professional Audio dealer can tell you about zero stop, pitch control and the MT2X's many other professional features.

See him soon. And start putting it all together.

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Editorial

EDITORIAL AFTER 20 YEARS

This issue you hold is our 20th year celebration issue. In 1967, when db Magazine was started, there were no other publications for the audio professional—save, of course, the AES Journal. Since the Journal was, and largely remains, the theoretical forum for new ideas, we formed, even then, **db, The Sound Engineering Magazine,** to be the practical book for the industry. And so it was born, and so it is today.

Of course, the audio industry has changed much in those twenty years. But in re-reading our first two issues, which were on the then state-of-the-art in professional audio, it is interesting to note how many of those futuristic ideas written then, have happened, indeed are happening, even now.

When we were 15 years old, the age of digital recording was becoming standardized. Today, it is mostly tape recorded, but as our September/October cover story indicated, the recording studio with no tape recorders is becoming a reality. The emerging R-DAT technology will bring professional digital mastering to a new level of the industry.

We expect that before a year is out, we will have articles on sophisticated smaller—all digital—recording studios built around a digital console and a compact R-DAT recorder.

db, The Sound Engineering Magazine was first on the pro-audio scene. And we intend to remain the best.

Watch these pages as we go into our 21st year, for more practical articles.

If you are a professional in broadcast audio, post production, recording studio, or sophisticated sound contracting, **db** is where we can both face the future. L.Z.

JOHN BARILLA

Visual Music: Harbinger of the New Age in Audio

In the lower east side of Manhattan, in a recently renovated, but otherwise unassuming building, two audio visionaries have put together a formidable production facility. Lean, clean and computer-centered, it is apparent that Visual Music has chosen the high-tech road in its approach to hardware. This alone, however, does not make a studio particularly noteworthy. People do. It is therefore the farsighted blend of technologies, as well as the production philosophy of its owners that make Visual Music a harbinger of the new age in audio.

The principals of Visual Music are Jay Henry and Gene Perla. Henry, the more vocal member of the duo, exudes the essence of a confirmed futurist. Always anticipating trends and pushing available technology beyond the boundaries of known use, Henry is the engineering strength of the company. He has earned a number of gold records in his fifteen years of engineering including one for his work on Shannon's smash debut album, "Let the Music Play." More recently he completed an album for Rainy Davis (on C.B.S.) and another for Run DMC (on Profile Records).

The counterbalance in Visual Music is Gene Perla. More laid-back in character, he is something of a musician's musician and business entrepreneur. Classically trained on both keyboard and bass, Perla went on to perform and record with some of the jazz giants. He has worked with the likes of Chick Corea, Miles Davis, Sonny Rollins and Joni Mitchell, and was formerly a co-owner (with Jan Hammer) of Red Gate Studio. Currently, he runs two independent record labels specializing in the international marketplace. Both Perla and Henry are colorful characters whose personalities animate every session at Visual Music.

PHILOSOPHY

To create audio tracks that "never have to kiss tape"—this, according to Jay Henry, is the *raison d'etre* for Visual Music. Through the extensive use of virtual tracking (linked together by SMPTE/MIDI controllers), it is certainly within the realm of possibility. Even if a few of the tracks do have to "kiss tape," (as in the case of vocals), the pristine purity of the overall mix is still quite awesome. While the usage of virtual tracking is not new—(drum machine tracks are often cut while mixing)—the logistics of preserving the vast majority of a 48-track mix as virtual can be unwieldy. So until practical and cost effective systems are universally available, most producers will still find it easier to take refuge in multi-track tape, but in the analog realm, the small amounts of noise inherent even in top quality recorders are multiplied in the dub-down. In the case of television where a mix element may be two to four generations down before the layback to video tape, the situation becomes critical. Even with the commonly used noise reduction systems, some of the openness of the tracks is forever lost. The system at Visual Music addresses itself to this and several other production problems.

A Macintosh 512K computer is the central hub of the Visual Music system. All control signals (MIDI) converge at this point while recording, and diverge from this point during playback. Since the Mac is capable of true multitasking, it can be used for real-time sequencing, synthesizer voicing or musical notation without having to reload software for each application.

Performance information is loaded into the computer simply and elegantly from a Fairlight CMI. (While other world-class instruments (like the Synclavier) were not within budget for Visual Music, the owners both praise Fairlight for its extremely intuitive programming. Henry asserts that a new client can be accessing all its creative potential within a few hours of starting a project with his company.)

Figure 1. Gene Perla at the keyboard at Visual Music.

John Barilla will be contributing an article on the electronic cottage aspect of pro audio in future issues.

Figure 2. Jay Henry is seen at the Visual Music console.

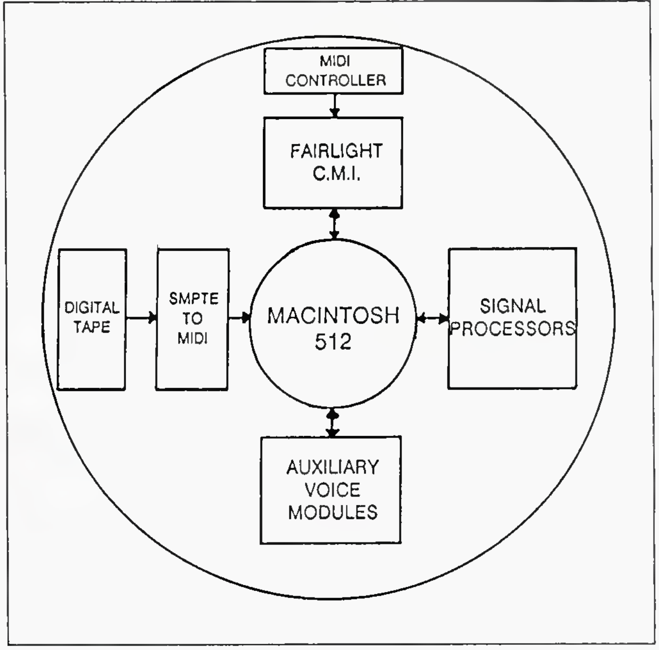


Figure 3. A Macintosh computer is at the hub of the system.

While the CMI is an incredibly versatile stand-alone instrument, it does have limitations in the timbres it can create. Sometimes a more common device will do the superlative job in the context of the need. Why spend a half an hour of the client's time programming a sound on the Fairlight that already exists as a stock sound in a Yamaha FB-01? Techno-pride is not an issue here. Saving the client time and money is, so Visual Music has a rack of outboard synthesizer and sampler modules as well as signal processors—all of which are controllable in real-time from the Fairlight keyboard. Since virtually all the gear in the system is MIDI-controllable, all performance data—even "on-the-fly" variations in reverb parameters—can be stored and replayed from the Mac.

A GENERAL SESSION

Although there is no such thing as a "typical" recording project, I asked Henry and Perla to sketch out the steps of a sort of generic session—something that would demon-

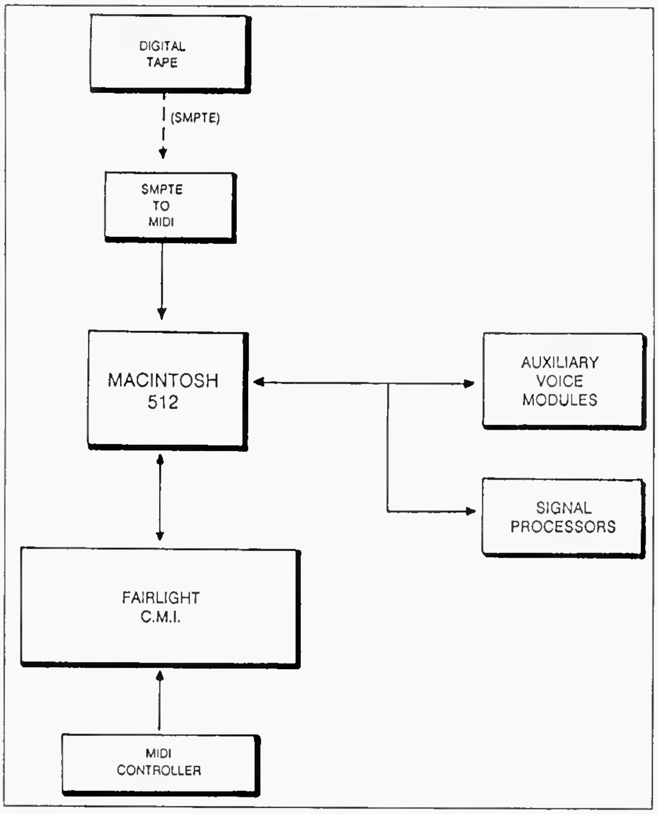


Figure 4. The MIDI signal flow at Visual Music.

strate the universal capabilities of their system, and they gave me the following chain of events.

A client would approach Visual Music to see if their production scheme would be a viable cost-effective alternative. (He might be a songwriter/artist seeking to make a master quality recording to give him the edge in marketing his product; he might be an audio producer with record company backing who wants to make a digital recording on an analog budget. He might also be a video producer who has become aware of the need for better quality television sound. Using an off-line video dub, even film producers can enjoy the benefits of first generation audio.) In any case, the production would usually start at the Fairlight CMI. Using the onboard sounds in the CMI and its internal sequencer for recording performance data, a series of basic tracks would be recorded. Provisional sounds would be assigned for parts performed on the Fairlight. (Later on, these sounds could be replaced by any other sounds available in the studio. That's when the real fun begins!) It should be noted that the Fairlight keyboard is not the only way to articulate performances into the sequencer: guitar controllers, breath controllers and old-fashioned (but ever so accurate) step time are also possibilities.

When the internal capacity of the Fairlight has been reached, the performance data are then dumped into the sequencer program on the Macintosh computer. While the Fairlight can only hold 8 tracks of data at a time in its onboard memory, the Personal Performer software running on the Mac allows sequencing up to 266 tracks! While few producers would need the total capacity,

Henry notes, "It does afford the luxury of taking say 40 tracks of a solo which will eventually end up as a single track in the mix. Then you can kick back and pick the best track or even the best parts of each of the 40 tracks and merge them into one." Although this might sound like a highly technical way of capturing a performance, he adds, "It actually preserves the spontaneity of the performance and encourages the artist to get creative and take some chances without fear of blowing it."

TO THE COMPUTER

This process of cutting tracks to the capacity of the Fairlight and then dumping them to the Mac can be repeated over and over. On playback from the computer, the Fairlight voice cards can be called up or sounds from any other piece of gear in the house can be utilized as a replacement. Any natural instrument (let's say the client's guitar) can be sampled on various strings and in several modes of articulation such as muting, strumming, picking etc., and re-performed as MIDI information through the Fairlight. Right down to the final mix the sounds, performances and even the "feel" of performances can be constantly adjusted without having to wear bald spots into your multi-track tape.

But let's face it folks, despite the protestation of audiophiles like Jay Henry, there will come a time when most projects will have to "kiss tape." (After all, who did the last "direct to CD" project that you know of?) If it's a matter of linking vocals or actual instrumental performances to the virtual tracks, digital or even high quality analog tape striped with SMPTE time code will allow them to run synchronously right down to the final mix.

The applications of this kind of production scheme in audio for video has just begun to be exploited. According to Gene Perla (who doubles as videographer for the company), "The market for this kind of service is voracious! With satellites beaming programs all over the world, 24 hours a day, we are just seeing the tip of the iceberg in terms of the demand for quality audio."

Henry, on the other hand, whose major interest still seems to be making hit records, sees another virtue in Visual Music's production system. Since their whole facility—including a Trident console (with a recently hot-rodded front end)—is on wheels, Visual Music can be transplanted to a client's home where the atmosphere is more conducive to creativity. The futurist philosopher emerges in Henry as he muses, "The species 'humanus musician' has a need to be creative—(a drive which often gets short-changed in many commercial sessions)—and what better place to be creative than the home. The way things are going it's going to be a cottage industry." Citing the well known mortality rate for relationships amongst media professionals, he notes, "Think how many marriages we can save" by bringing the "professional modular studio" to the client, thereby saving him the often required trips away from his family.

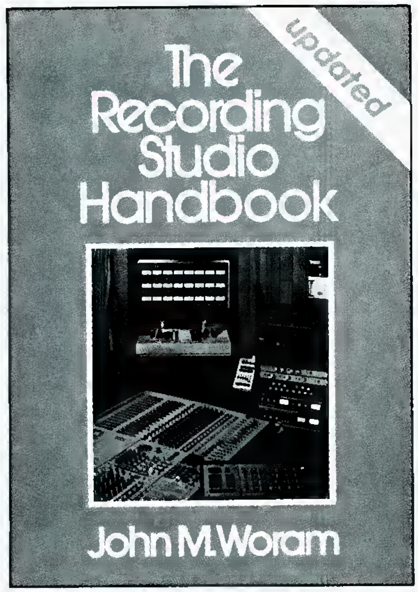
HOW COST EFFECTIVE IS THIS?

While the Utopian vision of the electronic cottage, ultra-creative musicians and happy families is all quite appealing, the obvious bottom-line question still remains: How cost-effective can this really be? According to Henry and Perla, at Visual Music's book rate of $750.00 per day, production costs can usually be reduced by 50 percent (as compared with turning out a similar product from start to finish in an "on-line" studio). By producing a project in the client's chosen environment where creative juices can flow freely and the workday can be maximized, a high quality product can be recorded quickly. If for mixing or layback the services of a major on-line studio (with extensive automation and outboard gear) is required, the Visual Music system can be transported there for the final mix.

The last question posed to Henry and Perla concerned the place of the dedicated on-line recording studio as we know it today. Will it disappear like the dinosaurs or will it adapt and prosper? According to the oracles at Visual Music, the big-time studio will always be with us, but will also interface nicely with the professional home studio. Already we can see major studios adding more intimate off-line suites with similar systems to Visual Music. While there is a great deal of compatability between systems today, total universality has not yet been achieved; until the new age in audio is in full bloom, companies such as Visual Music will continue to bring digital tracking from their space to your space—and anywhere else you might want to go. db

For Your Audio Library

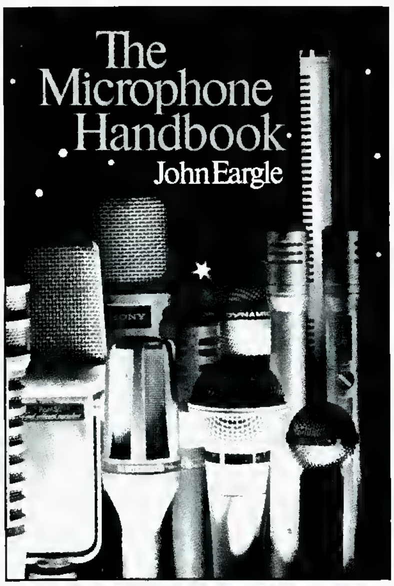
The Recording Studio Handbook



by John Woram **$39.50**

The Recording Studio Handbook is an indispensable guide with something in it for everybody. It covers the basics beautifully. It provides in-depth insight into common situations and problems encountered by the professional engineer. It offers clear, practical explanations on a proliferation of new devices. In this updated edition, among the items covered are: Transducers, signal processing, noise reduction, recording techniques and more . . . In addition, it has been expanded to feature three all-new chapters . . . chapters on the in-line recording studio console, digital audio and time code implementation.

The Microphone Handbook



by John Eargle **$31.95**

Completely up to date, this vital new handbook is a must for any professional whose work involves microphones. Among the topics covered are: Directional characteristics—the basic patterns, using patterns effectively, microphone sensitivity ratings, remote powering of condensers and proximity and distance effects. Other topics include: Multi-microphone interference problems, stereo microphone techniques, speech and music reinforcement, studio microphone techniques and microphone accessories. You'll find yourself reaching for it every time a new or unusual problem crops up.

Please send the books indicated below:

________ copy(s) of The Microphone Handbook at $31.95 plus $2.00 for postage and handling per book.

________ copy(s) of The Recording Studio Handbook at $39.50 plus $2.00 for postage and handling per book.

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COREY DAVIDSON

Room for a View

Frank Filipetti is a highly recognized producer/engineer. His work with artists such as James Taylor, Carly Simon, Hall & Oates, Foreigner, Kiss, and John Waite speaks for itself.

On Sunday, September 13, 1987, I met with producer Frank Filipetti at Right Track Recording Studios, New York City, where he was working on Foreigner's newest album. His opinions regarding the sonic differences between the analog and digital formats of recording are at the heart of the controversy that has been surrounding this subject for a few years now. The question raised by the analog versus digital dilemma might be, "If some of the differences are not measurable on currently available test equipment, are there differences?" We enter a very subjective realm when using words like "warm," "bright," "harsh," "forgiving," "musical," "round," etc. to describe these differences.

I told Frank about the many instances where it has been virtually impossible for numerous audiophiles to decide which storage medium is truly better and that the result is usually a trade-off as to the differences between them. He took exception to my statement and said, "About six months ago the people at New England Digital asked me to listen to their new direct-to-disk system. So they came over here. It was a very complicated procedure, involving a great deal of equipment, and they set up for a one-hour demo to let me listen to it because they know my feelings about digital. The first thing they did was play me their system at the 50 k sampling rate. I listened for a while and said, 'Nice try, guys. It's a bit better. The high-end is nicer, the compression of the stereo image is not quite as bad as with the other digital storage mediums, but it's still not right.' The Synclavier people (New England Digital) now wanted me to listen to the system at the 100 k sampling rate. I listened and for the first time I heard a digital system that I could use. I might actually prefer this NED system to an analog system. The problem with the 100 k system is that it does not interface well with a console. Right now the Synclavier system must be used with their keyboard. I have been in conversation with the NED people and we're working on a concept to integrate it to a console."

EVALUATING THE MITSUBISHI

Frank was asked by Eddie Germano, owner of the Hit Factory recording studio (Manhattan, N.Y.) to evaluate a Mitsubishi X850 digital tape machine that the studio had recently purchased. Frank's impressions were, "I listened to the Mitsubishi X850 and it was better than some other (digital tape) machines. There was a little less compression of the stereo image, however it still wasn't right. At the same time I was in touch with people at Apogee Electronics. They told me that they had analog filters that can be put on the front and back end of digital recorders. They also said that these filters are of a much finer quality than the filters that the digital tape recorder manufacturers use. I spoke to Eddie Germano about doing a test with these filters. He agreed and we set up the test to see if a difference could be heard between the modified digital X850 and an analog Studer. The people from Apogee Electronics put different filters on different channels of the X850. I had no idea which channels were which and I didn't want to know. We started recording an array of

We at db Magazine feel that Frank Filipetti certainly deserves the forum to voice opinions, however, these are his opinions and do not necessarily reflect the opinions of db Magazine or its staff.

tape. Inside of a couple of minutes I was able to pinpoint which channels were digital and which channels were analog. It was very easy to discern."

WHY WAS IT EASY?

I asked Frank to describe the differences that led him to the identification of the various digital channels and the reasons why he thought that the digital channels were sonically inferior to the analog channels. He replied, "There are about four major differences. One is that the top end is totally messed up. Psychoacoustically there is a hump that starts at about 10 k and continues on up to about 16 k where it drops off dramatically. With a test tone, the response looks like a table top. What we hear and what a test tone does are worlds apart."

I asked Frank to explain the psychoacoustic curves. He said, "What I feel is happening at approximately 10 k is that a massive phase-shift begins to take place. Now even though the test tone shows a wonderfully flat response, anyone who really understands recording knows that if you shift the phase of a signal in a certain area, it will sound like it's sonically enhanced or decreased. The low bottom from about 50 Hz down is abnormal when compared to the source. Once again this abnormality is probably due to more phase shifting. The end result is that it lacks fullness and power. The next thing that happens upon listening to a stereo mix through the digital machine is that the field and depth of the stereo image is reduced. The effect is that it's compressing the stereo image, approaching mono characteristics. For a machine that's supposed to have one hundredth of the crosstalk of an analog machine, this does not make sense. Digital proponents will even admit that ambiences and reverbs become somewhat dried up. One has to add a little more reverb and a little more ambience to compensate for this loss. What they don't understand is that it's not the reverb that is being dried up. The digital machine cannot tell the difference between reverb and any low-level signal. Any kind of low-level, low-resolution signal can't be properly interpreted by a digital machine. Digital machines are not only not handling the low-level reverbs, but also not handling the high order harmonics. That super-fine, high-order harmonic detail is giving us the cues that enable us to tell where a sound is coming from. That super-fine, high-order harmonic detail is affecting the field of the stereo image as well. It's these little cues that are happening well behind the main information that the digital machine just doesn't know how to handle."

Many proponents of digital recording make a comparison to analog based upon noise levels. I asked Frank to share his views on this matter. "Noise is not an issue. I never compare the digital return to the analog return. I always compare the digital return to the analog return from the SEND. When you do that, and analyze the send compared to the return from the machines, the digital always loses."

HOW DIGITAL IS EVALUATED

Recently, while working on a project in Nashville at Masterfonics Studios, owner Glenn Meadows invited Frank to make an A/B comparison between an unmodified JVC-VP900 digital mastering system and a Studer A820. Frank stated, "Every time I do a digital test, I simply compare the input with the output. The outcome of these comparisons usually stirs doubt in the minds of engineers and

Frank Fillipetti ready to work the console.

owners. They begin to wonder if the machines are properly aligned. The results of the test conducted at Masterfonics was very much like the other tests conducted elsewhere...analog 1, digital 0."

I spoke to Glenn Meadows at Masterfonics and inquired about the test that was run by Frank. Glenn confirmed the results. However, he also informed me that he has since installed the newest versions of the Apogee analog filters. Furthermore, Glenn asked me to inform Frank about this recent modification, and to tell Frank that his ears are once again invited to make a new test. Glenn insisted that the new modification has eliminated the problems that Frank reacted to the first time.

Although it is sometimes dangerous to make non-technical analogies to very complex physical properties, there are times when analogies help to explain those complex properties to less technically oriented people. Frank has formulated a very interesting analogy that clearly explains his point of view, "Corey, when you asked me to describe the differences that I hear in these A/B tests, I explained the results in terms of frequency, phase, psychoacoustics, etc. An analogy that I have come up with is one that explains my point visually. If you look at a landscape, you see with your eyes, a scene. This is the input to the console. You see an amazing, three-dimensional, high quality image. The image that a digital machine produces is analogous to a computer enhanced photograph, or more to my liking, a paint-by-numbers version of this landscape. We've all seen paintings by numbers. There is no transitional detail in paint-by-numbers. It goes from red to orange with no transitions in between. Now, because of that, it seems as if the edges are much more defined as it is with digital...the transients are much more defined. The average ear might interpret this as greater clarity because the edges are there and the sharpness is there. The analog is not the landscape either. The analog is a Monet painting of this landscape. It's the impressionistic version, fuzzy around the edges. When you look at a Monet up really close, you can't tell what's going on. Take a step back and it's fuzzy. Step back ten feet and the clarity is there. The bark of the tree isn't sharp and defined, it's fuzzy. If you look at the subject with your eyes, the real landscape is very defined. Again, the Monet is fuzzy. But if you take that paint-by-numbers with all its definition and you take the Monet with all its fuzziness and stand back from those two things, one is a much more realistic representation of that scene than the other. Granted, the paint-by-numbers has that clarity, but it doesn't nearly convey what the scene really is as well as the Monet. That's what I think people are responding to with the digital mediums. Yes, it is clear and it has the transient information, but doesn't provide the nuance and fine harmonic detail. The analog does compress the transients, however, in the end the analog is a much truer representation of what we actually hear. I would love to use a digital medium. I'm not anti-digital. I would love to be able to use this technology. But it isn't right yet nor is it as good as what we have."

WORKING WITH FOREIGNER

In so far as rock bands go, Foreigner is one of rock's more versatile, dynamic and well-orchestrated bands. Frank Filipetti's results with their last album, *Agent Provocateur*, (not to mention its multi-platinum success), has resulted in his involvement with their latest recording endeavors. He comments, "On this project, again, I stayed with the Studer A800."

THE CHOICE OF FORMATS

I asked him whether or not he had a choice of recording format and location for this project. He stated, "I have a choice on anything I work on. When we did James Taylor's album three years ago, it was the first time I seriously considered digital. We had a Sony 3324. I wanted to use a digital machine on James Taylor. If there's anything that cries out for digital, it's a James Taylor album. If digital gives us this 90 dB dynamic range and amazing clarity, how can you beat a James Taylor album? James is a guy that goes from one guy with an acoustic guitar, to a full orchestrated band sound, unlike most rock and roll bands. 20 dB dynamic range is a lot for most rock and roll. James Taylor is another story. So we had the digital machine come in. We had it here for four days and day after day we had all kinds of people coming in to work on the machine. For four days we recorded analog and digital simultaneously. There wasn't a single digital playback in that entire four days, not one, that was preferred by me, James, the band, or anyone over the analog. Now, there are some people that claim to use digital, and the artist comes in and says, 'I've never heard such clarity on my voice.' Fine. I don't denigrate anyone for using digital. Just don't go to the papers, and don't go telling me that it is more realistic than analog. If you tell me that you want to use digital because you can make copies with no generation loss, and that what you put in is exactly what you get out, I'll fight you every step of the way because, number one, not only does digital not reproduce exactly what you put in but, number two, it can't make copies that are clones. There is so much error correction going on in these machines. You are not listening to playback. You are listening to forty to fifty percent playback, forty to fifty percent error correction. These machines are constantly analyzing what information they have and what information they don't have, making assumptions about it and playing it back. Because of drop out rate and errors, it's always playing something different. You may not hear that, but what I'm saying is that if you keep making successive generations starting with copy one and you have to make one hundred copies, upon listening to copy number one hundred, I guarantee that you won't recognize what you're listening to. Now, analog can't do that either, but analog never made that claim. You can make generations more easily with digital, but it is not a one-to-one copy."

TWO CONCEPTS

There are two concepts that our readers should understand. One is that with digital, there is no distortion all the way up to the end of the 90 dB range. Upon going 1 dB beyond that range there will be nearly 95 percent distortion. With analog, distortion begins to occur earlier into its dynamic range (about 60 dB) and increases with each step up, giving us a continually increasing curve. Secondly, bandwidth plays an important role. The bandwidth for the digital machines is much wider than its analog counterparts. This culminates in digital's superior ability to handle transients information.

Frank comments, "To avoid reaching the sudden threshold that throws you into distortion on a digital machine, you can record a little lower than normal due to the 90 dB dynamic range. Right? Then why do the digi-techs tell us to 'record as hot as you possibly can?' Once again, digital does not handle low level information. From 1 dB to 100 dB is not one hundred steps. It is a logarithmic progression that winds up being one hundred thousand pieces of information. Digital has to make some assumptions about that. It's only got sixteen bits that it can handle. It's only got 64 k that it can handle. So it's got to make a lot of assumptions about those one hundred thousand pieces of dynamic information. It's got to compress that 100 k into 64 k. When it sees low level information from about 20 dB on down, the errors and assumptions are even greater.

An analogy can be drawn between dBs and the frequency range. In the realm of frequency, in the first octave from 30 Hz to 60 Hz, there are thirty cycles. An octave from 5 kHz to 10 kHz, contains five thousand cycles. There's a lot more information in that 5 k to 10 k range than in that first thirty cycles. In the decibel range, it's the other way around, yet similar to the case of frequency. There is a tremendous amount of dynamic change in that first 30 dB. In the end, the reasons for going analog on this and every project that I've worked on up to this day, is because, as far as I'm concerned, digital does not sound as good as analog. Some say that they like the warmth and compression of analog. Every time I listen to what the Studer does to my snare drum by compressing the impact of that drum, I am distressed. I don't like that. I don't like what it does to the transients. I don't like what it does when I have to make a vocal composite and go to a second generation. Anyone who thinks I like that **** is nuts! Nevertheless I've got two storage mediums. I've got a digital and an analog. In the end I have to look at what is more important than a transient, and that is the overall musicality of it. Again, going back to the Monet and the paint-by-numbers analogy, the Monet is a truer representation of the image. It may not be a correct representation but on the day that a medium is found that gives me what I see with my eyes in that landscape, I'll be the first one to jump on that bus." db

MICHAEL KLASCO and RUDY PAOLINI

San Diego Sports Arena

Sports arenas are not usually known for superb acoustics. At the same time, major musical artists want to perform in those huge arenas, so steps are being taken to not only improve acoustics, but to make them musically good. Here is one such arena in Southern California..

Acoustical engineering consultants, like most people in business, find that some new jobs come as a surprise, while other projects can be "heard" approaching.

The San Diego Sports Arena fell into the latter category. The twenty year old arena recently had the roof redone, seats re-upholstered and a video system installed. Wavelength Systems Design, Inc., the video contractor (and one of my clients), had remarked that the Arena was a needy candidate for a complete audio/acoustical renovation.

Initially, a local audio shop promoting an "omni" sphere-shaped speaker enclosure as a cure-all had spent months unsuccessfully attempting to achieve acceptable results. Luckily, the arena management was astute enough to realize that a few speakers filling the space with reverberent sound were not quite the solution, but the real goal

Figure 1. Reverb time plot for 80-125 Hz before acoustical treatment.

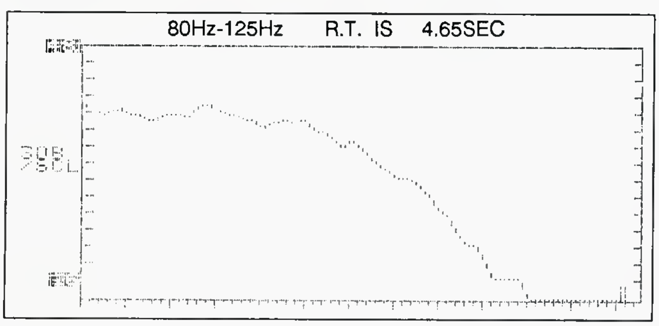
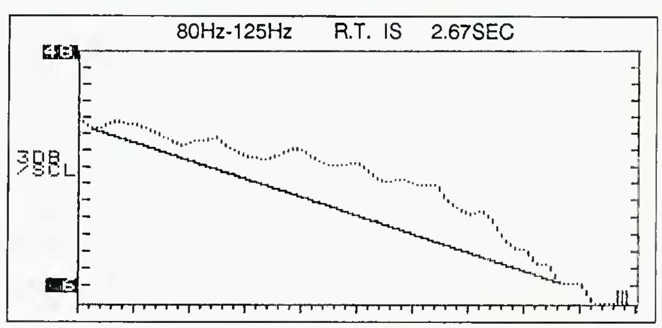


Figure 2. Reverberation time plot for 80-125 Hz after acoustical treatment.



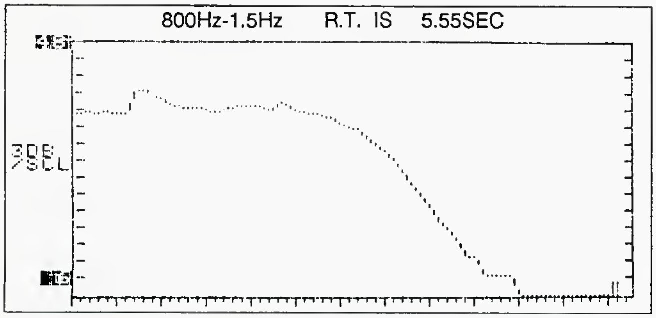


Figure 3. Reverberation plot for 800 Hz-1.5 kHz before acoustical treatment.

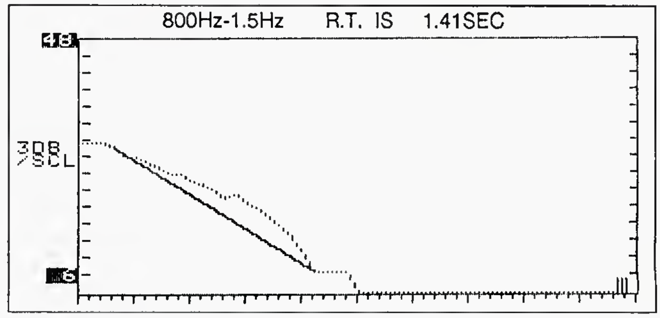


Figure 4. Reverberation plot for 800 Hz-1.5 kHz after acoustical treatment.

was sound that was both intelligible and aesthetically satisfying.

The project management of the ongoing facility renovation was handled by the construction consulting firm of Wagner-Hohns-Inglis for San Diego Entertainment (the operator/lessee of the Arena). To assure the lessor, the City of San Diego, that the acoustical/audio deficiencies of the arena would be competently remedied, this aspect of the renovation also came under the supervision of the project management team. As the outside acoustical consultant to Wagner-Hohns-Inglis, I was to develop and help implement a comprehensive plan to provide high-quality acoustics and sound reinforcement for the facility.

BACKGROUND

The San Diego Sports Arena is an enclosed oval-shaped structure originally designed for sports such as basketball and ice hockey. Seating capacity is 15,000, length is 400 feet, width is 300 feet and ceiling height is 95+ feet. The Sports Arena opened in 1966 and, like most facilities of this type and time period, the designers paid little attention to acoustics. Architects and owners are generally pragmatic and focus on ease of maintenance and durability through fifty years of anticipated use. Therefore, interiors tend to be hard, as acoustically reflective surfaces such as concrete, cement plaster, steel, masonry and hardwood are the dominant construction materials. Even the original seating was a thin foam pad with a vinyl covering.

When people entered large volume spaces in less aurally sophisticated times, echoes and poor intelligibility in the sound system were expected and this arena was no exception. The architectural acoustics of the Sports Arena were characterized by strong flutter echoes and a reverberation time of over five seconds through most of the audio band. Although crowd sounds and "excitement" are accentuated by noisy, cavernous acoustics, announcements and narrations of events tend to be fatiguing, and musical performances are aesthetically marred and generally unacceptable. Since 1985, the operating company of the Sports Arena has been actively upgrading the facility including a new roof, re-upholstered seating, added lighting and video monitors in all concession stands. In June 1986, the Arena management was ready to undertake the most complex part of the renovation, the acoustics.

Although originally designed for sports events, much of the Arena's revenue is now derived from rock concerts. It was determined that the Arena must meet the demands of the more discerning audiences of today. The great strides made by the music and audio industries over the last two decades has profoundly altered the expectations of the typical concert goer. As a concert ticket costs more than a digital disc, the live concert should have at least the real-

The plots in Figures 5, 6, 7, and 8 were created by the Sigma RS-4000, using gated pink noise. They show the contour of decay both before and after the acoustical treatment. Note that the time period shown on the after plots is 30 percent shorter than the before plots.

Figure 5. 3-D spectral plots before treatment.

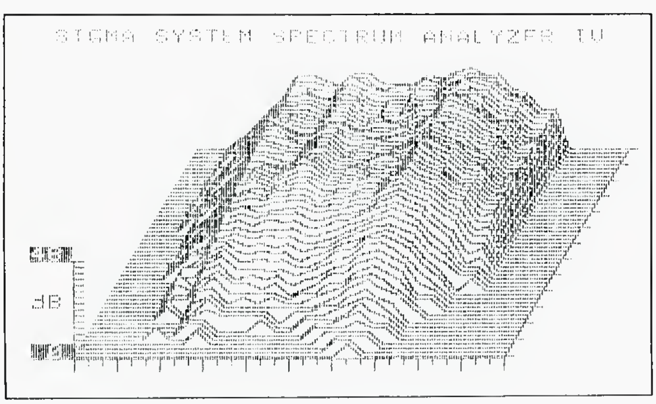
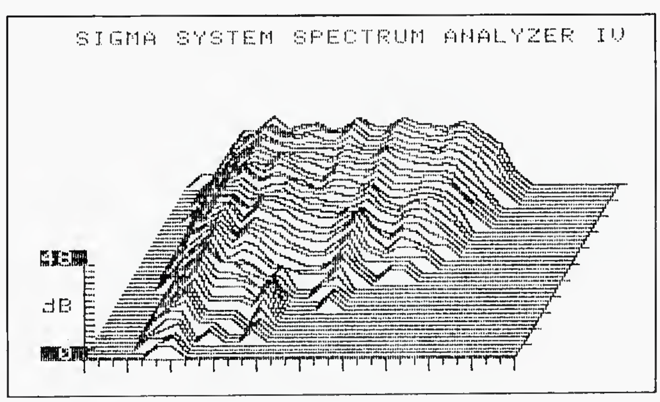


Figure 6. 3-D spectral plots after treatment.



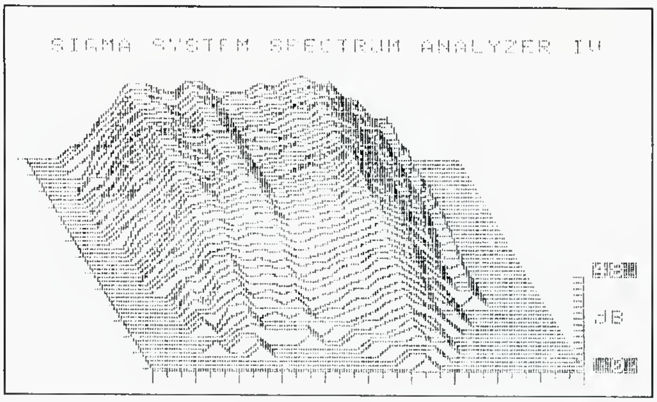


Figure 7. 3-D spectral plots before treatment.

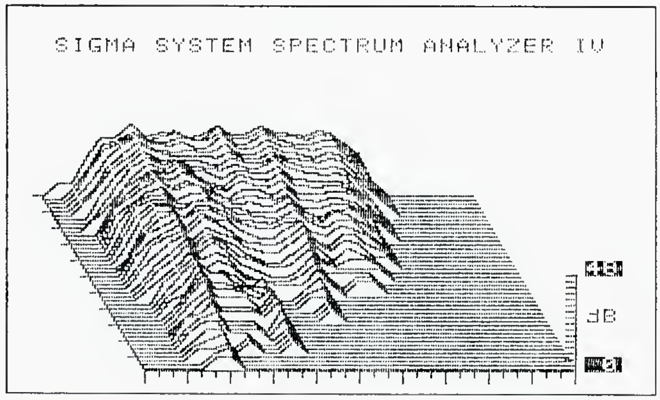


Figure 8. 3-D spectral plots after treatment.

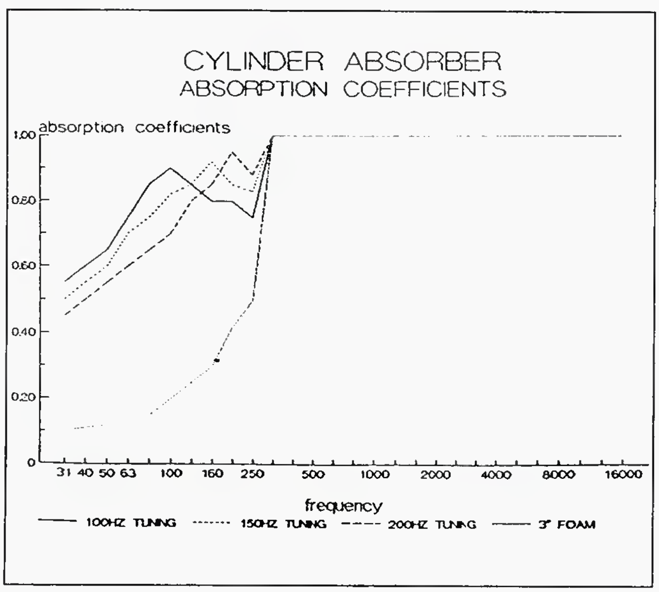
ism, dynamics and realism of the disc. After an unproductive experiment with the omni-speakers, the Arena management committed themselves to the full-scale acoustical/audio renovation proposed by Rudy Paolini, architect and senior vice- president of Wagner-Hohns-Inglis in Pasadena. Rudy asked me to devise, computer model and test the acoustical modifications. Additionally, the house sound system was to be evaluated and interim improvements were to be specified to bring it up to acceptable levels until more comprehensive revisions could be attended to the following season. Wavelength Systems Design, Inc., El Segundo, California was awarded the contract for installation of the acoustic treatment and repair of the sound system. Matt Blondon, vice-president of WSDI, headed the installation team.

ACOUSTICAL DESIGN STRATEGY

The approach we took for the Arena was to analyze the existing acoustics, determine what the target characteristics should be, and develop a program of acoustic renovation that would satisfy these requirements within the budget, aesthetic considerations and building codes.

The Arena's acoustics were measured (and listened to). The reverb was timed and echogram measurements were taken. The dominant problems were reflections concentrated in time (flutter echoes) and excessive reverb decay time. By observation, it could be seen that the cause of the problems was the exposed steel deck, parallel 400-ft. side walls and 300-ft. concave curve end walls, both of concrete. When the seating had been re-upholstered, the heavy padding of the seats was designed to have a similar acoustical absorption characteristic regardless of whether the chair was occupied or not. Essentially, the acoustic problems were created by the bare walls above the seating, and the metal ceiling. From the test signal echograms (captured by using an IQS FFT spectrum analyzer as a digital storage oscilloscope), the time/energy curves showed strong reflections from the central cluster speakers off the ceiling, and from the upper walls. Refocusing due to the concave end walls was also apparent. Some of the reflections from the ceiling could be avoided by properly aiming the horns (which was soon taken care of by WSDI). Further reduction of the ceiling reflections caused by the mid-band vertical diffraction of the Altec multicell horns would have to wait until these could be replaced. Later this year, large mouth constant directivity horns will be installed. In the interim, the crossover point was shifted upward slightly so the Altec horns and direct radiator bass boxes were used with the best compromise of pattern control and uniformity of seating coverage.

Figure 9. A graph of the cylinder absorber's absorbtion coefficient.



Since the entire upper half of the Arena consisted of acoustically reflective surfaces, unless the central cluster could have an extremely hard pattern (high Q constant directivity speakers), too much sound would spill over to the bare walls. Tightening the pattern of the house sound system to minimize wall reflections would be at the expense of uniformity of coverage of the seating areas near the walls. Regardless of the directivity of the house sound system, the rock groups use separate sound systems brought in by their touring companies, and these were not always ideal for the Arena's acoustics nor under control of the Arena's management. Other exhibitions, such as the circus, brought in their own overhead distributed cluster sound system. The crowd ambient noise is, of course, totally unaffected by whatever coverage pattern is used. The comprehensive solution is to control the acoustic space, which will allow more acceptable performance from the wide range of sound system configurations used in the Arena.

ACOUSTICAL MODIFICATIONS

Basically, our approach was to absorb and/or diffuse the

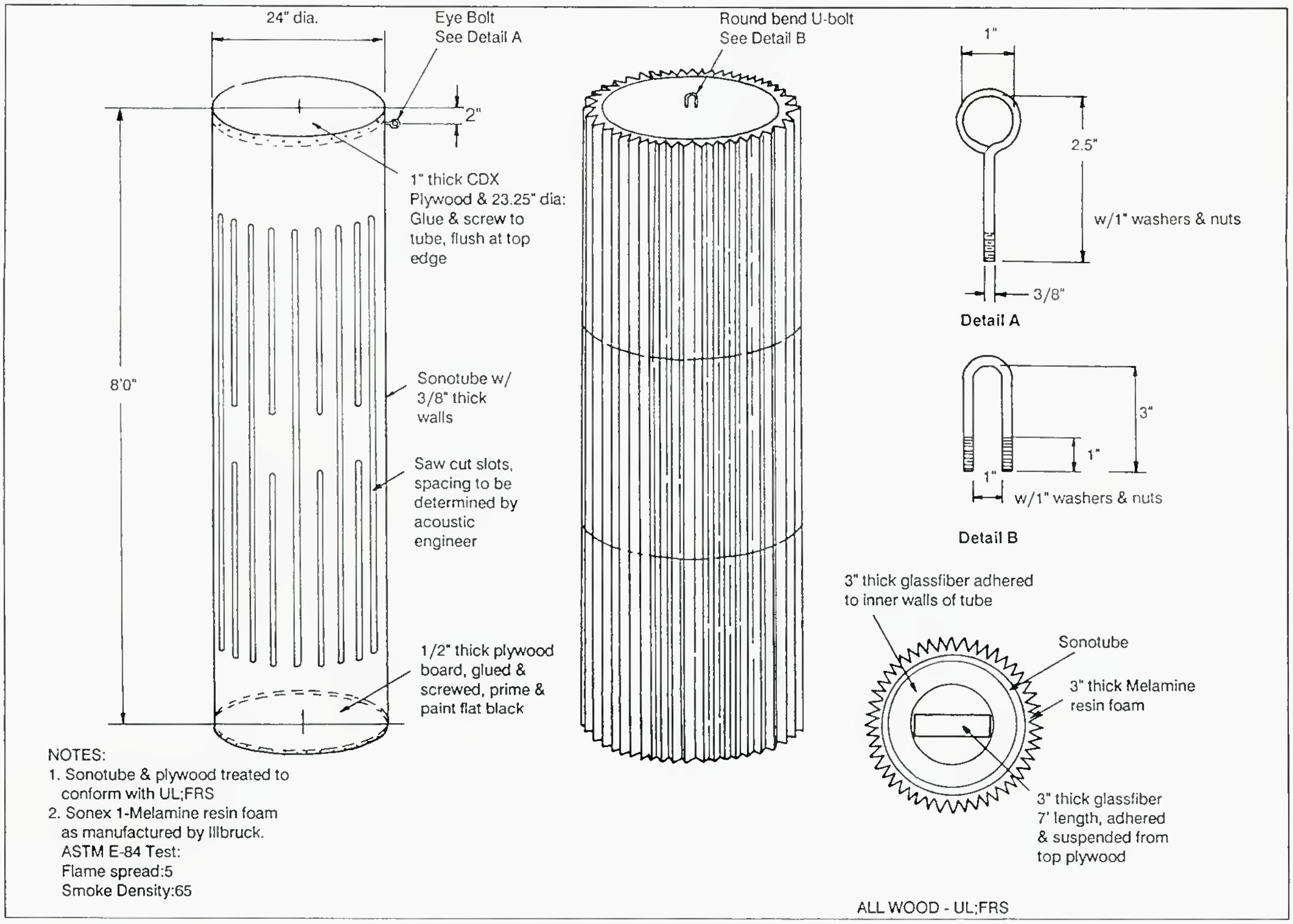


Figure 10. A detailed construction diagram of Helmholz tube-type resonators.

strong reflections and reduce and contour the reverberation time. Questions studied and resolved were appropriate reverberation times for a rock arena, how we were to achieve the absorption and diffraction, and how much of the budget should be spent on absorption versus diffraction.

The typical reverberation time of most performing halls is about 1.5 seconds mid-band. These halls are primarily for non-amplified classical performances with seating for 500 to 2500 people, not the rock/heavy metal concerts for 15,000 people that are popular at the Arena. Although rock concerts are the main revenue source for the Arena, Pavarotti and the San Diego Symphony also play there. Considering the large internal volume of the space, reverb times of 2 to 3 seconds at mid-range frequencies would be a practical target. Additionally, this range of reverb time (or less) would reduce the critical dependence of the sound systems used in the Arena on having the optimum

Figure 11. The view before acoustical modifications.

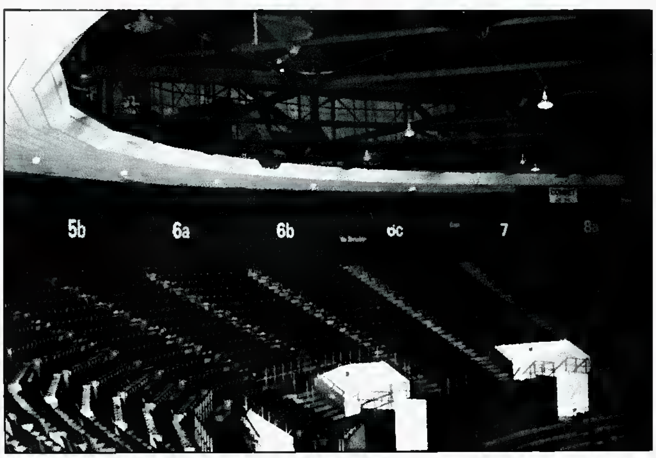


Figure 12. After acoustical modifications. Note the Helmholz cylinders.

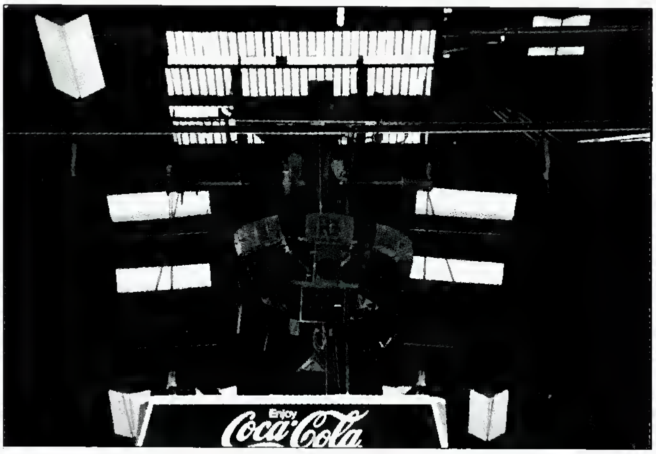


Figure 13. After acoustical modifications. Note the wedge absorbers above the speakers.

Figure 14. A bird's eye view of the Arena after treatment had been finished.

Q (directional characteristics) in order to achieve intelligible speech reinforcement.

The total budget for the Arena's acoustical renovation was $300,000. According to the computer simulations for the reverb time, this budget would allow us to use enough treatment to reach our 2-3 second mid-band goal, absorb much of the non-lateral reflections and diffuse or absorb the flutter echoes.

ACOUSTICAL SIMULATIONS AND TESTS

Reverberation: The acoustic tests were performed with a Sigma System RS4000 Computer Audio Workstation. The reverberation measurements used the sound system as the sound source, although gun shots, cannons or impulses could have been used as the test signal. For the test, the equalization was adjusted for the most extended response rather than for considerations of intelligibility, feedback or output capacity. The test system generated a pink noise source that was sent to the sound system under computer control. The test signal was gated off and the room decay was captured. The energy/time decay was displayed for each 1/3 octave band. The computer uses a curve-fitting algorithm to calculate the actual RT60 by finding a constant slope after the early discrete reflections and before the noise floor "knee" in the decay. To aid visualization of the spectral decay characteristic, a waterfall 3-D plot of the time/frequency/energy curve is plotted.

MORE ACOUSTICAL MODIFICATIONS

Instead of separate diffusers, low-frequency absorbers and mid-high frequency absorbers, unique multi-function devices were used to achieve the acoustical modifications. Considering that the RT60 had to be reduced by 3 seconds, our approach was to develop wide-band absorbers that would diffuse any energy that was not absorbed. To break-up and absorb the non-lateral floor/ceiling reflections and increase the average absorption coefficients for the ceiling, a wedge-shaped absorber was conceived. 8 feet long with two 2 foot faces, the wedge frame is covered with 3-inch wedge contoured acoustic foam.

The inside of the wedge is draped with 3-inch thick glass fiber in a W shape for maximum surface area. The absorption coefficients are over 1.0 for most of the frequency band due to the multi-plane geometry of this absorber. Even the low frequency absorption characteristic is high for a passive (non-resonant) absorber, due to the 8-ft. by 2-ft. wide face and 3-foot base. By suspending 200 wedges from the ceiling, we added 6400 sq. ft. of melamine resin foam and 10,000 sq. ft. of glass fiber. Each wedge contributed about 80 Sabines over most of the frequency range and all 200 wedges resulted in 16,000 Sabines absorption!

Initially, we had considered approaches that required less fabrication, such as hanging glass fiber bats with polyurethane covering but this lacked adequate low frequency absorption and robustness. Alternatively, we considered treating the ceiling with Certra-Spray, but the roof skin flexed too much to be an appropriate substrate for this material. The flexing, due to wind action and pedestrian traffic, would have caused chunks of the treatment to delaminate and fall into the seating area.

The second type of acoustic device developed for the Arena was to absorb and diffuse the reflections off the wall areas above the seating. Cylindrically shaped Helmholtz resonators 2.5-in. diameter and 6-ft. long were built. Low frequency active absorption was accomplished by slots cut into the 6-ft. tube. Glass fiber on the inside of the tube and 3-inch wedge-shaped acoustic foam on the outside provided aperiodic dampening of the resonator Q so almost an octave of absorption is achieved. The tuning of the 100 cylinder resonators is staggered from 100Hz to 200Hz. Above this range the absorption is crossed over to the acoustic foam. The cylinders are located about 25 feet out from the walls and suspended from the trusses, 5 feet below the "acoustic" ceiling that extends 30 feet out from the wall. Over 4000 Sabines of wide-band absorption are contributed by the cylinders.

The wall height above the seating extends 25 feet to the suspended ceiling, and was a major source of slap back. A 6-ft. band of 3-inch thick acoustic foam was run the entire 1200-ft. perimeter, contributing about 7200 Sabines of absorption.

To satisfy the rigorous Class I and ASTM-E84 fire code specifications set by the fire marshal, we used Ilsonic foam, fabricated by Ilbruck of West Germany.

The foam was cut to a 3-inch deep wedge contour by SAMCO of Westland, New Jersey. In the U.S., this foam material is also sold by Ilbruck's subsidiary Sonex and the product is marketed as Sonex I. SAMCO contours the foam with a blade cutting technique, allowing wedges of 1+ft. thickness. Sonex uses a hot wire technique which is presently limited to a maximum of 2-inch thickness for Sonex I. SAMCO's standard product, Tedron, is a pyramid pattern but for the arena I selected a wedge pattern similar to Super Sonex, as the absorption dropped off less rapidly below 500Hz, compared to the regular Sonex or SAMCO Tedron.

To reduce the speaker cluster reflections from nearby surfaces, the top of the four-sided score board was covered with 3-inch acoustic foam.

An acoustic ceiling is suspended 20 feet below the roof. Above the hung ceiling are bare concrete walls, the ventilation ducting, lighting, rigging and truss structure. This ceiling extends 25 feet from the walls, leaving a 340 x 260-ft. opening to the metal roof. To damp the reflections from this area, 20,000 sq. ft. of glass fiber was placed above the ceiling. 10,000 sq. ft. 1-inch was placed on top of the suspended ceiling and 10,000 2-inch material was positioned along the walls. Installation was quick as the material was semi-rigid, and materials that might be seen from the seating were supplied (by Manville) with black coloring. The glass fiber added another 20,000 Sabines of absorption.

THE HOUSE SOUND SYSTEM

The house sound system consisted of a central cluster of Altec multi-cell horns and direct radiator sealed bass enclosures with Altec 12-inch woofers. The entire system was powered by Altec amplifiers and an old broadcast style Altec tube console. The system was essentially unmodified since its original installation in 1967, except for the unintentional misalignment of the horns when the new score board had been installed a few years earlier.

Although complete renovation of the sound system was not planned for this phase of the operation, the most critical problems needed to be attended to immediately, as the house sound system was almost non-operational and many complaints were received by the management over the unintelligible sound quality.

The Altec woofers and enclosures were replaced with EV TL606 (single 15-inch woofer/vented box), new diaphragms were installed in the compression drivers. The horns were re-aimed, although the budget and time frame did not allow for a comprehensive analysis or computer simulations at this time.

In the control room, the console was retired and a 8 channel Ramsa mixer was installed. The flexible output section permitted individual level control of sections of the cluster to accommodate the different coverage requirements of soccer, wrestling, etc., as the playing field size was altered. The passive crossover networks were replaced with Ashly electronic crossovers and an Ashly parametric equalizer was put in. QSC 1400 amplifiers replaced the Altec amplifiers. Later this year, a comprehensive design analysis for a new cluster will be prepared, and large format constant-directivity horns and drivers will be installed.

SOUND SYSTEM LAYOUT FOR ROCK CONCERT USE

Each rock group that plays the Arena provides its own sound system. Typically, a concert sound company is contracted for a number of shows for the group and moves the sound system with the rest of the show. The configuration of the Arena for the rock concerts is to locate the stage at the east end with the concert stacks to the sides and above the stage. For consistent results for the various and hastily assembled concert sound systems, a large panel (1000 sq. ft.) of acoustic foam is suspended from the 90-ft. ceiling to reduce the mid and high frequency energy spectrum of the non-lateral "early" reflections.

One of the insights of recent studio control room design was the importance of reducing the mid/high frequency energy content of the early reflections thereby reducing the amount of higher frequency energy content of later reflections, (regardless of the absorption coefficient characteristics of the distant surfaces). The point is that the most efficient use of absorbing material is near the sound source, ie. the house cluster and stage stacks. Of course, a 15,000 seat arena is not a studio, or even a concert hall, and control of reverberation time and crowd ambient noise also requires a proportion of acoustically absorbent material distributed throughout the space for a reasonable average surface absorption coefficient.

RESULTS

In May, the *Ice Capades* performed at the arena. This show has always resulted in numerous complaints of unintelligible and fatiguing sound. This time the sound quality and intelligibility were excellent and the arena management realized that their acoustical problems were solved. A second problem the arena management had was the poor press they have always received on the acoustics of the facility. In early April, the (San Diego) Tribune reviewed a Merle Haggard/Judds/Alabama concert. I will provide a quote from the review: "The night's reigning star was the technology. The sound system undid 20 years of carping about acoustics in the arena." While the reviewer was not impressed with any of the groups' performances, he stressed that "the marvels in last night's concert were technological." Rather than having to refund tickets from unhappy concert goers due to the poor acoustics, the arena management now has a facility whose acoustics are used as a marketing feature to book shows (using the motto "We Broke the Sound Barrier").

Recently the popular rock group *The Pretenders* performed and a number of the improvements became apparent. Before the room treatment it was impossible to localize the direction of the stage (or the back wall was sometimes localized), now the stage was clearly the source of sound. Even the compression driver distortion from the group's overdriven PA was clearly audible, while previously this would have been lost in the mud. Another subjective difference was the audience applause originated from the audience, while previously the apparent sound source was the roof.

After the installation was completed, the reverberation time measurements and echograms were measured. The results exceeded projections, with the mid-band reverb time at 2 seconds for an empty arena and the echograms showing that the flutter echoes were absorbed.

Each following concert and sports event has received a very positive reaction from previously dissatisfied sponsors and spectators and the project has been deemed a success. db

Back to the Future—Audio for Video

COREY DAVIDSON

Find out what we discovered about an impressive production complex where speed and efficiency are the watchwords; setting the pace for conformity and production techniques is a daily routine.

THE TELETRONICS AUDIO POST-PRODUCTION SUITE, with its compliment of computer-based audio control and processing equipment, offers a wide array of sound services. At the heart of the Suite is a Solid State Logic SL6000E Stereo Video System, a sophisticated, automated processing routing and mixing console, which provides extensive control over all audio program elements. The time code synchronized multi-track audio and video tape machines outfitting the Suite allow a large number of discrete tape channels and formats to be manipulated while locked to picture. Additionally, outboard analog and digital audio processors, effects generators, live recording ability and a comprehensive sound effects library complete the full range of services available to the music, video and teleproduction industries.

Larry Rosen, chief audio engineer, describes a typical session, "Most things that are done in this room are worked to picture. This complex is primarily a video facility, so this (the audio suite) would be the last step in the chain of events here. First, a shoot might be done on our Center Stage, a very large stage with a 50 by 60 foot hard cyc complimented by a 60 foot lighting grid. The material comes back to the video complex for editing. The last stage of production before duplication and ultimate broadcast would be the arrival to the audio room. A common procedure for a typical project is as follows: I take a 1-inch video master and transfer the information that is on tracks 1 and 2 (audio tracks on the master) onto two tracks of the multi-track (A800 2-inch 24-track) and print SMPTE time code on track 24. At the same time I'm going to make a 3/4-inch cassette. This cassette will serve as a work picture with its own burnt-in time code for reference. Now, the 3/4-inch cassette and the 2-inch 24-track will have the same time code as the 1-inch master. Then, via the Adam Smith synchronizer on the SSL, the 24-track and the 3/4-inch cassette will be locked together. The 24-track becomes the master and the 3/4-inch becomes the slave by the numbers that are generated by the Adam Smith synchronizer. I can now work with 24 tracks as opposed to the 2 tracks that are on the video."

Basically, Larry has now created an overdub condition that can accommodate whatever narrations, effects, music, etc. that might be added to the original visual material. Larry says, "When all this is going on, there is usually a producer sitting in that chair that you are sitting in. He often gives continual input as to how he wants the piece to come across as well as what effects and edits are to be executed. Next I mix these elements together and when it's all done and everyone is happy I take the finished product, which might be stereo or mono depending on what the producer wanted, and lay it all back to the 1-inch master. The completed piece is now ready for duplication and/or broadcast."

The aforementioned procedures were executed that afternoon for a major client. The product was of an industrial nature (not for broadcast), so more than 20 thousand 1/2-inch VHSs will be run off. Larry states, "Making a piece ready for broadcast can be as little as just re-EQing, conforming it to certain sound for broadcast that a producer has in mind, to adding effects, maybe have a foley man in the booth doing live effects to tape while he's watching the picture, add music, mix the music, customize the music...anything a producer has in mind, we can do in this room."

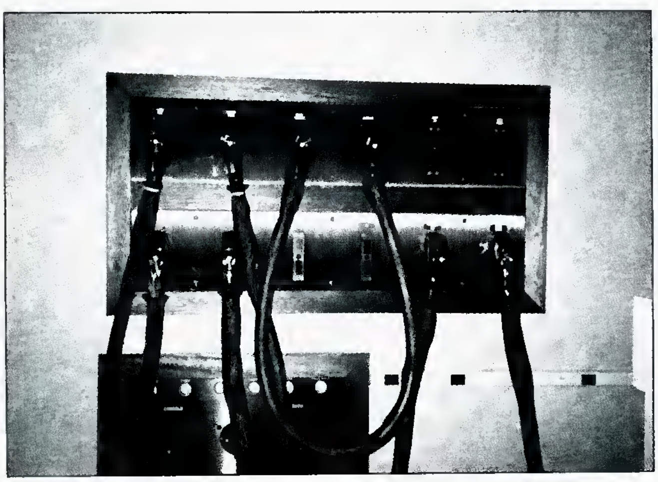
The types of projects that come in to VCA run the gamut from the mundane to the most wild and creative endeavors. This type of studio sees a far more eclectic array of projects than a typical recording studio would see simply by the nature of the audio for video studio's ability to satisfy a truly multi-media market. On the other hand, an engineer working a studio like this one (there aren't many) must be keenly aware of a client's needs, seeing to it that every step is taken to satisfy those needs.

Larry describes one of the more interesting sessions, "Recently I worked on a project funded by the Public Broadcasting System which is entitled *Steps*. The director and creator of the project is a Polish gentleman named Zbigniew Rybczynski who in 1984 won the Academy Award for Best Foreign Feature and has been producing music videos in this country for about 3 years now. He had this idea to adapt scenes from *Battleship Potempkin* by Sergei Eisenstein, a silent film made in 1925 and regarded by

Figure 1. Larry Rosen at the helm of VCA's "Audio Suite." You can just see under the telephone at the extreme right, the inter-studio routing matrix.

many circles to be one of the world's greatest feature films. Zbigniew's thought was to take the steps scene (the Odessa Steps), the most famous scene in the 1925 film, which is only about 3 minutes long, and create a scene, by re-editing, video wizardry, generations and the Ultimatte, that would be 25 minutes long. Although there really isn't the presence of a heavy philosophical message, the Rybczynski piece is a technical tour de force. The audio aspects were unique and unusual. Because we were using the Ultimatte IV, the set was made of wood and the sound of many feet walking up and down stone steps was needed. Normally, a foley man would be used to add sounds and effects after the piece is shot. He watches the visual and overdubs sounds accordingly. However, on this project, a foley man by the name of Mel Kutbay, from Germany, who did the special sound effects in the movie *Das Boot*, did the live sound effects for *Steps*. The audio was done at the production, live. Mel was watching the action, not a monitor, not a playback, no second chances. I had to be on the ball (to say the least) watching levels and keeping tracks organized. Because Mel uses his hands and his feet, I set up a booth/microphone rig that would capture the flurry of effects that this man was about to generate. I put an RE20 mic on the floor on a little pigeon stand and I set up a PZM

Figure 2. The custom-built, audio trunk-patch panel.



mic flush mounted on a piece of wood. The RE20 was perfect for the local, delicate, and highly directional sounds. Although the PZM was in a fixed position, the RE20 was accessible to Mel so that he could move the mic to suit the proximity to the sound that he was using at the moment. I kept his sounds split on one track of the 1-inch tape and on the other track I was organizing music and more sound effects. It was an exhilarating session. Mel opened a high-tech road case only to reveal bean bags, chains, knick-knacks, and dohickeys. His hands were going a mile a minute, changing effects at a staggering pace."

In the March/April and the May/June 1987 issues of db Magazine, articles by Marshall King appeared, addressing the issues regarding hi-fi stereo and network television. A quote from Mr. King's article in the March/April issue is as follows: "While it is the motion picture industry and not television itself that has inadvertently given the boost to TV stereo, there are those who for years have been methodically laying down fine stereophonic sound tracks in their day-to-day work recording television audio."

Larry adds to this subject, "Although most of my work is done in stereo, much of that will be mono upon broadcast. I occasionally have to take measures to reduce a wonderful stereo sound to a commercial, mono, compressed sound. Stereo TV and broadcast is good for people doing what I do, which is mixing for video. What's neat about a major network such as NBC broadcasting in stereo is that now, the audio is making as big an impact as the visual. Ad agencies now have to consider the impact of the audio as well. The percentage of the budget for post-production is now more in favor of the audio aspects. The bottom line is that the guy that has been recording stereo only to be used for mono reproduction is going to be leaned on by the producers more than ever. Say you're watching MTV and the artist's video looks and sounds great, and is in stereo; when an ad comes on following that video and that ad is in mono you have experienced a downer. Any effective advertising campaign is not going to let this happen. An audio room like this one is warranted in the business and we are seeing a trend that's not going to stop, yet we still have a lot of catching up to do. I'm an audio engineer that doesn't make records. I use an SSL, a 24-track machine, all this outboard gear, and it's all basically for television. There is a new frontier at hand and we are seeing the use of record technology applied to TV broadcast and production. Not only has my job become more interesting, but there are more opportunities now as well."

While listening to Larry, my attention was drawn to unusual multi-pin connectors on the wall. Larry explains, "All tech work and design is accomplished by in-house personnel. VCA is a truly self-contained facility where even repairs and maintenance are handled only by in-house people. Those connectors are VCA designed and built. They enable me to take the signal path of anything in this room and re-route it anywhere. I can re-wire this entire studio in less than half a minute. One of the most impressive pieces in this studio is our digital signal router. Every room is tied to every other room via this digital matrix routing switcher. I can call up any tape machine, any room, and monitor what's going on, or tie in directly and incorporate any location at VCA with this audio suite. We have 7 floors of studios, so, to get an idea of how much the digital router is handling, I'll give a brief rundown of the VCA sub-structure: We have edit rooms B, C, D, E, and F so that's 5 full-

Figure 3. The producer/engineer's view of the control room.

video machines. We have a Chyron 4 system in each room, 3-A.D.Os (special effects), and the Quantel special effects machine. We have a room called the Montage, an off-line system that utilizes 17 1/2-inch video machines. There are two Palette rooms that are our film-to-tape rooms. The concourse has mainly administrative offices; the second floor is the Montage, scheduling, operations managers and some offices; on the third floor we have three edit rooms and one of the Pallettes; the fourth floor has two edit rooms, one of the Pallettes, the Audio Room and the maintenance shop; fifth floor is the corporate offices; the sixth floor has a Paint Box, another edit room, an Animation stand as well as some offices; and the seventh floor has the sales offices and accounting. All of these locations show up at our routing system."

Larry showed me the room that houses the electronics for the Digital Router and it resembles the central routing facility for AT&T. Frank Lanzer, VCA's chief audio technician, expands, "The router matrix is 100 x 100. Every machine and processor in the house has input and output to the matrix, so, any machine can 'look at' any other machine." Larry adds, "Seven floors of equipment can be accessed in any configuration, for example; video from one machine, one channel of audio from another floor, code from a fourth machine, etc."

The most important aspect in a complex such as this one is the realization of a product that is consistent throughout the industry. In other words, the materials that are produced here must be visually and sonically consistent regardless of the playback and/or broadcast mediums. This notion can often turn out to be quite a dilemma. For example, if this studio uses references such as the UREI 813s and MDM time-aligned monitors, stereo processing, and numerous effects, isn't it possible that all that work can potentially be destroyed as the material is altered in order to fit within the boundaries of the commercial world? Furthermore, what is the reference one uses in order to anticipate what will happen to a given product as it goes through its many altered states in preparation for the broadcast and consumer markets? Frank comments, "The greatest challenge in this area of work is to have your products leave this nearly perfect environment and come across in the most effective manner possible to the outside world."

Larry explains, "The only true reference that can be used to anticipate changes that occur due to varying mediums and formats is your mind. Only experience can really tell you as to whether or not the piece will sound good when it leaves this environment. It really isn't all that complicated, however; there is no substitute for experience. When a producer comes in, one of the first things I ask him/her is, 'Where is this going to be played? Is this for a club? Is this an industrial product? Is this going to be broadcast on network TV? Is this going to be 3/4-inch? What is this going to be used for?' You don't mix on the 813s for something that is going to be broadcast over a 3-inch TV speaker. Conversely, you don't mix on Auratones if the product is going to be used in a high-tech video dance club." In light of the present state of the art (broadcast), one must take precautions in order to avoid the total degradation of a beautiful sound that was done in stereo so that it still retains its integrity as it makes its way to a mono format. Larry observes, "Any time you work in stereo that will later be broadcast, you have to keep listening to it in mono and make sure that it is mono compatible. Even if you do a project that has stereo broadcast potential, you have to bear in mind that 75 percent of the audience will be hearing it in mono."

The important projects almost always require a close communication network between engineers and producers. Larry says, "It helps to know the producers...what they like and dislike. They usually appreciate input, but it is even better if you know what they want. There is salesmanship intertwined with the technologies and the bottom line is that they must leave here happy. The work that is done in this room is not arts and crafts. This is a business and the meter is running. You've got to give them the best you can while maintaining the conformity factors which will establish you as a facility that not only gets the job done to spec, but can also expand and realize ideas that producers have in mind."

CAREER OPPORTUNITIES

As an added side to this story, we chose to investigate the possibilities of careers for young people in the audio/technical fields. We at **db Magazine** forsee a new type of growth that will be stimulated by the expansion of stereo audio in the broadcast and production fields. The following comments by Frank Lanzer and Larry Rosen might help the career-minded people gain some insight and stimulate interest in this particular area of work and related fields.

Frank comments, "My involvement in the industry has been an ongoing process which began around 1975 when I attended Thomas Edison Technical Vocational High School. It was there that I acquired a basic electronics background. My interest and involvement was keen outside of school. I'd pick up technical manuals and try to figure them out. By simply doing various side experiments on small, not very high-tech pieces, such as transistor radios, I would often discover or confirm different principles of electronics and their practical applications. This eventually led to bigger radios and consoles. If I would find myself with nothing to do, rather than reading a newspaper or Playboy, I would pick up an equipment manual or a db Magazine. That's where most of my background came from. I was building studios for Musac for a while before I came here to VCA which is where I've been for five years now. So, an individual must maintain a very high motivation level in order to enter into the various areas and levels of this industry. Anyone that wants to get started has to pay their dues. You can't expect to start at the top. Respect and recognition have to be earned. You might start in the library (sound library) pulling tapes. You might have to type labels or make dupes of elevator music. Even if you are doing a job that you don't want to do, don't wait around for somebody to come and teach you something that you might have to know in order to progress in the company. In your spare time, go and hang out in the departments that interest you, look and listen very carefully while remaining unobtrusive. Ask questions, but know when not to interfere with someone's work. For instance, Mary or Jon X has been typing labels all day. After work every day, she/he goes to one of the edit rooms and watches. Then one day someone can't show up to do a job. The answer could very well be to use Mary or Jon X on that particular session." Larry contends, "There are few instances where a person was hired for the position that they eventually wound up in. I was originally hired as a record tech in the library to do filing."

IN CONCLUSION

Larry concludes, "For years producers have been asking little things like 'more high end please' or 'a little less lows please.' But now it has got to sound as good as it can because someone out there is watching this program and listening to it on a very good hi-fi. The video industry realizes that the potential for audio's impact is greater than ever." Audio is no longer an afterthought in the television business. As a matter of fact, it is becoming so important that the pre- or post-video production houses are making every effort to maintain a "world-class" audio profile. db

VCA TELETRONICS AUDIO EQUIPMENT ROSTER

Console:

Solid State Logic SL6000E stereo video system, computer controlled, automated audio recording and mixing, with total recall, real-time, events, master transport selector, synchronizer controller and hard copy printer.

Tape Machines:

Studer A800 (interchangeable head stack)—2-inch 24-track and 2-inch 16-track at 30 and 15 in./sec. with dolby A noise reduction.

Studer A800 (interchangeable head stack)—1-inch 8-track or 1/2-inch 4-track or 1/2-inch 2-track at 30, 15 and 7.5 in./sec. with dolby noise reduction.

Studer A810—1/4-inch 2-track with center channel time code at 30, 15, 7.5 and 3.75 in./sec. with dolby noise reduction.

Studer A80VU (interchangeable head stack)—1/4-inch 2-track and 1/4-inch 4-track at 30 and 15 in./sec. with dolby noise reduction.

Studer A80RC—1/4-inch 2-track pilot and neopilot at 15 and 7.5 in./sec.

Studer A710—stereo cassette deck.

Synchronizers:

Adams-Smith SY2600 tape machine synchronizer serially interfaced to SSL6000E console.

Audiokinetics Q-lock 3.10 tape synchronizer.

Monitors:

UREI 813B control room monitors with MXR 31-band equalization powered by Bryston 4B amplifier.

MDM-4 nearfield monitors powered by Bryston 4B amplifier.

Yamaha NS-10M monitors powered by Yamaha PC1002 amplifier.

Auratone nearfield monitors powered by Yamaha P2200 amplifier.

Outboard Equipment:

Lexicon 224X with larc digital reverberator
Quantec QRS digital room simulator
Neve 33609 stereo limiter/compressors
dbx 160X stereo pair limiter/compressors
Dynaflex D2B stereo single ended noise reduction
UREI LA-4 stereo pair limiter/compressor
Teletronix LA-2A limiter/compressors
Eventide H949 Harmonizer
Lexicon prime time II digital delay and effect generator
UREI 537 stereo pair graphic equalizers
Sontec MEP-250C stereo parametric equalizer
Pultec MEQ-5 stereo pair mid-range equalizer
Pultec HLF HI/LO filter
UREI 565 peak/notch filter
EMT 140 stereo reverberation plate
Dolby 43A film processor
Scamp de-esser
Scamp ADT
Scamp dynamic noise filter, gate (2)
Scamp pan
Valley People Kepex II noise gates (4)
Technics S11800Mk2 quartz direct drive turntable with stanton 680SL cartridge
Tektronix 1420 NTSC Vectorscope
Tektronix 528 waveform monitor

Microphones:

AKG 451EB with CK5 Capsule, C12, C28
Neumann U67, U48A, M49B
Shure SM54
Beyer M69N (c)
Electro-Voice RE20
RCA 77-DX

Console Automation in South Florida

KEITH MORRISON

MUCH HAS BEEN WRITTEN ABOUT THE BURGEONING music industry in South Florida. Like Motown's R&B in the 60s and the jazz-flavored "west-coast" sound in the 70s, Afro-Cuban sounds and rhythms are making their mark on commercial music in the 80s. Supported by an active South Florida and New York dance club market and radio, artists such as Madonna, Miami Sound Machine, Lisa Lisa and Expose have been able to develop and crossover into mainstream pop. Dance music is essentially club music. Because of the high-energy, non-stop nature of the discotheque, DJs need extended intros and percussion breaks in addition to "radio" versions of songs. This need, teamed with the widespread use of drum machines and synths, creates a unique challenge to the recording engineer, who must mix a given song in at least two, and often in as many as four versions. Given the ever-present budget considerations and the fact that the mixes generally have to be done consecutively, efficiency and accuracy become increasingly important. Obviously, some sort of console automation is necessary to accomplish this.

The ideal system would be one that is SMPTE based, with MIDI song position pointer implemented, a user interface that is simple to operate and easy to learn, one with fast response time, tight synchronization and of course a low noise floor and good distortion figures.

The alternatives that an engineer has in South Florida are the Solid State Logic 4000E, Neve with Necam 96, MCI 500 Series with ARMS automation and the Sony 3000SL. Each of these is an on-board dedicated system and while each of these systems has its advantages, I have found a system which is more practical. It automates mutes and faders, has faster response time than any of the VCA-based systems, is simple to learn and operate, is portable and sounds great. All this for about the cost per channel of an analog gate. The system is called the XZ-100 and is manufactured by Kia Electronics of Miami, Florida.

Keith Morrison at his personal recording system. Note the computer screen at the rear of the console.

I first heard about the XZ-100 from a friend who invited me to his studio to meet the inventor, Odded Zamir. Mr. Zamir invited me to take a unit with me and try it, and, after using it for a short time, I realized it was equal or superior to many of the on-board systems I have used. The XZ-100 RSC (Recording Studio Computer) consists of a hardware interface, software supported by Commodore 64, and a MIDI interface. (Atari ST software is under development). The system interfaces to the existing console at the patchpoint or at the tape machine return/line input. The portability this affords makes it possible to take the unit to other studios which may not have console automation. The system is particularly useful in mixing dance music because of its method of programming.

Many of the producers in the dance market were at one time DJs and tend to think in terms of a rhythm-section groove, hooks, and lines, instead of a more traditional (ie., song-form) approach. They tend to record parts from the beginning to the end of the song and make their arranging decisions at mix-down time. The XZ-100 allows me to set up fader snap-shots of each successive section of the song off-line and then to assign them to SMPTE locations in real-time. The RSC lets me think in song form even if the parts aren't laid that way.

The program allows twenty-four snap-shots (or statuses) to be stored at any one time on board the computer for on-line recall. (Unlimited status storage is available on 5 1/4-inch floppy disc). What makes the XZ-100 different from VCA or moving-fader based systems is resolution and response time. To make a precise change between sections involving mute and level changes normally means mixing section by section to 2-track and razor-blade editing. Because the response time of the XZ-100 is about two milliseconds as oppposed to more than 40 milliseconds in any of the VCA-based systems, it is possible to mix in a single pass without editing. This makes it possible to use sonically superior, digital 2-track formats, which are difficult, costly or impossible to edit physically.

In sound quality, the XZ-100 rivals the more costly dedicated systems. Its noise floor of -126dB makes it quieter than most of the consoles I have used it with. Since it is not

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a VCA-based system, it has none of the coloration associated wih VCAs. Mr. Zamir calls the gain-control circuit a DCA (Digital Control Amplifier), a system of his own design, which does not use off-the-shelf parts. The system's distortion figures are also impressive, THD at 0.008 percent.

After setting the gain structure through the successive parts of the song, I add fader dynamics. This can be accomplished by either the QWERTY keyboard or the mouse. One of my first reservations about the XZ-100 was not having faders to control. In practice, however, it has turned out to be quite an advantage. The QWERTY keys are more accurate than a manual fader move, previous gain settings are precisely repeatable, and, much to our surprise, we find ourselves talking in terms of the keyboard (ie. No, T is too much, try V.) In addition, since the faders are left in some uniform nominal gain configuration (ie., 0dB) and remain functional, any "last-minute, inspirational" moves may be carried out manually.

When the song's gain structure is in shape, I add mute moves in sections where a channel is left open but no part is played. This is where any automation system can make a tremendous difference in the cleanliness of a record. Because of its unusually fast response time, and that its resolution to synch is based on the computer's internal clock, which is much higher (in resolution) than either MIDI clock or SMPTE, the XZ-100 is capable of much tighter mute moves than some of the mega-buck systems. I have found this especially useful in removing or, in some cases, emphasizing breaths in a vocal part. This job is too precise for many automation systems, forcing the engineer to use a gate or compressor which adds an effect he may not want through the entire song.

In the edit mode, the XZ-100 is similar in function to the Necam system. The "fader" positions are always current, for example, there is no need to go through a "fader null-point" search operation in order to make an update. It is possible to either over-write a section or to overdub moves on top of existing moves. Either of these methods may be executed one channel at a time or in groups of up to 128 channels in either step time or real time.

The program also supports automated fades from user selectable start and end points for single channels or groups, individual channel and group solo, SMPTE synchronization and on-board capacity for memorizing 12,000 moves per song (that's one level change per second on 32 tracks throughout a six-minute song.)

The XZ-100 is easy to use and does jobs that much more expensive RSCs don't do as well or in some cases, don't do at all. It makes a mix that used to have a lot of tedious, time-consuming elements more efficient, more accurate and most importantly, I think, more musical. db

DREW DANIELS

Condensed Glossary of Audio Terms—Part I

ABSORPTION
The ability of a material to absorb sound energy and reduce sound intensity by converting sound (vibration in air) to heat by means of friction in the material's structure (adiabatic heating).

ABSORPTION COEFFICIENT
The efficiency of a material to absorb sound at a particular frequency (which relates to sound wave length and material thickness). An absorption coefficient of 1.00 indicates total absorption, while a coefficient of 0.00 indicates total reflection. (see also, SABINE)

ACOUSTIC
Related to pressure changes or propagating mechanical waves in air or any other sound transmission medium, that comprise sound in its conventional form, as humans hear it.

ALNICO
An alloy of cobalt, nickel and aluminum used as permanent magnet material in magnetic structures of loudspeakers and microphones. In the early 1980s, alnico was largely supplanted in favor of ferrite in loudspeaker design because of political upheavals in the African countries that produce cobalt, the prime constituent of alnico.

AMBIENCE
The distinctive acoustical characteristic of a room or acoustic space due to the many sound reflections in the space. For example, rooms that are said to be acoustically "dead" lack ambience.

AMPERE
The ampere is that constant current which, if maintained in two straight parallel conductors of infinite length, of negligible circular cross section, and placed 1 meter apart in vacuum, would produce between these conductors a force equal to $2X10^{-7}$ (0.0000002) newton per meter of length.

AMPLIFICATION
An increase in signal quantity of either amplitude or power level.

AMPLIFIER
A device which increases the voltage and/or power level of signals fed through it.

AMPLITUDE
The extreme range of a fluctuating quantity, as an alternating current, swing of a pendulum, etc., generally measured from the average or mean to the extreme.

ATTACK
The beginning of a sound or the initial transient of a musical note.

ATTENUATE
Reduce. In audio parlance, to reduce the level of an electrical signal as with a volume control, pot (potentiometer), fader or pad.

AUDIO FREQUENCY
Any frequency which humans hear, typically between a lower limit of about 12 hertz and an upper limit of about 20 kilohertz. This range of audio frequencies is also known as the "audio spectrum."

AUTOFORMER – AUTOTRANSFORMER
A single-winding coil, often on a magnetic core, resembling a transformer in physical appearance. When used for audio, the autoformer is fed a high-level signal such as that from a loudspeaker line, and produces desired changes in voltage at one or more taps along the coil's length. These taps are usually spaced so as to produce specific impedance ratios between inputs and outputs. For example, a 1:2 autoformer connected to an 8-ohm loudspeaker will convert its impedance to either 4 ohms or 16 ohms, depending on which way the connections are made.

AXIS
An imaginary center point. Looking down the center of a horn places the viewer "on axis" to the horn, while moving to the side so that the horn throat is not visible places the viewer "off axis."

B

BAFFLE – ACOUSTIC

An absorptive board or sound barricade that can be placed around or between acoustic sources to provide sound isolation or deadening and reduce acoustic leakage between multiple microphones, such as in a recording studio or live musical performance stage setup.

BAFFLE – SPEAKER

The enclosure surface, wall boundary or mounting board on which loudspeaker drivers are mounted.

BALANCED – BALANCED LINE

(see FLOATING)

BAND

In terms of audio frequency, a band is a portion of the audio frequency spectrum in the same way that green is a portion of the visible frequency spectrum. The audio frequency spectrum covers a range of over 10 octaves. The visible light frequency spectrum covers a range of less than 1 octave.

BAND PASS

A set of two filters that attenuate frequencies beyond the frequency limits of a given band of frequencies. The telephone, for example, is a band pass filter that eliminates low frequencies below about 300 hertz and high frequencies above about 5 kilohertz, causing the characteristic telephone sound most people are familiar with.

BAUD

The rate or frequency of data bit or byte transmission in a data transmission line.

BUS or BUSS

Like a bus that may carry many passengers, an audio bus is a wire or circuit that may carry more than one audio signal at a time.

C

CAPACITOR

An electronic circuit component part designed to store electricity. The value of such a part in farads (F) is a measure of the amount of electricity that can be stored. A theoretically ideal capacitor with a one-farad capacity is charged, from a discharged state, to a voltage of one volt, by applying a current of one ampere for a period of one second (see COULOMB). Capacitors are made of two metal conductors separated by a non-conducting dielectric material such as paper, oil, glass, air, mylar, polypropylene, polystyrene, etc.

CARDIOID

Heart-shaped. Pronounced "car-dee-oid," in terms of microphones, refers to the relative sensitivity of the microphone with respect to the angle from which sound strikes the front (on-axis). Cardioid microphones decrease gradually in sensitivity as they are rotated away from the source of sound they are aimed at. Cardioids perform best if their off-axis frequency response is similar to their on-axis response.

CENTER FREQUENCY

The particular frequency at which the most boost or cut is available in a peak-dip type equalizer such as a graphic type, or a notch filter or parametric type.

CHANNEL

The individual audio signal path through a system which has more than one such path or as in the case of a single-channel amplifier a device which passes signals along only one electrical path.

CLIPPING

A distortion of audio signals caused by input signal peaks or voltage amplitudes which cause a circuit to attempt to exceed its own maximum voltage capabilities.

COMPRESSOR

An audio amplifier whose output amplification rate of change is less than its input signal amplitude rate of change. Compressors are used to reduce the dynamic range of program signal either to make everything sound louder, or to automatically control sudden large changes in signal amplitude as in the case of recording vocalists. Compressors sometimes include circuits that allow the user to adjust the time it takes to start compressing (attack), to ease up on the compression (release), and also the input and output gain. (see also, LIMITER)

COMPRESSION DRIVER

A loudspeaker designed specifically to drive a horn, matching the horn's acoustic impedance to achieve higher efficiency.

CONDENSER

(see CAPACITOR)

CORNER FREQUENCY

The frequency that defines the lower or upper limit of an audio frequency band, and where the power level is half of that in the middle of the band or "center frequency."

COULOMB

The coulomb is the quantity of electricity transported in 1 second by the current of 1 ampere.

CROSSOVER—ACTIVE, or ELECTRONIC

An electronic device which filters and selectively amplifies frequencies, separating the frequencies into sections or bands, and routing them to outputs designed to drive power amplifiers and in turn, speakers. The frequencies filtered depend on the electrical value of the component parts in the circuits of the device, but not on the source or load impedances connected to the device, except in the case where the crossover is actually a passive crossover designed for insertion in the medium-level signal lines of an audio system rather than in speaker lines.

CROSSOVER—PASSIVE, or HIGH-LEVEL

An electrical device composed of coils of wire (inductors) and electrical capacitors, that separates audio frequency bands by filtering action and routes them to different places (such as a woofer and a tweeter). The frequency of the crossover's action is determined by the value of the electronic components inside, and by the loudspeaker driver's impedance in ohms, which implies that replacing a 16-ohm driver in a particular system with an 8-ohm driver, will change the crossover frequency; in such a case, the frequency will rise an octave and the shape of the crossover frequency response slopes will be distorted.

CROSSTALK

The leakage between audio signal carrying channels, typically heard as bleed-over between left and right stereo speakers, or as leakage of high-frequency sound between busses or circuits in audio mixers, microphone cable snakes, and multiple circuit audio signal wiring. Crosstalk is often caused by the electrical coupling by capacitance between the metal traces on printed circuit boards or the proximity of conductors in mixer wiring harnesses.

CUE

Also called "foldback," cue is a portion of audio signal in a system which is diverted and used for pitch and tempo reference by musicians or for timing reference by voiceover announcers for jingle production and motion picture dialog replacement dubbing (as from monitor speakers or headphones). The term "cue" is also used to describe the circuits within an audio mixer unit or an audio system designed to provide this reference.

CUTOFF FREQUENCY

All audio systems are limited to a band of frequencies in which they can do useful work. The frequencies are defined as the corner frequencies of a filter. Since for example, an amplifier cannot reproduce infinitely high notes, it is a low-pass filter whose cutoff frequency is the point (in hertz) where it can no longer produce full-power output, and where the actual output power falls to half the midband power or 3 decibels below the reference full-power output at midband (-3 dB point).

D

DAMPING or DAMPING FACTOR

The difference or ratio of an amplifier's output impedance and the impedance of the driven load. For example, an amplifier whose output impedance is 0.8 ohm driving a speaker whose impedance is 8 ohms has a damping factor of 10, while an amplifier whose output impedance is 0.08 ohm driving an 8-ohm speaker gives a damping factor of 100. Inserting a speaker cable whose resistance is .08 ohms in series with an 8-ohm speaker and an amplifier with a .08-ohm output impedance lowers the overall system damping factor to 50 (8 divided by .16).

DECIBEL or dB

A comparison of two similar values, like apples vs. apples, oranges vs. oranges or volts vs. volts. A voltage doubling (or halving) produces a 6 dB increase (or decrease), and a power doubling (or halving) produces a 3 dB increase (or decrease). The amount of power increase required for us to hear a twice-as-loud increase is +10 dB. The amount of power decrease it takes for us to hear a half-as-loud decrease is -10 dB. Thus to produce sound twice as loud as that produced by a 100-watt amplifier would require a 1,000-watt amplifier.

The dB is a power ratio. Calculating dB for power is done by multiplying the difference between two numbers by 10 times the base-10 logarithm of the numerical ratio.

For example: 50 watts = $10\mathrm{Log}_{10}(50/1)$ = 16.99 dBW or 16.99 dB above one watt.

Quantities that are calculated using $10\log_{10}$ are:

Watts	Energy level
Illuminance	Intensity level
Power level	Energy density level

Quantities that are not power ratios must be calculated using $10\log_{20}$ as the multiplier. These include:

Volts	Vibratory acceleration
Amperes	Vibratory velocity
Sound pressure level	Vibratory force

VOLTS	dBV	dBu	WATTS	dBm	dBW
.02449	-32.2	-30	.0001	-10	-40
.03162	-30	-27.8	.001	0	-30
.07746	-22.2	-20	.002	3	-27
.1	-20	-17.8	.01	10	-20
.24495	-12.2	-10	.1	20	-10
.31623	-10	-7.8	1	30	0
.77459	-2.2	0	10	40	10
1.0	0	2.2	100	50	20
10.0	20	22.2	1000	60	30

There are several significant decibel variations used in audio:

dB - used alone as reference for level changes.
dBV - ratio of volts referred to one volt.
dBu - ratio of volts referred to 0.7746 volt.
dBm - ratio of watts referred to one milliwatt.
dBW - ratio of watts referred to one watt.
dB SPL - ratio of sound pressures referred to 20 micropascals.

NOTE: dBm should not be use to denote a voltage, since that implies that a specific load impedance is known. dBm improperly used where dBu should be used must, therefore, include a statement of circuit dependency on a 600-ohm load, since dBm and dBu are equal only if the 1 mW dBm reference is driving a 600-ohm load:
atts = volts^2/ohms, $\therefore$ 0.7746 volt^2 = 0.6/600 ohms = 0.001 watt

DECAY

The fading away of a musical note after its onset or attack. In acoustics, the time it takes for echoes and reverberation to fade away. The term "RT_{60}" is used to describe the reverberation time of a room or acoustical space under study when a period of time has elapsed after a calibrated noise excitation is stopped, until the reverberation in the room drops to a sound pressure level 60 dB below the reference level of the excitation. RT_{60} values of 5-10 seconds are typical of large cathedrals, RT60 between 1-5 seconds are typical of churches or gymnasiums and RT_{60} values between .1 and 1 second are typical of recording studios.

DIAPHRAGM

The moving part of a loudspeaker, particularly compression drivers and tweeters. The part of a loudspeaker that actually pushes on the air causing air motion.

DIFFRACTION

The phenomenon of sound waves bending around objects which are small compared to the length of the waves (see "wavelength"). Objects such as posts tend not to affect bass sounds but will shadow higher pitches (frequencies) to the extent that listeners will not hear tweeters that are not visible from their listening position.

DISPERSION

The directional pattern of sound radiation from a loudspeaker. The dispersion of horns is controlled by the horn's mouth walls, the overall size of the mouth and the length of sound waves emanating from the mouth. Low frequency loudspeakers normally radiate omnidirectionally at low frequencies, gradually forming beams of sound as frequency rises and sound wavelength becomes a smaller fraction of the loudspeaker's diameter. (see WAVELENGTH)

DISTORTION

An alteration in the shape, voltage, phase, timing relationships and frequency response of an audio signal caused either intentionally or unintentionally by circuitry that is driven to overload, or by poorly designed audio components such as microphones, mixers, effects, crossovers, amplifiers or speakers which do not accurately reproduce signals fed through them. (see OVERLOAD)

DIRECTIVITY

Directivity is a measure of the output of loudspeakers or horns based on the included angle within which the sound pressure level drops no more than 6 dB (one-quarter power). For example, a horn which covers a horizontal angle of 90 degrees (a quarter circle) where the two 45 degree off-axis points are 6 dB quieter than the on-axis measurement is said to have a (horizontal) "Q" of four, because it directs sound from what would have been an omnidirectional radiator (the horn's driver) into a quarter circle. Vertical directivity is derived in the same manner as is horizontal directivity, but the two figures are usually printed as two separate pieces of information on horn specification sheets since most horns radiate into different horizontal and vertical angles. A horn whose output covers angles of 90 degrees both horizontally and vertically, or one-quarter of a sphere, is said to have a total Q of 4, and a DI (Directivity Index) of 6 dB, since the same acoustical power from an omnidirectional radiator, forced to radiate into a quarter-sphere, is 6 dB louder at the same distance from the source than it would be radiating omnidirectionally, producing four times the apparent acoustical power to an observer such as a measurement microphone.

DIVIDING NETWORK

(see CROSSOVER)

DOPPLER EFFECT

For sound in air, the Doppler Effect takes the form of a shift in pitch which is proportional to the speed of any movement between a sound source and a listener such as the shift in the whistle on a passing train or the bells on a passing ice cream truck. In the same manner, a loudspeaker cone reproducing bass frequencies with their attendant long cone excursions will add a vibrato to any high-frequency tones being simultaneously reproduced by the same cone. The vibrato's rate will be that of the frequency of the lower reproduced pitch or pitches, and the vibrato depth will depend on the particular pitches that are interacting and the amplitude of low-frequency cone excursions. This vibrato is also called Doppler distortion, and is cited as one of a number of compelling arguments in favor of multi-way speaker systems.

DRIVER

Another name for loudspeaker; the word "driver" is used by non-engineers to designate a compression driver like those used to drive horns for acoustic amplification and directional control of sound.

DRY

An audio signal or sound without reverberation. An audio signal or sound with reverb is called "wet."

DUCT or **DUCTED PORT**

A tube attached to a speaker enclosure to "tune" and define the lowest usable frequencies of the enclosure. Like a bottleneck, a duct produces one distinct tuned pitch determined by its size relative to enclosure size. Such tuning is virtually independent of the bass driver mounted in the box, but grossly affects performance both in terms of frequency response and distortion.

DYNAMIC RANGE

The difference, in decibels, between the loudest and the quietest passages in a musical or audio program. Also, the difference between the maximum signal level that can be produced under nominal operating distortion levels by an electronic circuit, and that circuit's obnoxious noise level (called the "noise floor").

DYNE (per square centimeter)

An obsolete term used to designate 0.1 pascal, or 74 dB SPL (Sound Pressure Level). Also a unit of pressure equal to 0.1 newton per square meter. (see SPL chart on last page)

E

EARTH

(see GROUND)

ECHO

Any or all audibly discrete delayed sound images. In contrast, reverberation produces a wash of sound, with no discrete echoes.

ECHO BUSS

A typically dedicated audio channel within an audio mixing console, through which is routed signals intended to be sent or received to or from an echo or reverberation device such as an echo chamber.

EDDY CURRENT

Electrical currents caused in electrical conductors (metals) by the presence of magnetic field variations. These eddy currents in turn cause local magnetic fields which act counter to the fields producing them. Most electric power meters are eddy current motors which rotate in direct proportion to the amount of current (amperes) flowing through them. Loudspeakers and transformers are designed to avoid or take advantage of eddy currents to enhance performance.

EFFECTS

Effects devices can be broadly classified as anything that changes the sound of signals passing through them. In this sense, a distorted amplifier is an effects device, although effects are usually thought of as the product of one of the following:

limiter	filter	compressor
expander	equalizer	graphic EQ
noise gate	parametric EQ	tone control
VCO	VCA	envelope filter
envelope generator	echo	reverb
digital delay	digital reverb/echo	phaser
flanger	exciter	de-esser
stresser	parametric limiter	direct box
preamplifier	octave divider	vocoder
boom-box		

EFFICIENCY

Generally, efficiency is the ratio of input and output. Efficiency is usually expressed in percent, thus a loudspeaker which produces 8 acoustic watts when fed 100 electrical watts is 8% efficient, this would represent quite a high efficiency for a cone type loudspeaker. Typical hi-fi speakers and studio monitors range between 0.01 percent and 2 percent efficiency in their ability to convert electrical watts to acoustical watts. Power amplifiers give typically 50 to 98 percent efficiency, converting 60 hertz A.C. line power into audio frequency A.C. power.

EIGHTH SPACE

One eighth of a sphere. An acoustic boundary condition where the corner of a room causes low-frequency radiation from a speaker to be folded onto itself three times; once from the floor and once from each wall, producing a 9 dB increase in sound pressure over what the source would measure if hung in free space away from reflecting surfaces.

ELECTRET

A permanently electrically polarized microphone diaphragm used in place of an external high voltage supply to allow condenser microphone operation by the variable capacitor method.

ELECTROMAGNET

A magnet formed by the presence of electrical current in a coil of wire. A loudspeaker's voice coil is an electromagnet which alternately attracts and repels the permanent magnet in which it is situated, in response to the alternating electrical input from a power amplifier.

ELECTRONIC CROSSOVER

(see CROSSOVER—ACTIVE, or ELECTRONIC)

ENCODE—ENCODED

Alteration of audio signals prior to recording on tape, discs or other recording media. The alteration usually consists of pre-equalizing the incoming audio signals so that media noise is unaltered but signals on the media contain more high frequency energy, and often compressing the incoming audio signal so that less dynamic range is required of the media to store the audio signals. Decoding is normally the exact reverse of the encode functions, allowing signals to be re-expanded by a greater amount than normal expansion of the intrinsic playback noise of the recording medium.

ENVELOPE

The trend of waveforms that forms a composite waveform that may contain all the frequencies and signal components, sidebands and interactions of the signals in the envelope.

EQUALIZATION or EQ

The intentional alteration of levels of portions of the audio frequency spectrum to fit the requirements of frequency response defined by a listener. Traditionally the term equalization was used to describe the replacement (always a boost) of energy lost as a result of long telephone line runs of wire, but today the term is used to describe any change in frequency response or spectral balance done intentionally by using any device which includes circuits that can produce these changes.

EQUALIZER

An electronic circuit or device that selectively increases or decreases gain as a function of frequency. An equalizer may boost or cut only, or may do both. It may be a fixed circuit such as the equalizer in a phonograph preamp that restores the frequency response of a phono cartridge's output to flat from the record's normal non-flat output, or the equalizer may be a sophisticated self-contained device that allows user adjustment of frequency selection or continuous frequency tuning, bandwidth or Q and amount of boost or cut (parametric equalizer).

ERASE HEAD

A magnetic tape head used to remove recorded signals from tape using a high-level, high frequency bias signal that is turned on when a tape recorder's record circuits are active.

EXPANDER

An electronic device that makes loud signals louder and quiet signals quieter, thus expanding the dynamic range of the original signals.

F

FADER

An electronic component such as a potentiometer, or a circuit such as a voltage-controlled amplifier, that varies the amplitude of all the audio signals passing through it. Faders can be physically linked to the user's control by straight line knobs as with linear faders, rotary knobs such as those on trim and monitor controls or by means of computer and digital-to-analog converters that supply the necessary control voltage to operate the voltage-controlled amplifier circuit comprising a VCA fader.

FARAD

The farad is the capacitance of a capacitor between the plates of which there appears a difference of potential of 1 volt when it is charged by a quantity of electricity equal to 1 coulomb.

FEEDBACK

A portion of a signal which is fed into the audio signal chain or signal-carrying circuits, either in-phase or out-of-phase with the main portion of the signal, causing a reduction or increase of signal level in the system or circuits. In acoustic situations with microphones and speakers near each other, in-phase or "positive" feedback causes the familiar howling sometimes heard when too much system gain leads to recirculating sound build-up between mic and speaker. In electronic situations such as amplifiers, out-of-phase or "negative" feedback is put to use in the amplifier's circuits to reduce distortion, and lower output impedances.

FERRITE

A mixture of ceramics, iron powders or oxides, barium or strontium carbonate or other elements such as rare earths, which is cast and sintered (heated) and used as magnetic material to make permanent magnets or transformer or inductor cores. Ferrite magnets are also known as "ceramic magnets."

FET

Field Effect Transistor. A special type of transistor noted for its very high input impedance and linear operation, as compared to common bipolar transistor types which have lower input impedances and require higher bias currents to operate. Field Effect Transistors exhibit some of the operating characteristics of vacuum tubes which suits them for applications where tubes may have been favored over bipolar transistors.

FIDELITY

As with the common definition of fidelity, true to (the original), the term is used to describe the accuracy of the reproduction of audio signals by audio devices and components usually as the sound ultimately heard from the sound system by the listener.

FIGURE EIGHT

The sensitivity vs. direction or angle pattern of a bipolar microphone or loudspeaker, as described on a rotating graphic level recorder chart by a pen responding to changes in level caused by the rotation of the device past a stationary sound source, or in the case of the bipolar speaker, a stationary measuring microphone.

FILTER

A circuit that selectively attenuates portions of the audio frequency spectrum. A filter is the opposite of the traditional equalizer, which selectively boosts, but for the purposes of modern convenient control of sound on mixers and equalizer units, the circuits of tone-altering controls usually incorporate the dual abilities to equalize and filter by simply rotating a knob one way or another.

FLAT (Frequency Response)

The common term used to denote circuits, devices or audio systems that pass signals of different frequencies with equal amplitude over some range of frequencies (typically 20 hertz to 20 kHz for the audible range).

FLETCHER-MUNSON CURVES

One of several published sets of curves that graphically show how our ears perceive equal loudness for changes in frequency and for changes in sound level. Our ears are not flat and do not hear in a linear manner. At the normal threshold of hearing for humans (0 dB SPL or Sound Pressure Level), it takes some 10,000 times more acoustic power to enable us to hear a 20 hertz pure tone than it does to hear a 4000 hertz tone, while at 90 dB SPL, it requires only about twelve times more power to achieve the same relative perception of volume.

FLOATING

A circuit which passes signals without reference to a ground. Typically, a floating audio circuit is characterized by a three-wire configuration called a "balanced line" where two wires carry the audio information and one wire acts as an electrostatic shield. The two shielded conductors of such a cable both carry equal voltage potentials of opposite polarity, from their driving source to their driven input, and so share a balanced voltage with respect to a neutral or imaginary reference. Electromagnetic radiation striking both signal carrying conductors at once is canceled by the input of the device being fed by virtue of the fact that the input circuit responds only to the differential voltage of the two signal carrying conductors, and the electromagnetic interference appears equally on both conductors, producing no differential voltage at the input. The reference wire or shield, may or may not be grounded, depending on ground loop currents that may be amplified causing hum in the system. Often, grounding is accomplished by mechanical connection of audio component chassis within a metal rack enclosure, in which case shielded wiring is unnecessary for balanced inputs and outputs.

Early telephone technology used transformer balanced inputs where a center tap of the transformer winding was grounded to dump electrostatic potentials. This type of wiring used an actual ground connection as the zero- voltage reference against which the two signal carrying conductors were balanced, thus enabling use of simple twisted pair, unshielded conductors for transmission of signals over long land lines where shielded cable would have proved prohibitively expensive but immunity to radio interference was required.

FLUTTER

Output amplitude variations from an audio reproducer such as a tape or record player due to one of several types of mechanically-based problems. Flutter may consist of simple amplitude modulations (AM) in output caused by rough tape handling or out-of-round idlers, or may take the form of frequency modulation (FM), small pitch variations, from bent or unevenly machined capstans or drive motors, pulleys or belts. AM components of flutter may also include tape modulation noise caused by uneven magnetic coatings or amplitude variations caused by loose magnetic oxide particles preventing good tape-to- head contact. FM components of flutter may also include scrape noise from tape-to-head, tape-to-tape guide or tape to flutter idler contact. Flutter is usually thought of as rapid variations of 10 hertz or more, and in fact, FM flutter components often extend up into the upper frequencies of the audio range. Very low frequency phenomena of a similar nature are called "wow," and are characterized by the 0.56 hertz pitch variations of 33 revolution/minute records with off-center spindle holes.

FOLDBACK

(see CUE)

(Continued and completed in our next issue.)

LEN FELDMAN

Lab Report

Fostex E-2 2-Track Recorder/Reproducer

GENERAL INFORMATION

The Fostex E-2 and E-22 mastering recorder/reproducers are similar in all operating respects and in their control features, except that while the E-2, that we tested for this report, uses 1/4-inch tape and operates at 15 and 7-1/2 in./sec., the Model E-22 employs 1/2-inch tape and operates at 15 in./sec. and 30 in./sec.. Both machines are basically 2-track recorders with a center "Cue" track provided for time-code recording and playback.

The E-2 is equipped with three in-line heads (record, record/play, and erase) and that, of course, means that overdubbing while listening to one previously recorded track is easily accomplished. A feature that many small studios will undoubtedly find useful is the wide-range pitch control that lets you alter pitch by at least 15 percent in either direction. (Our sample had as much as 17 percent pitch adjustment capability in either direction.) Punch-in/punch-out is easily accomplished and what's more, the output channel that's put in the "record ready" state for punch in and the VU meter indications are alternately switched between input signal and playback signal each time the RECORD button only is depressed. That allows free access to tapecueing and rehearsal monitoring for musicians during punch in.

In addition to the Punch-in/Punch-out facility provided on the front panel, a remote punch-in/punch out jack is provided on the rear panel of the E-2 and Fostex has available an optional foot switch that toggles punch-in and punch-out functions. The E-2 handles reel sizes up to 10-1/2 inches. A five digit electronic tape counter is provided and it reads in hours, minutes and seconds. If you touch the pitch control, the display mode immediately changes to read the new tape speed and the percentage deviation from normal tape speed. Underneath the tape head cover, there is a cue lever that allows tape cueing when the tape is lifted away from the heads. The usual tape sensor puts the transport into stop mode whenever tape is completely wound onto the takeup or supply reel, if transparent tape passes the sensor or if tape is accidentally broken.

The E-2 has been optimally calibrated for Ampex #456 tape. However, if you want to use another tape, there's a second bias setting that you can adjust yourself to work optimally with the tape of your choice. Instructions for properly adjusting bias for that second tape setting are included in the brief, but adequate owner's manual that comes with the E-2.

Although we didn't try this during our extended use of the E-2, it is possible to remove the entire meter section of the unit and to remount it in a special Meter Bridge so that it is more easily viewed if you decide to install the E-2 itself in a horizontal rather than a vertical position. Instructions for making this change are also clearly described in the owner's manual.

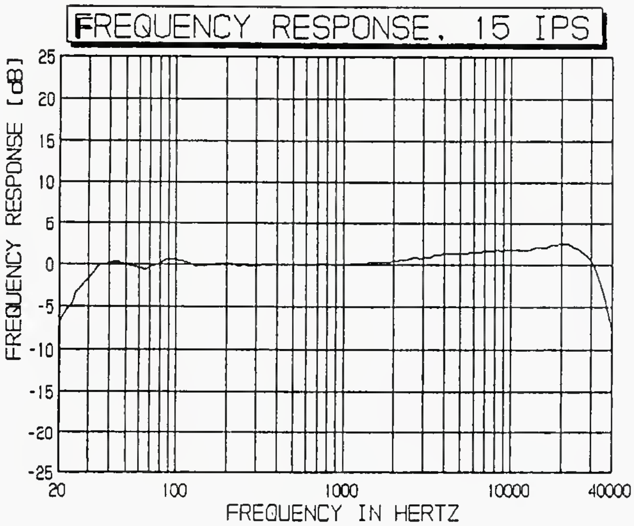


Figure 1A.

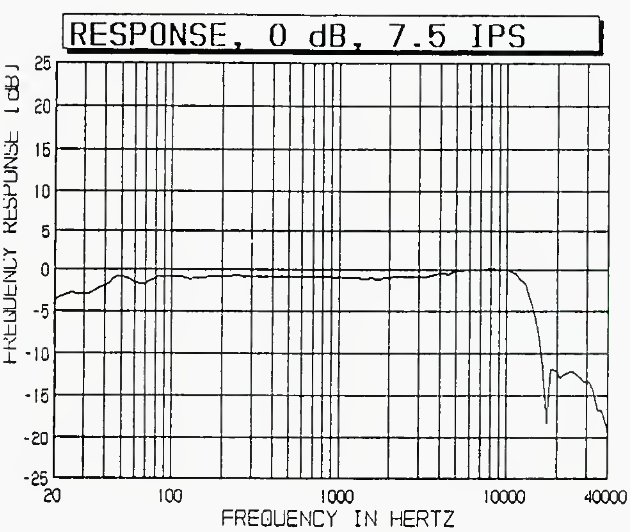
The E-2 is provided with XLR input and output connectors that can be used in either balanced or unbalanced modes. In addition, RCA type phono jacks are also available, but these are wired in the unbalanced mode.

CONTROL LAYOUT

The upper section of the E-2 is pretty much what you'd expect from any 2-track master recorder. Reels mount near the top of the unit and the tape path is pretty conventional. The power on/off switch is beneath the left-hand reel as are the tape speed selector and indicator LED and the meter selector, its indicator and a "Bias 2" indicator that illuminates when the alternate tape bias position is selected. The three illuminated VU meters supplied on this recorder (the third meter is for the time-code "cue" track) are also offset to the left side of the recorder housing. Beneath the meters are the input level controls and the output level controls.

Beneath the right-hand reel are the pitch control and switch and below them are the counter display, record track selector switches, a "Dump Edit" switch, monitor

Figure 1B.



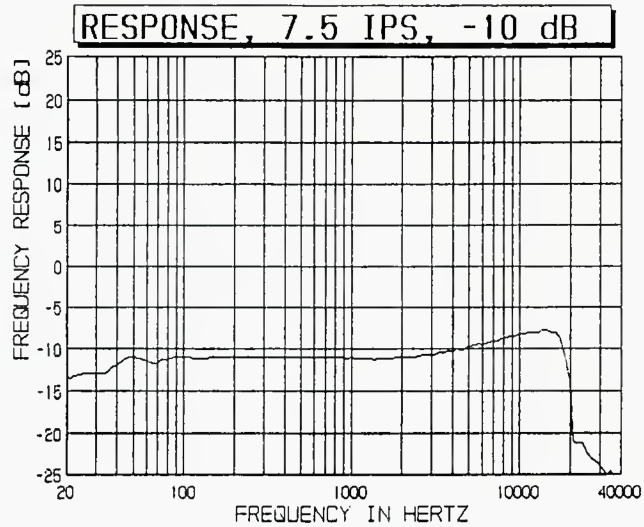
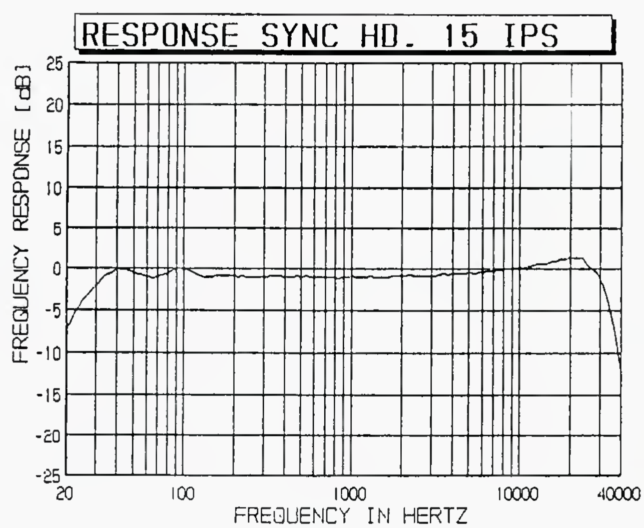


Figure 1C.

selector switches, a counter reset button, a "Locate 0" button, two memory reset buttons, an "Auto Play" button and LED, an "Auto Return" button and its LED, the usual tape transport touch buttons, the "Record" button and the record indicator. The "Auto Play" button, if depressed, causes the transport to enter the "Play" mode after coming to a stop during one of the memory rewind options. The "Auto Return" button, if depressed, will cause the tape to return to the "Memory 1" location on the tape after reaching the "Memory 2" position.

In addition to the XLR and RCA jacks found on the rear panel of the E-2, there are two foot-switch connection jacks. The first of these is for connection of a "Punch-in/Punch-out" foot switch already mentioned. The other jack allows use of a foot switch to put the tape transport into the "Play" mode. Another accessory connector is available for connecting a Fostex Model 4030 "Auto Locator," while still another connector is for connecting a Fostex Synchronizer, using appropriate optional cables available from Fostex. Finally, there's an external unit connector that's used to interconnect between the E-2 and the

Figure 2A.



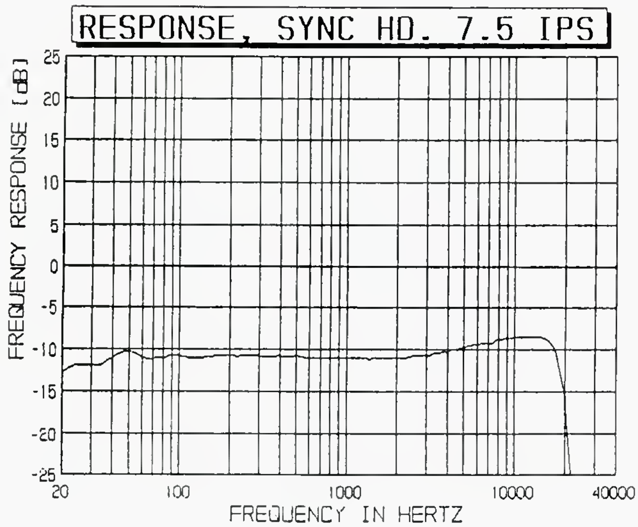


Figure 2B.

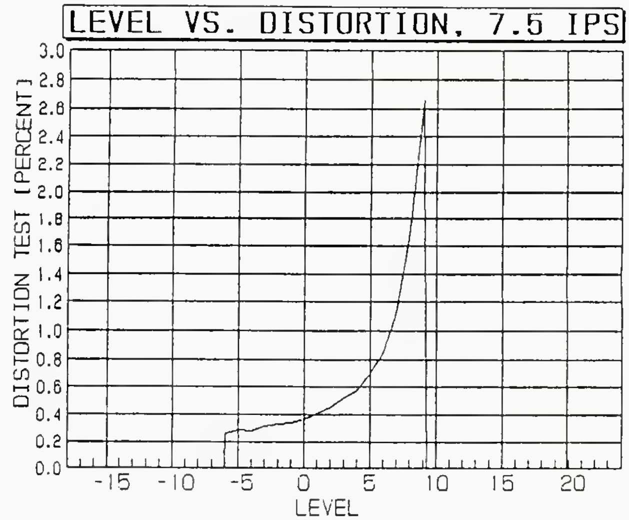


Figure 3B.

Meter Bridge when the meter panel section of the E-2 is mounted externally, as mentioned earlier.

LABORATORY MEASUREMENTS

While much of the data that we amassed in our lab tests of the E-2 is summarized in the VITAL STATISTICS chart at the end of this report, you will gain a better and more complete understanding of just how well this excellent mastering deck performed by consulting the sixteen graphic plots generated by our computerized tape equipment test setup that links our PC to our Sound Technology 1500A Tester.

Figures 1A, 1B and 1C depict the record/play frequency response of the E-2 under various test conditions. Note that *Figure 1B* was plotted using a nominal input of 0 VU (+4 dBm). At the slow 7-1/2 in./sec. speed, high frequency tape saturation occurs, so that response extended only to approximately 13 kHz. A more realistic way to check response at this slow speed is to reduce input levels to -10 dB VU and, at that lower recording level, response, as plotted in *Figure 1C*, extended all the way out to 19 kHz. We checked the record/play response using the sync mode as well. Normally, response tends to fall off a bit when the record heads are used for sync playback, but that was not the case for the Fostex E-2, as you can see by examining *Figures 2A* and *2B*. Once again, for purposes of fair comparison, the plot for 7-1/2 in./sec. operation was made at a -10 dB VU level.

Third order distortion *versus* record level was plotted for both 15 in./sec. operation (*Figure 3A*) and 7-1/2 in./sec. operation (*Figure 3B*). In both cases, distortion was well below the 1 percent value for 0 dB VU record level specified by the manufacturer.

Unweighted signal-to-noise ratio measured 68 dB at the higher tape speed; 70 dB at the 7-1/2 in./sec. speed. Weighted S/N was 1 to 2 dB better than that. Notice, that the plots shown in *Figures 4A* and *4B* represent the one-third-octave noise spectra from 20 Hz to 20,000 Hz, rather than the overall, or composite signal-to-noise ratios. As you might expect, major noise contributions occur at the high frequency end of the spectrum.

Figure 3A.

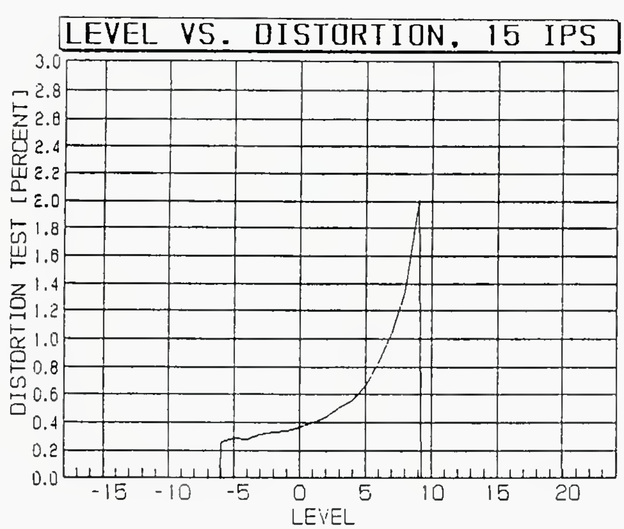
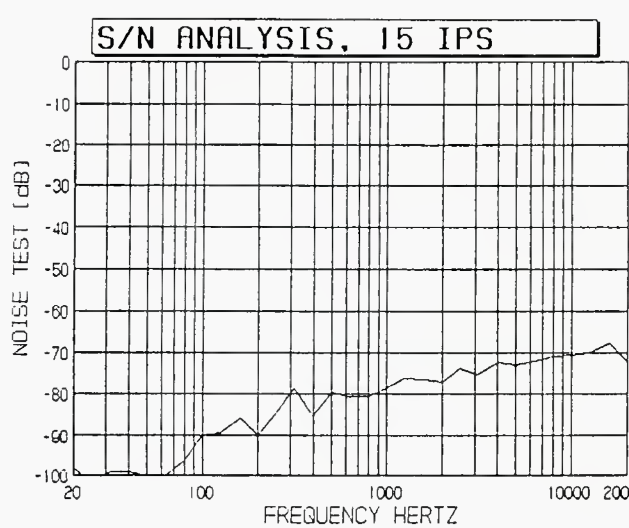


Figure 4A.



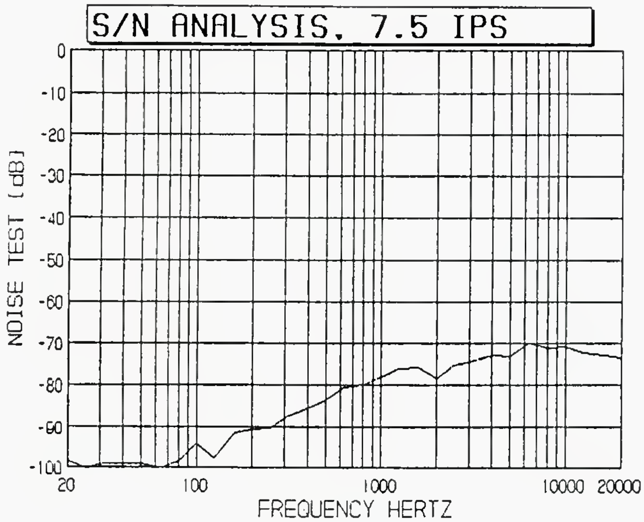
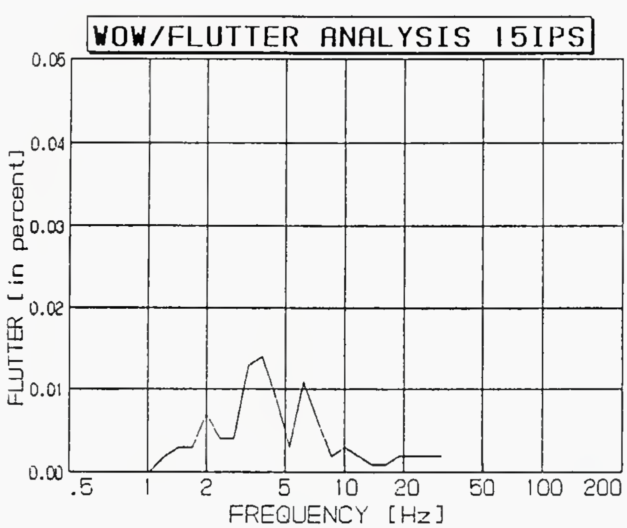


Figure 4B.

Wow-and-flutter for our sample was far lower than the nominal value specified by Fostex. We measured 0.024 percent WRMS at 15 in./sec., and 0.037 percent WRMS at 7-1/2 in./sec.. Unweighted peak figures were, of course somewhat higher. The plots in *Figures 5A* and *5B* show that major wow contributions occurred in the frequency region between 2 and 5 Hz.

Linearity at high frequencies, sometimes referred to as Maximum Output Level, or MOL, is a function of both the tape used and the tape recorder on which it is used. Normally, this test is conducted using an input frequency of 10 kHz. If the system were perfectly linear, we would see a straight line at or near the 0 dB calibration point on the vertical axis of the graphs plotted in *Figures 6A* and *6B*. In fact, that is almost precisely what is shown in *Figure 6A*, plotted for 15 in./sec. operation. In other words, a 10 kHz input of increasing amplitude resulted in a 10 kHz output during playback that was linear at least up to +9 dB or so. For 7-1/2 in./sec. operation, the 10 kHz test signal remained linear during playback up to about +1 dB VU, as shown in *Figure 6B*.

Figure 5A.



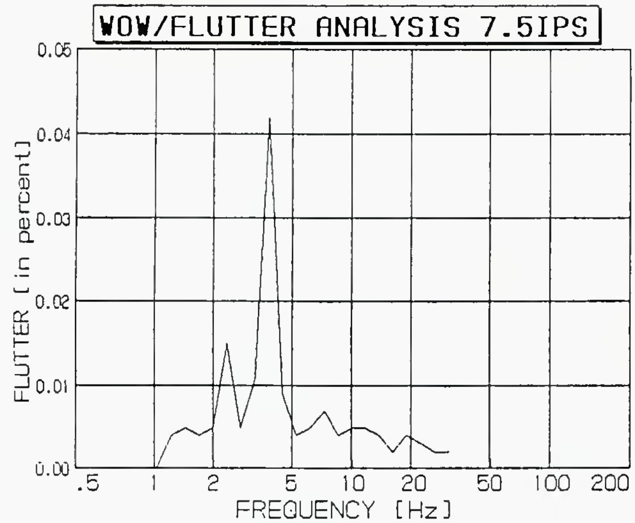


Figure 5B.

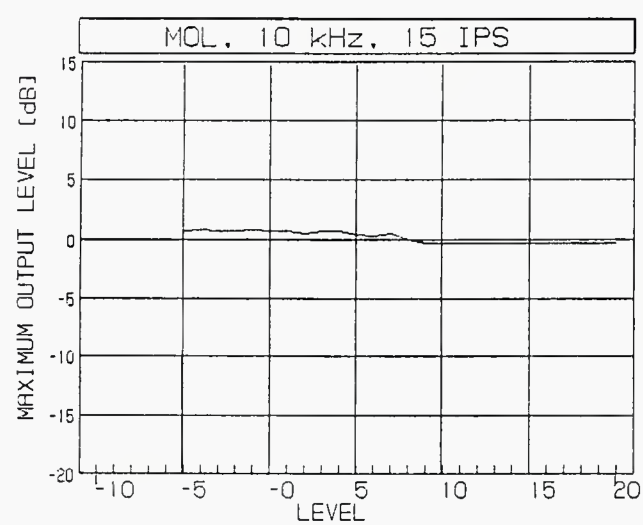
Separation, at the 15 in./sec. speed, is plotted in *Figure 7A*. At 1 kHz, separation between left and right channels measured 75.5 dB, while at the 7-1/2 in./sec. tape speed, (plotted in *Figure 7B*) separation was virtually the same at mid and high frequencies but was somewhat higher at low frequencies.

Azimuth alignment of the record and playback heads was superb, as illustrated by the azimuth error plots of *Figure 8*. These plots represent the average of 15 separate measurements at the four frequencies shown. The azimuth phase error plotted along the vertical axis represents the angular error or phase shift of the frequencies shown, and not of the tape heads with respect to each other. Thus, what appears to be a 30-degree error at the 5700 Hz test frequency in *Figure 8* is, in fact, only a 30 degree displacement of a single sinewave at that frequency—a negligible amount when translated to actual tape head azimuth error.

COMMENTS

We were favorably impressed by the features of this well-

Figure 6A.



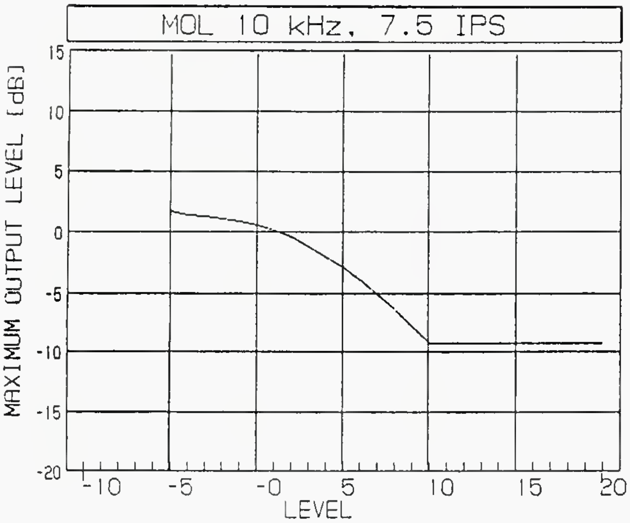


Figure 6B.

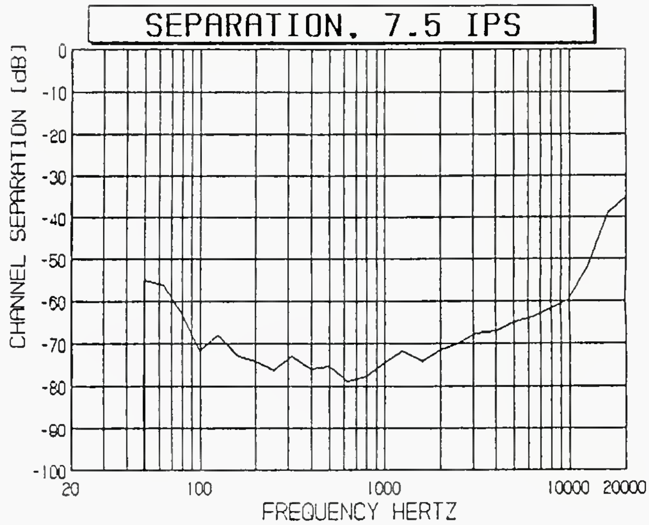


Figure 7B.

designed deck as well as by the logical placement of its controls and its smooth tape handling capabilities. The cue lever is arranged so that fast speed searching as well as tape rocking are easily accomplished. While our lab was not equipped to add a cue SMPTE Time Code track to the experimental recordings we made, we did check out the "Cue Track" input and metering system and feel that they will be a great aid in small studio applications and in mix-down operations from multi-track masters, editing, etc.

With all the talk of digital mastering, professional DAT machines and consumer DAT machines, we tend to forget sometimes that analog tape recording technology is better than 40 years old. During that time, the state of the art has advanced mightily, and the folks at Fostex have incorporated much of what is superior in a two-track reel-to-reel recorder/reproducer.

Whether you select the E-2 and settle for 1/4-inch tape mastering or choose the similarly configured E-22 for 1/2-inch tape mastering with what is undoubtedly its somewhat better dynamic range and signal-to-noise ratio capability, it is our feeling that Fostex has designed and produced a reliable, full-featured deck ideally suited for small-studio use. db

Figure 7A.

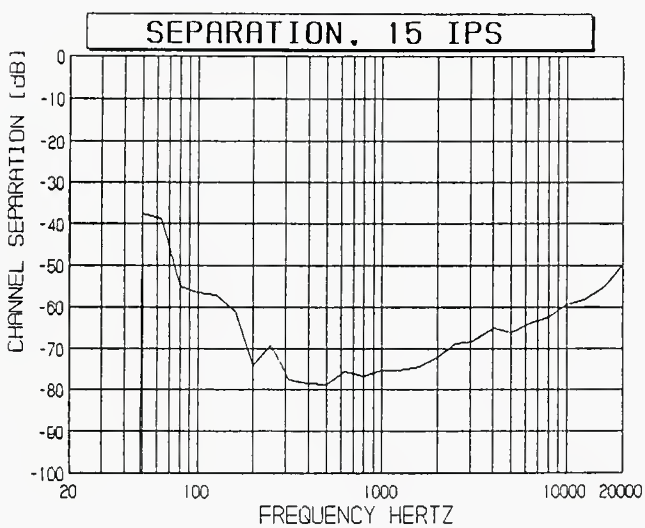
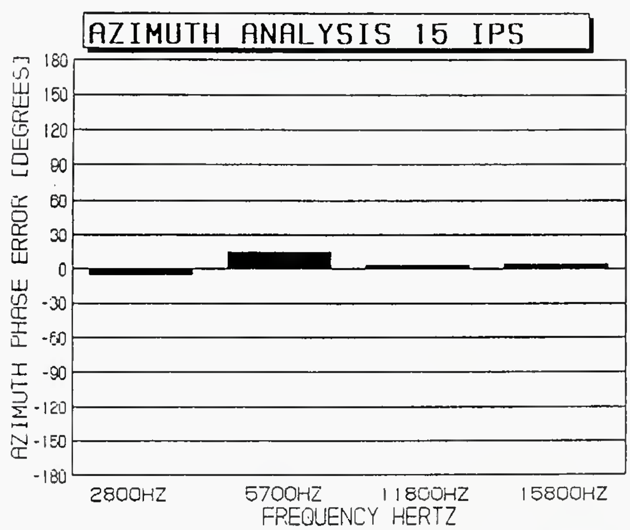


Figure 8.



(Continued on page 71)

MICROPHONE ACCESSORIES—CONNECTORS AND CABLES

CANARE CABLE, INC.

Audio connectors, microphone cables:

L4E Series cable is designed for use with mics but is also excellent for line-level signals. Its Star Quad configuration plus high shield density reduce hum and noise to less than 10 percent than that of conventional 2-conductor mic cables. Also offered are cables with multi-channel (in any configuration) braided shield and individual channel jackets, 2-conductor with braided copper shields and common drain wire, and 4-conductor with aluminum tape (foil) shield and individual channel jackets. Also offered is a complete line of gold and nickel plated audio connectors from $2.51 to $3.13 each.

CONNECTRONICS CORPORATION

Wire and cable, audio connectors:

Musiflex mic cable is ultra flexible PVC jacketed cable in 11 different colors and is 100 percent shielded using conductive thermoplastic (a drain wire enables termination). Two polyethylene insulated 22 AWG conductors comprising 30X36 AWG plain copper wire strands.

Studiflex shielded insulation cable is ultra flexible PVC jacketed balanced installation cable in 8 colors and is 100 percent shielded using conductive thermoplastic. Two 24 AWG conductors comprising 7X32 AWG strands of TCW.

Starquad mic cable, comprised of 2 spiral shields with opposing spirals laid one on top of the other, is used in areas which have an unusually high noise environment.

Rockflex instrument cable has a flexible PVC jacket covering 100 percent shield of conductive thermoplastic. The center conductor is 22 AWG comprising 19X34 AWG TCW strands, and is surrounded by a foam dielectric which ensures low capacitance.

Phonoflex is similar to Rockflex but with a smaller outside diameter.

Speakerflex loudspeaker cable is a PVC jacketed cable and is available in 2x15, 2x13, and 4x13 AWG configurations. Multipair foil shielded cable has pairs of individually shielded (with aluminum) conductors wrapped with a polyester film: this ensures extra flexibility and discretion of the grounds. Also offered are the UX high power audio connectors that are custom built loudspeaker connectors designed with 2 contacts and have no male or female parts but each connector is hermaphroditic and will mate with any other UX connector.

FOUR DESIGNS CO.

Audio connectors, adapters:

The RTS-321 is an audio interface that interconnects various pieces of equipment that use either 1/4-inch (ring, tip, sleeve) or XLR type connectors. RTS-321 eliminates the need for numerous adapters or special cables. Instead, five miniature slide switches allow instant user selection of several internal wiring schemes to suit each application.

Price: $29.95.

FURMAN SOUND, INC.

Patch bays, patch cords:

PB-40 series are 40-point patch bay mounts in a single rack space. Available in three connector options, 1/4-inch phone, 1/4-inch ring-tip-sleeve, or RCA front or rear, or any custom combination to meet special needs. Each vertical pair (four connectors, two front and two rear) is on a separate PC board supplied with removable normalling jumper(s). Price ranges from $145.00 to $160.00. Also offered are highly flexible, synthetic rubber jacketed patch cords 75cm in length. Available in 1/4-inch phone, 1/4-inch tip-ring-sleeve, and RCA.

Price: $15.00, $20.00, and $15.00 respectively per set of ten.

GOTHAM AUDIO CORPORATION *See our ad on page 24*

Audio cable, audio snakes, cable ties, mic cables:

GAC 3/1 is 3 conductor audio cable with a PVC outer jacket, 25 AWG. The strands are 96x44 AWG with a diameter of 0.49mm. The shielding is double Reusen. Available in 7 colors the prices range from $175.00 (492 feet) to $320.00 (986 feet).

GAC 4/1, 4 conductor audio cable has a PVC outer jacket. The conductor has a gauge of 25 AWG, strands consisting of 96x44 AWG, and a diameter of 0.49mm. The shielding is double Reusen and is available in black only.

Price: $425.00 (986 feet).

IC-3/10, IC-3/25, IC-3/50, IC-3/100 are made with GAC 3/1 audio cable, have integral loss proof cable tie, and are available in 7 colors.

Price: $15.00, $20.00, $30.00, and $50.00 respectively.

MARSHALL ELECTRONICS, INC.

Wire, audio cable, audio connectors, patch cords:

Mogami Bantam patch cords (tip, ring, sleeve) are super flexible and utilize an annealed high conductivity oxygen-free copper in a balanced quad configuration. With nickel plated connectors.

Price: from $14.95.

Mogami Puroflex interconnect cables (high definition molded audio cables) are available in various formats and lengths.

Price: from $4.50.

Mogami Neglex OFC multi cable consists of multiple twisted pairs with cross-link polyethylene insulation to prevent shrink-back during soldering and has super flexible PVC jacket.

Mogami Neglex OFC quad mic cables have multiple strands of high definition oxygen-free copper with matching served shield in a super flex PVC jacket.

Mogami Multi series professional speaker cables have conductors consisting of 224-strands of oxygen-free copper, a thin black PVC outer jacket and color coded PVC insulation.

Mogami Neglex X series multi-cable is similar to the Neglex series but with a unique color-coded/number-coded pair system, each pair with drain wire.

Econector heavy-duty audiophile connectors and Econector Jr. have gold plated contacts and Teflon insulation.

Sound Runner oxygen-free copper speaker cables are 8-AWG 2-conductor and 13-AWG OFC quad with round PVC jacket, and come in black, clear, beige and brown.

MONSTER CABLE *See our ad on page 13*

Mic cables, instrument cables, studio/reinforcement cable:

Prolink Microphone series cables have 3 phase aligned wire networks (gives extreme accuracy in areas of imaging, depth and maintains excellent frequency response and wide dynamic range), MicroFiber dielectric (to reduce signal loss, intertransient noise and quickens transient response), double shielding (reduces RFI and EMI).

Prolink tube mic cables have 3 phase aligned audio wire networks, 4 additional conductors (to power, preheat and polarize microphone tubes).

Prolink series multi-pair snakes have 4 pair balanced and individually shielded mic lines, 3 phase aligned wire networks.

Prolink high performance patchbay cables have 2 phase aligned wire networks, inner conductors individually wrapped with MicroFiber, 4 color coding, ultra flexible.

High performance professional speaker cables have primary and secondary grouped networks. Pre-terminated pairs are packed in carrying case with an owner's manual.

XP Pro speaker cable has precision wound wire network around a magnetic flux tube (to increase current handling ability), small and flexible characteristics.

NEUTRIK

Audio connectors:

NP2C and NP3C plugs are solid brass, nickel plated. Plug consists of 4 parts, no screws, no crimping, rear mountable insert, all metal lathed one-piece plug insert available, NP2C is also available with built-in transformer.

GNS gooseneck set are available in 7, 14, and 20-inch lengths. Set includes a female XLR module with locking ring, a 5/8-inch thread adapter, gooseneck, mounting hardware. Stainless steel with black matte finish.

Speakon loudspeaker/amplifier connectors come in a cable and a chassis version. All contacts are touch proof. Terminations are solderless. Chassis connectors are airtight. Available in right angle versions.

X series XLR audio connectors have only 4 parts and no screws or crimping. Shell of die-cast zinc. Satin nickel or black chrome finish. Rear-mountable glass fiber insert. Flex relief.

NC3FX-S female cable connector with built-in on/off switch attaches to the cable. Rotary switch action. Wires up like a normal XLR connector.

Heavy duty XLR cable connectors are available in matte nickel finish with gold contacts.

NP2RC right-angle phone plug is die cast zinc shell in satin nickel or black chrome finish.

NP3TT professional 3-pole miniature 4.4mm phone plugs are used mainly in high-density jack fields. Features include hard optalloy plating, insertion and retention forces are directed to metal surfaces only, contact elements can be soldered or crimped. Handles available in black or red.

PENN FABRICATION (USA) INC.

Audio connectors, audio faders:

Cliff Electronic's range of connectors and phone jacks offer high quality metal connectors both locking and non-locking as well as low cost plastic versions.

Penn Fabrication also offers a wide range of knobs and sliders for many applications. Options include size, shape and color, as well as a choice of stem size. Aluminum machined knobs are also available.

PENNY & GILES INC.

Faders, controllers, jacks and jackfields:

PGF 1100 and PGF 1500 series is 104mm travel faders available with built-in switches to provide remote start and 'overpress' cue facilities. These are also available with connectors, mounting panels and escutcheons for mounting direct into a console's fader bay as a separate module. Faders are available with one to four channels and in balanced and unbalanced configurations.

PGF 1900 series is 128mm travel faders and have features similar to those of 1100 and 1500 series.

PGF 2200 series is 128mm travel stereo faders with individual control sliders for each channel which can be operated independently or clipped together to act as a stereo pair.

PGF 3000 series is a range of unbalanced mono and stereo faders with five alternative travel options: 45mm, 65mm, 75mm, 83mm and 104mm. These units all feature a non-contacting shield which excludes dust and spilled liquids from the interior of the fader and can be equipped with switch options to suit most applications.

PGF 4000 series is 104mm travel faders incorporating similar dust and liquid exclusion features to those of the 3000 series. This range can incorporate four elements to provide up to four matched, unbalanced channels or balanced stereo operation.

PGF 6000 series is a range of 104mm travel motorised faders available in several configurations to suit the requirements of most moving fader automation system concepts whether analog, digital or VCA based. They are compact units having very rapid response times and smooth, light operating feel when used in the manual 'updating' mode. Motorised faders are available with integral touch sensor electronics providing an output signal to the control processor when the control knob is touched by the operator.

PGF 5000 series is a range of T-Bar controllers designed for use in video switchers and special effects units.

The controllers, with single or twin potentiometer elements, are suitable for video only or audio-video applications.

Penny & Giles are the sole U.S. importers of Mosses and Mitchell Jacks and Jackfields.

Pre-wired and unwired jackfields are available with either standard 1/4-inch or miniaturized telephone jacks. Pre-wired units can be configured to individual requirements on request.

A comprehensive range of jack sockets, panels and accessories is also available for users who prefer to assemble their own jackfields.

WIREWORKS CORPORATION *See our ad on page 8*

Mic cables, audio/video cables, audio/video connectors:

Mic cables come in five different types and are ideal for in-studio and outdoor broadcast use; on-stage and on-camera use where aesthetics are important. Jackets are ethylene propylene diene monomer, hypalon, neoprene, or PVC.

Mic multi-cables are multi-pin disconnectable and hard-wired multi-pair cable assemblies which enable you to configure a system exactly suited to your needs. Jackets are PVC.

Audio/video multi-cables incorporate audio and video cables in one outer jacket. For broadcast, ENG, EFP, and CCTV applications transmitting sound and picture for monochrome and color TV cameras as well as for remote control and monitor/cue line systems.

Coaxial cables feature completely assembled video cables available in male/female connector configurations for video use and RF applications. Crimped assemblies assure accurate, reliable connections.

Multiboxes and multiracks in 3 to 50 channels.

Transformer isolated mic splitters, box or rack, 3 to 50 channels, XLR or multipin input, up to 3 isolated and 1 direct multipin output.

Multitails in 3 to 50 channels, 4 or 8 foot tails.

Mic cable tester-$99.00.

Multipin chassis mount or line type connectors, 3 to 50 channels, assembled or kits-$42.00 to $482.00.

YORKVILLE SOUND INC.

Cable; MIDI, instrument, speaker, mic, patch

Transformer cable LHNT-1, LHNT-25 provide impedance matching between low Z microphones and high Z inputs. Made from ultra-flexible cable in 1 and 25-foot lengths, these cables provide matching of sources ranging from 50 to 500 ohms and inputs of 2k ohms or more. Neutrik transformers are housed in compact casing.

Microphone cables feature Neutrik brand connectors, Canare and Belden cable. Other features include rugged flexible outer sheathing, spiral shielding, and 4-conductor types that offer 40-strand conductors and provide over 90 percent more isolation from electro-magnetically induced noise than 2-conductor cables.

Patch cables include models that offer Switchcraft 1/4-inch connectors ranging from 1 to 30-feet. The Neutrik series provides a more flexible cable in conjunction with Neutrik 1/4-inch connectors.

Speaker cable comes in a number of types and configurations such as 1/4-inch to 1/4-inch 18 AGW, 1/4-inch to 1/4-inch 16 AGW, XLR male to bare tinned ends 16 AGW, XLR male to 1/4-inch 16 AGW. Other features include Switchcraft plugs or Neutrik XLR connectors.

ADDRESSES

Studio Accessories

Canare Cable, Inc.
832 N. Victory Blvd.
Burbank, CA 91502

Connectronics Corporation
652 Glenbrook Rd.
Stamford, CT 06906

Four Designs Co.
6531 Gross Ave.
Canoga Park, CA 91307

Furman Sound Inc.
30 Rich St.
Greenbrae, CA 94904

Gotham Audio Corporation
1790 Broadway
New York, NY 10019-1412

Marshall Electronics Inc.
PO Box 2027
Culver City, CA 90230

Monster Cable
101 Townsend St.
San Francisco, CA 94107

Neutrik (Dialight)
1913 Atlantic Ave.
Manasquan, NJ 08753

Penn Fabrication (USA) Inc.
PO Box 356
29 B Ethel St.
Hawthorne, NJ 07506

Penny & Giles Inc.
2716 Ocean Park Blvd., Suite 1005
Santa Monica, CA 90405

Wireworks Corporation
380 Hillside Ave.
Hillside, NJ 07205

Yorkville Sound Inc.
56 Havester Ave.
Batavia, NY 14020

WIRELESS MICROPHONES

BEYER DYNAMIC INC.

TS-185 wireless instrument mic'ing system is a body pack transmitting system with MCE-6 omni or MCE-10 hypercardioid miniature microphones for interface with brass, stringed, wind or percussion instruments. Available in full diversity or non-diversity modes. System incorporates the following instrument mounts: MTH-5 brass/woodwind, MAG-5 acoustic guitar, MFH-5 flute mount, MGH-5 violin/cello/acoustic bass.

SCV-185 wireless shotgun microphone system has interchangeable polar capsule capability, available in full diversity or non-diversity modes, hand held or body pack transmitter. Receiver is AC or DC powerable. Transmitters interface with the following interchangeable polar capsules: CK 706 short shotgun, CK 707 long shotgun, CK 708 bidirectional, CK 703 cardioid, and the CK 701 omni-capsule.

SBM-185 wireless vocal system has interchangeable polar/response heads, available in full diversity or non-diversity modes. Transmitter with on-board limiter can be supplied with the following vocal responses: BM 85 cardioid ribbon head, EM 85 cardioid condenser head, and the EM 81 supercardioid condenser head.

TS-185 wireless lavalier system is a body pack transmitting system with MCE 5 or MCE 10 lavalier mics, available in diversity or non-diversity modes. Optional AH-85 adapter permits interface with other popular brands of lavalier mics.

CETEC VEGA

Pro Plus 87/42 diversity wireless microphone system consists of the model T-87 handheld transmitter and the model R-42 diversity receiver. The T-87 is housed in an contoured black case with internal dipole antenna. It uses a Shure SM87 condenser element and dark gray windscreen. The R-42 true dual-diversity receiver features an FET front end for ultra low noise and widest RF dynamic range (typically 108dB, A-weighted).

Pro Plus 77/87 portable diversity wireless microphone system consists of the model 77/DII body pack transmitter and the model 67B portable diversity receiver. The 77/DII accepts most electret mics equipped with a lemo connector. It operates from a 12 V camera pack (or other external source of +10.5 to +18 Vdc) or from its internal battery pack (four 9 V alkaline cells).

Pro Plus 77/33 portable wireless microphone system consists of the model 77/DII bodypack transmitter and the model R-33 miniature portable receiver. The R-33 is small enough to mount on a camera, or on a tiny corner of a sound cart, or in a pocket, or on a belt. It measures only 3.3 inches wide by 5.5 inches deep by 0.8 inches high. System range is up to 1000 feet, due to the improved sensitivity and intermodulation performance. A special miniaturized true helical resonator filter, along with 10 poles of IF filtering, provides sharp selectivity.

Traveler 1-B portable wireless microphone system consists of the model T-37 bodypack transmitter and the model 66-B portable receiver. The T-37 pocket transmitter accepts most electret mics equipped with a miniature XLR connector. A mic on/off toggle switch and a recessed battery-power on/off switch are mounted on the control panel. It runs on a standard 9V alkaline battery. The 66-B features a GaAsFET preamplifier transistor for improved RF sensitivity and range. It uses 13 poles of IF filtering for superb IF selectivity and adjacent-channel rejection.

HM ELECTRONICS, INC.

System 50-Highband VHF, wireless body pack

The RX520, switching diversity receiver and TX550 body pack transmitter have dual frequency capability, permitting interference free operation. Balanced mic or line output, NRX-II noise reduction and an LED bargraph indicates audio level display. The System 50 also includes whip antenna and an AC adapter.

Price: $1095.00.

System 55-Highband VHF, wireless handheld mic system

TX555 transmitters combine with the RX520 switching diversity receiver. Available with SM85, SM87, SM58, SM57 and HM58 mic capsules (other capsules available by special order). Along with NRX-II noise reduction, the antenna radiation characteristics increase the range (up to 1/4 mile). Locking mic-mute and RF switches provide optimum control. Multiple handhelds can be provided to the advantage of the dual frequency nature of the RX520. The System 55 comes with whip antennas, AC adapter, pouch and mic clamp.

Price: $1110.00.

PEAVEY ELECTRONICS

Wireless Performer is a dynamic cardioid high-band system which can be expanded to diversity by using a second receiver. The receivers are standard rack mount. Dimensions of the hand-held mic are 9.75 inches by 1.75 inches.

Price: $649.50.

SAMSON TECHNOLOGIES CORPORATION

CTD-757 Concert Series has E-V N/Dym 757 mic element transmitter, 10 available VHF frequency channels, operating range of 300 feet, frequency response of 30Hz to 18kHz ±3 dB, S/N ratio greater than 100dB, THD less than 0.5 percent at 1kHz, transmitter has power and mute switches and audio trimpot.

Price: $1075.00.

CTD-58 Concert Series has Shure SM-58 mic element transmitter, 10 available VHF frequency channels, operating range of 300 feet, frequency response of 30Hz to 18kHz ±3dB, S/N ratio greater than 100 dB, THD less than 0.5 percent at 1 kHz, transmitter has power and mute switches and audio trimpot.

Price: $1075.00.

CTD-BK1 Concert Series has E-V BK-1 mic element transmitter, 10 available VHF frequency channels, operating range of 300 feet, frequency response of 30 Hz to 18 kHz ±3 dB, S/N ratio greater than 100 dB, THD less than 0.5 percent at 1kHz, transmitter also has power and mute switches and audio trimpot.

Price: $1100.00.

CTD-85 Concert Series has Shure SM-85 mic element transmitter, with power and mute switches, and audio trimpot, 10 available VHF frequency channels, operating range of 300 feet, frequency response of 30 Hz to 18 kHz ±3dB, S/N ratio greater than 100 dB, THD less than 0.5 percent at 1 kHz.

Price: $1300.00.

CTD-87 Concert Series has Shure SM-87 mic element transmitter, with power and mute switches, and audio trimpot, 10 available VHF frequency channels, operating range of 300 feet, frequency response of 30 Hz to 18 kHz, THD less than 0.5 percent at 1 kHz.

Price: $1300.00.

CTD-55 Concert Series has 10 available VHF frequency channels, operating range of 300 feet, frequency response of 30Hz to 18kHz ±3 dB, S/N ratio greater than 100 dB, THD less than 0.5 percent at 1 kHz, with Sony ECM-55 mic element lavalier clip transmitter.

Price: $1100.00.

The above models have true diversity circuitry that monitors dual receivers to select the stronger signal at all times, and dbx type I and type II noise reduction integrated to improve overall audio companding.

SHURE BROTHERS INC. *See our ad on page 3*

W1020S wireless body-pack system consists of a W10BT body-pack transmitter, a W20R receiver, and WL83 lavalier microphone. System features full-range frequency response with Shure microphone quality. Receiver features p-pole linear-phase filters for high selectivity, low distortion. The W10BT transmitter accommodates virtually any input source form low-level, low impedance Shure mics to high level, high impedance electric guitars. Transmitter also features separate power and on-off switches to permit non-pop muting, and will operate up to 7 hours on a 9V alkaline battery.

Price:
W1020S system-$1200.00
W10BT transmitter-$400.00
W20R receiver-$700.00
WL83 microphone-$165.00

W1025S Diversiphase wireless body-pack system consists of our W10BT body-pack transmitter, WL83 lavalier mic, and W25DR Diversiphase receiver. The W25DR monitors signals from 2 antennas (included), locks them in-phase to prevent multi-path cancellation, and adds the signals together to provide maximum gain. Typical A-weighted dynamic range is 98 dB. W10BT transmitter accepts either mic or line-level output. A wide range of options is available. Computer selected frequencies allow 12 or more W25DR/W10BT systems to operate at one location without interference.

Price:
W1025S system-$1700.00
W10BT transmitter-$400.00
W25DR receiver-$1200.00
WL83 microphone-$165.00

TELEX COMMUNICATIONS, INC.

FMR-2 wireless microphone system with dual antenna diversity reception includes the FMR-2 receiver and WHM-500 hand held condenser microphone/transmitter. Typical specifications are a 50-15,000 Hz frequency response and a system dynamic range of 102 dB A-weighted.

ADDRESSES

Microphones

Altec Lansing Corporation
10500 W. Reno
Oklahoma City, OK 73128

AMR
PO Box 1230
Meridian, MS 39301

Audio-Technica
1221 Commerce Dr.
Stow, OH 44224

Beyer Dynamic Inc.
5-05 Burns Ave.
Hicksville, NY 11801

Bruel & Kjaer Instruments Inc.
185 Forest St.
Marlboro, MA 01752

C-Tape Developments Ltd.
3050 SW 14th Place, Suite 3
Boynton Beach, FL 33434

Calrec Audio Limited
PO Box 31864
Seattle, WA 98103

Carvin Corporation
1155 Industrial Ave.
Escondido, CA 92025

Cetec Vega
9900 Baldwin Place
El Monte, CA 91731-2204

Countryman Associates Inc.
417 Stamford
Redwood City, CA 94063

Crown International
1718 W. Mishawaka Rd.
Elkhart, IN 46517

Electro-Voice, Inc.
600 Cecil St.
Buchanan, MI 49107

Gotham Audio Corporation
1790 Broadway
New York, NY 10019-1412

HM Electronics, Inc.
6675 Mesa Ridge Rd.
San Diego, CA 92121

Milab International, AB
11288 Ventura Blvd. Suite 304
Studio City, CA 91604

Paso Sound Products
14 First St.
Pelham, NY 10803

Peavey Electronics
711 A St.
Meridian, MS 39301

Ramsa/Panasonic
6550 Katella Ave.
Cypress, CA 90630

Samson Technologies Corporation
485-19 S. Broadway
Hicksville, NY 11801

Sennheiser Electronic Corporation
6 Vista Dr.
Old Lyme, CT 06371

Shure Brothers Inc.
222 Hartrey
Evanston, IL 60204

Tascam/Teac Pro Division
7733 Telegraph Rd.
Montebello, CA 90640

Telex Communications Inc.
9600 Aldrich Ave. S.
Minneapolis, MN 55420

Yamaha Music Corporation
6600 Orangethorpe Ave.
Buena Park, CA 90620

Yorkville Sound Inc.
56 Havester Ave.
Batavia, NY 14020

	Model	Type	Pattern(s)	Frequency Response, dB, +, −dB	Impedance, ohms	Sensitivity, 1 kHz, −dBm	Sound Pressure Level at % distortion	Dimensions,L,D,W	Weight, oz.	Finish	Connector	Price,$	Features
AKG ACOUSTICS	C-3-	cond stereo	multi	20-20k	200	-62 0.5	132 1.29	7.92 9.88		nik	12 pin din	$1995.00	small diaphragm stereo with FET pre-amp, nine polar patterns selected by remote control.
	C-422	cond stereo	multi	20-20k	200	-65 0.5	133 1.29	9.25 15.88		satin blk/chr	12pin din	$2895.00	large diaphragm stereo with same features as the C-34.
	AKG Tube	cond	multi	30-20k	200	-60	128	8.9 1.65	24	text Tuchel suede		$1995.00	large diaphragm using low noise 6072 vacuum tube, nine polar patterns via remote.
	C-414B/ ULS	cond	multi	20-20k	180	-38	134 0.5	5.6 1.8	11	satin blk/chr		$995.00	FET with large diaphragm capsule, offers four patterns with 3 bass roll-offs, phantom power.
	C-414B/TL	cond	multi Xformer	20-20k	180	-38	140 0.5	5.6 1.8	11	satin blk/chr		$1195.00	same as C-414B/ULS, but has transformerless output for improved low end performance.
	C-460B/ CK61ULS	cond combo	card	20-20k	120	-43	134 0.5	6.8 0.8	4.9	satin black	XLR	$530.00	switchable bass roll-off/attenuator, 4 positions.
	C-522	elect ster	X-Y stereo	20-20k	300	-40	128 0.1	8.5 2.05	10	satin black	5 pin XLR	$950.00	hand held or boom X-Y stereo mic. Internal rech. battery or 9-52V phantom powerable.
	C-562	cond	layer boundary	20-20k	600	-33	130 1.0	0.375 6.3	2.1	satin black	XLR	$395.00	mic is mounted on flat round plate, screw holes for mounting, phantom powering.
ALTEC LANSING CORPORATION	C641L	cond	omni	80-15k	150	-45	141	0.85 0.42	0.7	matte black	A3F	$284.00	clip-type lavalier with cord and battery pack with belt clip.
	C643	cond	card	50-18k	150	-49	152	7.95 1.90	10.2	matte gray	A3M	$236.00	
	C644	cond	card	50-18k	150	-50	130	7.5	12	matte black	A3M	$228.00	
	D645	dyn	super card	80-15k	150	-56	130	6.56 1.38	6	matte beige	A3M	$368.00	
	D646	dyn	super card	80-15k	150	-57		7 1.78	8	silver beige	A3M	$400.00	
	D647	dyn	omni	80-15k	150	-55		7 1.41	6	matte beige	A3M	$152.00	
	C649	cond	card	40-18k	150	-45	141	7 1.06	8	matte beige	A3M	$380.00	
	654A	dyn	card	50-15k	200	-56		7.25 2	8	matte silver	A3M	$200.00	

	Model	Type	Pattern(s)	Frequency Response, dB, +, −dB	Impedance, ohms	Sensitivity, 1 kHz, −dBm	Sound Pressure Level at % distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features
AMR (Audio Media Research)	ERO-10	cond	card	40-20k	250	-52	135	5.5 0.865	5	texture gray	XLR	$199.50	operates with 9 to 52V phantom power, plush lined case.
	ERC-12	cond	card	60-20k	250	-57	140	5.5 0.865	5	texture gray	XLR	$199.50	operates with 9 to 52V phantom power, plush lined case.
	DM-12	dyn	card	80-19k	250	-77	140	6.5 2	8	champagne	XLR	$129.50	integral hum compensation coil, plush lined case.
	DM-14	dyn	hyper card	80-18k	250	-77	140	6.5 2	8	champagne	XLR	$139.50	integral hum compensation coil, plush lined case.
AUDIO-TECHNICA *See our ad on page 9*	ATM-41a	dyn	card	50-16k	250	-56		7.2 2.1	10	matte black	XLR	$180.00	
	ATM-63	dyn	card	50-17k	250	-56		6.25 1.3	9.5	matte black	XLR	$170.00	
	ATM-73	dyn	card	60-15k	400	-60	140		1.1	matte black	XLR	$225.00	headworn microphone.
	ATM-10	elect cond	omni	40-18k	600	-48	125	7.5 1.3	5.6	matte black	XLR	$140.00	
	ATM-11	elect cond	card	50-20k	600	-56	130	8.2 1.4	6	matte black	XLR	$155.00	
	ATM-11R	elect cond	card	30-20k	200	-49	141	8.2 1.4	6	matte black	XLR	$205.00	
	ATM-15	elect cond	card	50-20k	400	-50	130	0.9 0.39	0.1	matte black	XLR	$185.00	
	ATM-21	dyn	card	50-15k	600	-60		7.75 7.7	7.6	matte black	XLR	$175.00	
	ATM-31	elect cond	card	60-20k	600	-55	125	8.0 2.8	6.5	matte black	XLR	$160.00	
	ATM-31R	elect cond	card	30-20k	200	-49	141	8.0 2.0	6.5	matte black	XLR	$210.00	
	ATM-33R	elect cond	card	30-20k	150	-45	141	7.0 1.2	4.7	matte black	XLR	$225.00	
	ATM-5R	elect	card	40-20k	200	-53	140	5.06 1.5	4	matte black	TB3M	$265.00	

	Model	Type	Pattern(s)	Frequency Response, dB, +, −dB	Impedance, ohms	Sensitivity, 1 kHz, −dBm	Sound Pressure Level at % distortion	Dimensions,L,D,W	Weight, oz.	Finish	Connector	Price,$	Features
BEYER DYNAMIC INC.	MC740	cond	five	40-20k	150	-40	144	8.2	10.2	anod	XLR	$1150.00	multi pattern condenser, 48V phantom power.
	MCE10	cond	hyper card		200	-42	144	0.75 0.6	1.5	anod black	XLR	$395.00	super-small condenser, battery or phantom powered.
	MCE81	cond	super card	50-18k	200		138	8.0 2.2	6.2	anod black	XLR	$325.00	weight balanced, 12-48V phantom powered.
	M700	dyn	super card	40-16k	200		140		6.3	anod black	XLR	$260.00	high gain before feedback.
	M380	dyn	bi-dir card	15-20k	200	-46	140	7.0 5.0	8.5	anod black	XLR	$280.00	internal shockmount.
	M420	dyn	hyper card	100-12k 200	-57	150	4.5	6 1.0		anod black	XLR	$195.00	for close mic'ing of percussion instruments.
	M422	dyn	super card	100-12k 200	-59	150	4.0	4.8 1.0		anod black	XLR	$125.00	small, excellent for close mic'ing of percussion instruments.
	MCE81	cond	super card	50-18k	200	-50	138	8.0 2.0	6.0	anod black	XLR		12-48V phantom powered, for vocals.
BRUEL & KJAER INSTRUMENTS, INC.	4003	cond	omni	10-20k 2	30	-26	135	6.5 0.63	5.3	brass/ black	XLR	$1135.00	for low level applications, externally powered from the B&K supply type 2812.
	4004	cond	omni	10-40k 2	30	-40	168	6.5 0.47	5.3	brass/ black	XLR	$1135.00	for high peak levels, high frequencies, powered from B&K power supply.
	4006	cond	omni	20-20k 2	30	-38	135	6.5 0.63	5.3	brass/ black	XLR	$1135.00	for low noise applictions, P48 phantom powered, with lower sensitivity.
	4007	cond	omni	20-40k 2	30	-52	155	6.5 0.47	5.3	brass/ black	XLR	$1135.00	for high level, high frequency applications, lower sensivity than 4004, P48 phantom powered.
	4011	cond	card	40-20k 2	180	-40	158	6.89 0.75	6.6	brass/ black	XLR	$NA	
C-TAPE DEVELOPMENTS LTD.	CXS/8	contact cond	42-22k	600 3		150	8.0 0.1	1.9 0.75		brown plast	XLR/ phone	$399.99	stereo system, 24-48V phantom power, designed for piano, mic to line attenuator.
	CX2/8	contact cond	42-22k	600 3		150	8.0 0.1	3.1 0.75		brown plast	XLR/ phone	$299.99	same as CXS/8 but mono, for larger instruments, designed for live applications.

Model	Type	Pattern(s)	Frequency Response, dB, +, −dB	Impedance, ohms	Sensitivity, 1 kHz, −dBm	Sound Pressure Level at % distortion	Dimensions,L,D,W	Weight, oz.	Finish	Connector	Price,$	Features
B1/8N	contact cond	42-22k	10k 3		150	8.0 0.1	1.9 0.75		brown plast	0.25in phone	$118.00	uses single 8-inch tape, 9V battery powered, includes belt clip and battery.
B2/8	contact cond	42-22k	10k 3		150	8.0 0.1	3.1 0.75		brown plast	0.25in phone	$185.00	same as B1/8N but uses two tapes, covers full range of larger instruments.
B2/8	contact cond	42-22k	10k 3		150	8.0 0.1	3.7 0.75		brown plast	0.25in phone	$199.00	same as B2/8 but with pan pot, used to balance high and low registers of instrument.
SX1	contact cond	42-22k	10k 3		150	8.0 0.1	0.7 0.75		nickel plated	0.25in phone	$179.00	
APT 5	contact cond	42-22k	600 3		150	8.0 0.1	1.9 0.75		brown plast	XLR/ phone	$699.00	for use on drums, provides high quality signal and trigger outputs for synths.
APT 2	contact cond	42-22k 600	3	150	8.0	1.9 0.1	0.75		brown plast	XLR/ phone	$499.00	same as APT 5 but with two channels, used to extend APT 5, for toms and percussion.

CALREC AUDIO LIMITED
(Calrec by AMS)

Model	Type	Pattern(s)	Frequency Response, dB, +, −dB	Impedance, ohms	Sensitivity, 1 kHz, −dBm	Sound Pressure Level at % distortion	Dimensions,L,D,W	Weight, oz.	Finish	Connector	Price,$	Features
CM1050C	cond	card	30-20k 3	200	0.8mV/ uBar	130 0.5	5.5 0.875	4.0	black st.steel	XLR	$215.00	fixed capsule.
CM1051C	cond	card	40-20k 3	200	0.8mV/ uBar	130 0.5	5.5 0.875	4.0	black st.steel	XLR	$215.00	fixed capsule.
CM2001C	cond	card	20-20k 3	200	0.8mV/ uBar	130 0.5	6.625 0.875	4.2	black st.steel	XLR	$295.00	detachable capsule, uses 48V phantom power.
CM2003	cond	omni	20-20k 3	200	0.8mV/ uBar	130 0.5	6.25 0.875	4.2	black st.steel	XLR	$280.00	detachable capsule, uses 48V phantom power.
CM2150C	cond	card	30-20k 3	200	0.6mV/ uBar	130 0.5	6.25 0.875	4.2	black st.steel	XLR	$325.00	detachable capsule, uses 7.5-50V power.
CM2151C	cond	card	40-20k 3	200	0.6mV/ uBar	130 0.5	6.25 0.875	4.2	black st.steel	XLR	$325.00	detachable capsule, uses 7.5-50V power.
CM2056C	cond	card	40-20k 3	200	0.8mV/ uBar	130	7.25 1.875	4.5	black st.steel	XLR	$330.00	detachable capsule, uses 48V phantom power.
CM4050	cond	omni	20-20k 1	100	various	140	9.5 1.5	18	black st.steel	XLR	$3950.00	Soundfield microphone, polar patterns may be varied post session if 4 directional information channels are stored.

Manufacturer	Model	Type	Pattern(s)	Frequency Response, dB, +, −dB	Impedance, ohms	Sensitivity, 1 kHz, −dBm	Sound Pressure Level at % distortion	Dimensions,L,D,W	Weight, oz.	Finish	Connector	Price,$	Features
CARVIN CORPORATION *See our ad on Cover 3*	CM67	dyn	card	40-15k	250	-77	128	6.313 1.531	10	alum/ cast	XLR	$99.00	for vocal or instrument applications, takes higher levels.
	CM68	dyn	card	45-15k	150	-74	128	6.563 2.031	10.25	gray enamel	XLR	$99.00	for vocals, unidirectional pattern minimizes backround noise,"pop" filter.
	CM90E	cond	card	30-20k	150	-75	128	7.625 0.75	9.0	cast enamel	XLR	$139.00	high sensivity, for church choirs and drums.
COUNTRYMAN ASSOCIATES, INC.	Isomax 2	elect	all	30-20k 3	600	-58 -38	150 1	0.625 0.312	0.1	black colors	XLR	$203.25	extremely small. Very flat response and a 150dB overload level.
	Isomax 3	elect	all	30-20k 3	600	-58 -38	150 1	goose neck		black colors	XLR	$224.70	same element as the Isomax 2 but mounted on a flexible gooseneck boom.
	Isomax 4	elect	hyper card	65-15k 3	600	-48	125 3	goose neck		black colors	XLR	$331.70	designed especially for podiums,featuring electronic vibration isolation.
	Isomax TVH	elect hyper	65-15k card	600 3	-48	125	0.875 3	0.3 0.468		black colors	XLR	$310.25	picks up much less room sound and has electronic vibration isolation for low noise.
CROWN INTERNATIONAL	PZM-30FS	elect	hemi	20-15k 3	240	-46	150 3	6.0 0.750	6.5	silver alum	XLR	$349.00	flat response for studio. High rising response for bright sound, integral electronics
	PZM-6FS	elect	hemi	20-15k 3	240	-46	150 3	3.0 0.375	5.0	silver alum	XLR	$349.00	inconspicuous, flat response, for conference or suspended over ensemble, permanent cable.
	PCC-160	elect	half card	50-18k 3	150	-31	120 3	6.7 0.840	11.5	black steel	XLR	$275.00	surface-mounted supercardioid for stage floor, lectern, news desk. Bass-tilt switch.
	GLM-100	elect	omni	20-20k 3	240	-50	150 3	0.755 0.310	2.8	black PVC	XLR	$199.00	miniature, clips to instruments or clothes, studio quality, many mounting accessories.
	GLM-200	elect	hyper card	60-20k +4,-6	100	-44	150 3	0.755 0.310	3.7	black PVC	XLR	$229.00	miniature, clips to instruments or clothes, studio quality, many mounting accessories.
	CM-100	elect	omni	20-20k +5,-4	240	-52	150 3	7.311 1.700	7.8	black alum	XLR	$189.00	handheld stage vocal mic., no proximity-effect, pop filter, low noise.
	CM-200	elect card	80-15k	200 +3,-6	-52	142	7.530 3	7.0 1.800		black alum	XLR	$259.00	handheld vocal/instrument mic., pop filter, has premium wood handle for add. $50.00
	CM-300	elect	card	60-18k 6	150	-55	148 3	7.335 2.040	7.1	black alum	XLR	$309.00	handheld differoid stage vocal mic. Pop-filter, high gain, excellent isolation.

Manufacturer	Model	Type	Pattern(s)	Frequency Response, dB, +, −dB	Impedance, ohms	Sensitivity, 1 kHz, −dBm	Sound Pressure Level at % distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features
ELECTRO-VOICE INC. *See our ad on page 7*	N/D 757	dyn	super card	25-22k	150	-50		7.12 2.05	7.7	matte black	XLR	$247.50	is a lead vocal mic with switch-selectable rolloff.
	N/D 457	dyn	hyper card	25-21k	150	-50		7.12 2.05	7.05	matte black	XLR	$185.00	is a high output vocal mic specially tuned for superior gain-before-feedback.
	N/D 357	dyn	super card	25-20k	150	-53		7.12 2.05	7.05	matte black	XLR	$150.00	is a vocal mic with a midrange presence peak and extended high-end response.
	N/D 257	dyn	card	35-19k	150	-53		7.12 2.05	7.05	matte black	XLR	$108.00	is a vocal mic that permits greater freedom of movement due to type of cardioid pattern.
	N/D 408	dyn	super card	30-22k	150	-50		4.55 2.85	6.7	matte black	XLR	$190.00	has large-diaphragm design for extended high frequency response.
	N/D 308	dyn	card	40-20k	150	-53		4.55 2.85	6.7	matte black	XLR	$155.00	same features as the N/D 408.
GOTHAM AUDIO CORPORATION (NEUMANN) *See our ad on page 24*	SM-69fet	cond	2X various	40-16k	200	19mV/ Pa	123 0.5	10.2 1.9	20	nickel dk.matte	spcl.	$3650.00	is a concert hall standard, m-s/x-y stereo microphone.
	KM-84	cond	card	40-20k	200	10mV/ Pa	120	4.3 0.83	3.0	nickel matte blk	XLR	$449.00	is the affordable Neumann, uniform off-axis response ± 135 degrees), same quality as the SM69.
	KMS-84	cond	card	40-18k	150	5mV/ Pa	138	7.0 1.6	9.0	nickel matte blk	XLR	$1000.00	is Neumann's only"live" performance mic.
	KMR-81	cond	lobe	40-18k	150	18mV/	128	8.9	5.0	nickel	XLR	$885.00	is a shor t"shotgun" with low off axis.
	KMR-82	cond	lobe X1	40-20k	150	21mV/ Pa	128 0.5	15.5 0.82	8.75	nickel matte blk	XLR	$985.00	is a"shotgun" type with low off axis coloration.
	RSM-190	cond	hyper card	40-18k	50	23mV/ Pa	134 0.5	8.4 1.2	10.5	dark matte	XLR	$1995.00	is a transformerless, m-s/x-y stereo short "shotgun" with active matrix.
	U-89	cond	omni card	40-18k	150	8mV/ Pa	122 0.5	7.3 1.8	14	nickel dk matte	XLR	$1475.00	is for on-air broadcasting, narration, voice over and film scoring.
	U-87A	cond	omni card	40-16k	200	28mV/ Pa	117 0.5	7.87 2.20	17.7	nickel dk matte	XLR	$1600.00	is an improved studio standard mic, 10dB greater output than previous U87.
HM ELECTRONICS, INC.	RM77	elect	card	150-15k low	-72		7.5	2.0		silver metalic	XLR	$144.00	has reverb (analog bucket brigade) built right into the mic.

Manufacturer	Model	Type	Pattern(s)	Frequency Response, dB, +, −dB	Impedance, ohms	Sensitivity, 1 kHz, −dBm	Sound Pressure Level at % distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features
MILAB INTERNATIONAL, AB	HM58	dyn	card	80-14k	low	-75		6.6 2.0		charc. gray	XLR	$164.00	designed for professional applications, mic-muting.
	EM43-4	elect	omni	20-20k	2.2k	-63		0.79 0.29		black	Tini QG	$70.00	is designed for"wireless" (includes by-pass capacitor), mic clip, wind screen, case.
	VIP-50	cond	multi	40-20k 2.5	180	14mV/ Pa	123 0.5	6.69 14.1 1.85		matte black	XLR	$1295.00	
	LC-28	cond	card	40-20k 2.5	170	6mV/ Pa	130 0.5	7.28 11.2 1.06		matte black	XLR	$495.00	
	DC-96B	cond	card	40-20k 2.5	170	6mV/ Pa	128 0.5	5.7 7.05 1.06		matte black	XLR	$695.00	
	VM-41	cond	card	40-20k 2.5	170	10mV/ Pa	128 0.5	5.27 0.74		matte black	XLR	$445.00	
PASO SOUND PRODUCTS, INC.	M501	dyn	card	50-15k 3	250			6.25 1.75	15	charc gray	att. cable	$100.00	All models incorporate electroplated scratch and stain resistant finish on the housing of the microphone. A doubly redundant shock mount system effectively isolates the element from handling vibrations thus reducing overall noise. The mics come complete with carrying case, stand holder and cable.
	M601	dyn	card	50-15k 3	250			6.5 1.75	9.0	charc gray	XLR	$125.00	
	M701	dyn	card	40-16k 3	250			6.25 2.0	11	charc gray	XLR	$156.00	
	M800	dyn	card	40-18k 3	250			6.25 2.0	11	charc gray	XLR	$201.00	
PEAVEY ELECTRONICS	PVM-38	dyn	card	50-16k	300	-56		5.75 1.875	7.0	slate gray	XLR	$199.50	features a dual-element pop filter, hum compensation coil, case, cable, windscreen.
	PVM-45	dyn	hyper card	40-16k	300	-56		5.75 1.438	7.0	slate gray	XLR	$199.50	features a probe style construction and hum compensation coil, case, cable, windscreen.
	PV Mic	dyn	card	60-14k	600			6.25 2.25	8.0	slate gray	XLR	$99.50	offers zinc die-cast handle with built-in on/off switch. Low Z cable is included.
	PVM-48	elect cond	card	60-20k	150	-57		5.75 1.25	8.2	slate gray	XLR	$219.50	comes with flite-type carrying case, swivel adaptor, foam windscreen, pop filter, cable.

	Model	Type	Pattern(s)	Frequency Response, dB, +, −dB	Impedance, ohms	Sensitivity, 1 kHz, −dBm	Sound Pressure Level at % distortion	Dimensions,L,D,W	Weight, oz.	Finish	Connector	Price,$	Features
RAMSA/PANASONIC	WM-S1	elect cond	card	50-18k	600	-42	148	0.5 1.312	0.56	black	XLR	$199.00	miniature mic for use with cymbals/hihats, as well as acoustic stringed instruments.
	WM-S2	elect cond	card	120-15k 250	-56	138	0.5	0.56 1.312		black	XLR	$149.00	miniature mic for brass instruments, uses 12-48V phantom power or battery.
	WM-S5	elect cond	card	70-16k 600	-52	158	0.5	0.56 1.312		black	XLR	$269.00	miniature mic allows for mic'ing snare drums and tom toms as well as other percussion.
	WM-S10	elect cond	card	120-15k 250	-56	138	0.5	3.17 1.25		black	XLR	$199.00	miniature headset mic for vocals, harmonicas and flutes. Mic is left or right mount.
SENNHEISER ELECTRONIC CORPORATION	MD-421	dyn	card	30-17k	200				18	black	XLR	$389.00	has 5-position bass roll-off, and withstands high SPL's.
	MD-441	dyn	super card	30-20k	200				14	chrome	XLR	$515.00	available with 5-position bass roll-off and brilliance boost switch.
	MKH-40	cond	card 5	40-20k	150		142		6.5	black	XLR	$839.00	is well suited for digital sampling.
	MKE-42PU	elect	card	50-20k	1k				4.5	black	XLR	$359.00	is a pencil thin gooseneck designed for phantom powering off mixers.
	MKE-2	elect	omni	40-20k	1k					black	XLR	$239.00	available in wireless and wire formats.
	MKE-816	cond	shot gun	40-20k	10				14	satin	XLR	$1119.00	available in a variety of formats.
	MKH-20	cond	omni	20-20k 2	160	25mV/ Pa	134	1.0 6.0	3.6	matte black	XLR	$839.00	low noise, transformerless, symmetrical capsule design, 48V phantom, EQ switches.
	MKH-30	cond	figure eight	40-20k 2	150	25mV/ Pa	134	1.0 7.0	4.0	matte black	XLR	$889.00	excellent for digital M-S recording. Includes bass roll-off, pre-attenuation switch.
SHURE BROTHERS INC. *See our ad on page 3*	SM7	dyn	card	40-16k	150	-57		5.843 7.531	27	dk gry alum st	XLR	$542.00	switchable bass roll-off and presence boost and shock isolation, pop filter, case incl.
	SM15-CN	cond	card	50-15k	150	-33	141 1	0.781 0.625	2.8	black& silver	XLR	$275.00	is a high-output headworn mic, uses 5-52V phantom or standard 9V battery.
	SM81-LC	cond	card	20-20k 150	-40.5	146	8.343 1	8 0.937		chpgne steel	XLR	$367.00	has lockable 10dB attenuator, 3-pos. bass roll-off, omni directional cartridge opt.
	SM84	cond	super card	80-20k	150	-46	129 1	1.031 0.437	1.58	blk/ brass	XLR	$300.00	lavalier design, chest resonance dip filter, windscreen and mounting options included.

Manufacturer	Model	Type	Pattern(s)	Frequency Response, dB, +, −dB	Impedance, ohms	Sensitivity, 1 kHz, −dBm	Sound Pressure Level at % distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features
	SM87	cond	super card	50-18k	150	-49	142 1	7.562 1.937	6.3	gray alum	XLR	$329.00	wind/pop filter, shock mounted cartridge. Accepts 11-52V phantom power.
	SM91	cond	hemi card	20-20k	150	-45	144 0.1	0.625 3.75	9.3	matte black	XLR	$300.00	external preamp has selectable 12dB/oct. low frequency roll-off. For piano, bass drum.
	SM94-LC	cond	card	40-16k	150	-48	141	7.5	8.8	gray	XLR	$250.00	accepts almost any power source up to 52V.
	SM98	cond	card	40-20k	150	-54	153 0.1	1.25 0.468	0.4	blk/ brass	XLR	$250.00	full range response in unobtrusive package, available with many optional accessories.
TASCAM/TEAC PRO DIVISION *See our ad on Cover 2*	PE-50	elect cond	card	20-20k	200	-67	127	7.6			phone	$75.00	
	PE-80	elect cond	card	20-20k	200	-67	127	7.6			XLR	$125.00	
	PE-125	elect cond	card/ omni	20-20k	200	-67	127	7.6			XLR	$150.00	utilizes phantom powering.
	PE-150	elect cond	card	20-20k	200	-76	127	7.6			XLR	$175.00	
	PE-250	dyn	movg coil	20-20k	250	-73	150	7.0			XLR	$275.00	
	MC-701G	dyn	uni	20-20k	600	-70	127	12.0			XLR	$125.00	
TELEX COMMUNICATIONS, INC.	TE 10	cond	card	30-20k	150-200	-75	140		7.4	matte black	XLR	$166.00	mic element is suspended by flexible "fingers" that isolate from shock and vibration.
	TD 11	dyn	card	50-16k	100-250	-77			9.2	matte black	XLR	$130.00	has low distortion, tight pattern, multi-stage pop filter.
	LM-100	cond	omni	20-20k	150	-74		0.75 0.437	1.0	matte black	XLR	$225.00	is a lapel mic system that includes an LM-101 mic, 3-foot cable, and PS-10 power supp.

	Model	Type	Pattern(s)	Frequency Response, dB, +, −dB	Impedance, ohms	Sensitivity, 1 kHz, −dBm	Sound Pressure Level at % distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price,$	Features
YAMAHA MUSIC CORPORATION *See our ad on page 18*	MZ 101	dyn	uni	40-17k	250	-76		6.5 1.875	9.35	met brown	XLR	$120.00	polyester film diaphragm, for vocal use, well defined midrange and high end.
	MZ 102Be	dyn	uni	40-18k	250	-76		6.5 1.875	9.35	met brown	XLR	$170.00	beryllium diaphragm, wide range with tight response and feedback rejection.
	MZ 103Be dyn	uni	40-18k	250		-76		6.062 1.875	9.9	met gray	XLR	$210.00	beryllium diaphragm for tight response, wide range, resists off-axis sounds (fdbk prot.).
	MZ 203Be dyn	uni	40-18k	200		-76		7.25 2.0	9.7	met dk gray	XLR	$295.00	for vocal use, beryllium diaphragm, ideal for 'fast attack' instruments.
	MZ 204	dyn	uni	20-18k	250	-77		4.625 1.4	7.94	met dk gray	XLR	$275.00	for percussion, right angle connector and special mount for close placement.
	MZ 205Be	dyn	uni	40-18k	250	-77		3.5 1.33	6.53	met dk gray	XLR	$275.00	for higher pitched drums, compact right-angle connector and special mount.
	MZ 104	dyn	uni	30-17k	250	-77		7.0 1.4	9.9	met brown	XLR	$130.00	is an instrument mic with lowered sensitivity to avoid high SPL overload.
	MZ 105Be	dyn	uni	40-18k	250	-77		6.062 1.44	9.7	met gray	XLR	$180.00	beryllium diaphragm for tight response, used for close mic'ing of instruments.
YORKVILLE SOUND INC.	IM-200	cond	card	20-20k	200	-66			5.29	silver	XLR	$110.00	for sensitive vocal, instruments, recording.
	IM-400	dyn	card	50-17k	250	-77			11.28	silver	XLR	$140.00	for lead vocal, instruments and speech.
	IM-400A	dyn	card	50-18k	300	-76			9.7	black	XLR	$115.00	utilizes an AKG capsule, for lead vocals.
	IM-600	dyn	card	50-18k	250	-73			9.7	black	XLR	$65.00	for lead vocals, instruments and speech.
	IM-700	dyn	card	50-15k	500	-76			9.52	black	XLR	$50.00	for lead vocals, instruments and speech.
	IM-800	dyn	card	50-19k	600	-74			11.64	black	XLR	$95.00	for lead vocals, instruments and speech.
	IM-900	dyn	card	80-13k	600	-74			9.52	black	XLR	$50.00	for lead vocals, instruments and speech.

(Continued from page 52)

VITAL STATISTICS

SPECIFICATION	MFR'S CLAIM	db MEASURED
Pitch Control	±15%	±17%
Line Input		
Nominal, Balanced	+4 dBm	Confirmed
Nominal, Unbalanced	0.3 V	0.305 V
Max. Line Input	+24 dBm	+24 dBm
Line Output		
Nominal, Balanced	+4 dBm	Confirmed
Nominal, Unbalanced	0.3 V	0.305 V
Equalization, 15/7.5 in./sec.	NAB (3180+35μS)	Confirmed
Record Level Calibration	0VU=250 nWb/m	Confirmed
Wow and Flutter		
Weighted (15/7.5 in./sec.)	0.05/0.08%	0.024/0.037%
Unweighted (15/7.5 in./sec.)	0.10/0.12%	0.07/0.09%
Starting Time	0.5 sec. or less	Confirmed
Fast Wind Time (2500 ft.)	140 seconds	97 seconds
Signal-to-Noise Ratio (Sync/Repro)		
Unweighted (15/7.5 in./sec.)	66/67 dB	68/70 dB
Weighted (15/7.5 in./sec.)	69/70 dB	70/71 dB
Signal-to-Noise Ratio (Cue Sync)		
Unweighted (15 & 7.5 in./sec.)	58 dB	60 dB
Weighted (15 & 7.5 in./sec.)	62 dB	63 dB
THD at 0 VU (15/7.5 in./sec.)	Under 1%	0.64/0.38%
Frequency Response		
15 in./sec. (±3 dB)	30 Hz to 26 kHz	26Hz to 33 kHz
7.5 in./sec. (±3 dB)*	30 Hz to 20 kHz	22Hz to 19 kHz
Erasure	More than 70 dB	Confirmed
Crosstalk (1 kHz)	More than 77 dB	75.5 dB
Power Requirements	120V, 60 Hz, 115W	Confirmed
Dimensions (cm)	43Wx44.4Hx30.3D	Confirmed
Weight (kg)	30 (66 lbs.)	Confirmed
Price	$3,750.00	

New Products

MINISTUDIO

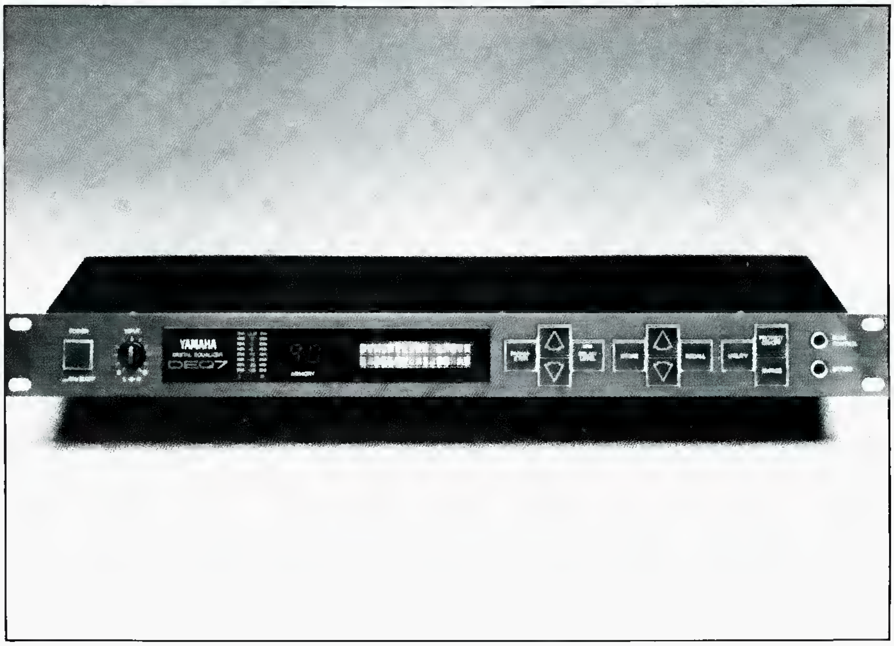
The Porta 05 is the latest addition to Tascam's line of ministudios, and utilizes Tascam's proprietary head technology. The full function 4-channel/4-track mixer-recorder offers complete professional-style channel strips with linear fader, pan and effects send level controls. The 4-in/2-out mixer configuration provides flexible signal routing, while built-in defeatable dbx noise reduction delivers noise-free recording. Separate high and low EQ allows precise sound contouring and complete tonal control. The Porta 05 is completely MIDI-compatible. A special Sync Out function dedicates Track 4 for midi clock-FSK signals and features a special band pass filter for error-free data transmission. The Porta 05 also has four tape cue level controls, a stereo bus fader and +/-15 pitch control.

Mfr.- Teac Professional Division
Price- $449.00
Circle 60 on Reader Service Card

DIGITAL EQUALIZER

The Professional Audio Division of Yamaha Music Corporation, USA has introduced the DEQ7 Digital Equalizer. It is a dual channel digital equalizer/filter system featuring 44.1kHz sampling, 16-bit conversion with 32-bit internal processing. 30 factory programs include full graphic EQ and parametric EQ configurations and shelving, notch and dynamic/sweep filters. There are 60 user-programmable RAM memory locations. Program location and bulk Dump capability is accessible via MIDI. Digital I/Os permit "converterless" operation in Yamaha format digital audio systems.

Mfr.- Yamaha Professional Audio Division
Price- $1,295.00
Circle 61 on Reader Service Card

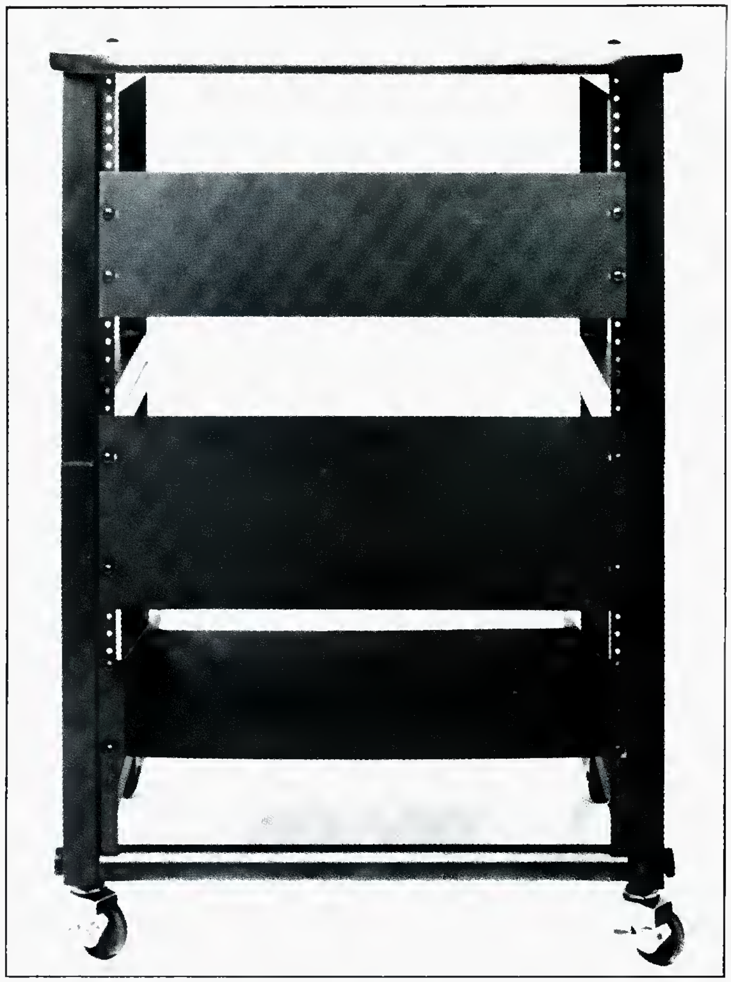


STACKABLE RACK

Solid Support Industries has introduced a new modular stackable equipment rack for the audio industry. Designed in eight rack space modules, the unit can be expanded to accommodate additional equipment as needed. All modules are constructed of lightweight 1-1/4 inch powder coated black steel tubing and are designed to fit one on top of another for continuous rack spaces, and feature recessed rack rails. The SR-8A base unit features 4 casters (2 locking) and is slot fitted to the SR-8B modular rack units. An optional 5/8-inch formica top completes the unit. The units are open at the back and sides, allowing easy access to connections, and open air cooling of equipment. An additional new option is the RS-2 two space rack shelf which allows an operator to easily access equipment that is frequently used (such as MIDI controller) or for cables, discs or other non-rack mountable accessories.

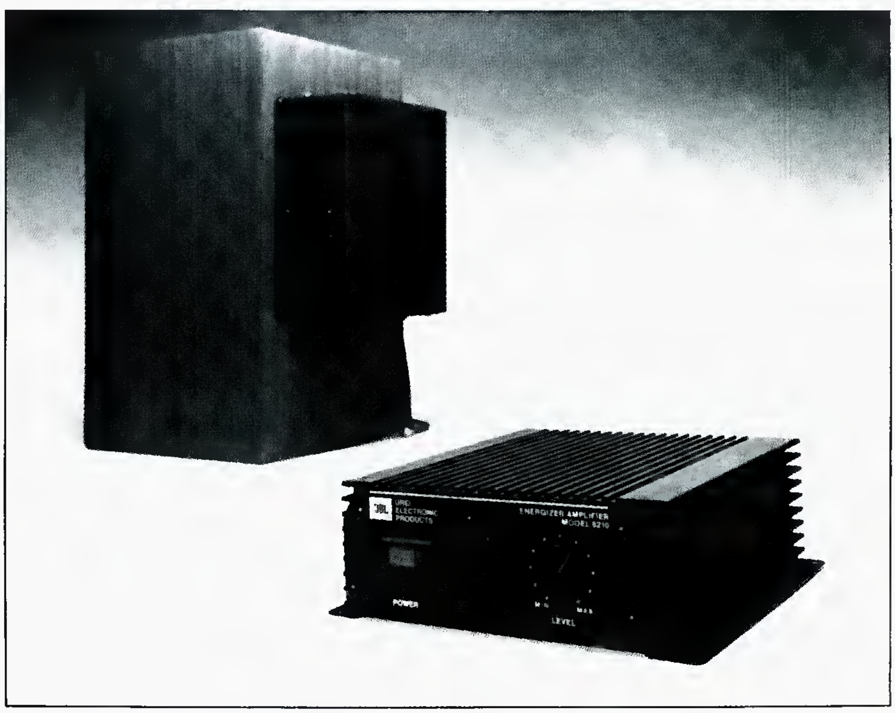
Mfr.- Solid Support Industries
Price- $120.00 (SR-8A)
$110.00 (SR-8B)
$55.00 (RS-2)
$34.95 (optional top)

Circle 62 on Reader Service Card



SELF-CONTAINED AMPLIFIERS

JBL Professional announces two new UREI Electronic Products amplifiers designed to convert the JBL Series 4400 or any other 8 ohm speaker system into a complete self-contained sound system. Both the 6210 and 6211 power loudspeakers by providing increased flexibility in mobile mixing situations, broadcast studio monitoring and other sound reinforcement applications. Fitting on the back of the speaker enclosure, the amplifiers have a symmetrical mounting pattern allowing the unit to be mounted vertically for the best cooling efficiency. Both energizer amplifiers feature 40 watts output into 8 ohms and have complementary output stages with minimum negative feedback for low TIM (transient intermodulation distortion). Both units have three pin XLR and 1/4-inch phone jack input connectors wired in parallel and active balanced inputs which will accept balanced or unbalanced line level sources. The 6211 is additionally fitted with a switch-activated preamplifier for low impedance microphone inputs and a user selectable high pass filter for reducing microphone proximity effects and wind pops. Both models include necessary template and hardware for mounting purposes. Also available is an optional 19-inch rack mounting kit, Model MR6202, allowing for the mounting of two Energizer Amplifiers side-by-side.



Mfr.- JBL Professional
Price- $297.00 (6210)
$312.00 (6211)

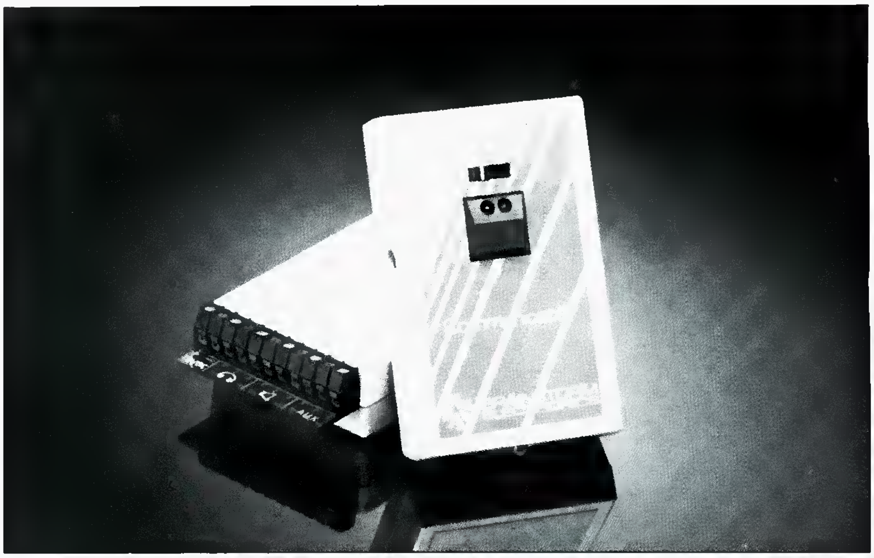
Circle 63 on Reader Service Card

REMOTE TALKBACK SYSTEM

Brainstorm Electronics, Inc. presents the Communicator, an infrared remote talkback system. It uses bidirectional PCM infrared transmission, providing total remote capability to the producer without conventional line of sight; it features 4 switchable talkback functions, and interfaces with virtually every recording console. The system includes a transmitter, a receiver, a charger for the transmitter and a power supply for the receiver. Additional transmitters are also available.

Mfr.- Brainstorm Electronics, Inc.
Price- $395.00 (complete system)
$225.00 (additional transmitter)

Circle 64 on Reader Service Card



REFERENCE CASSETTE

BASF introduces Ultra-Precise Reference Cassette for Azimuth Standard. The milled, metal-alloy reference cassette housing fills the demand of cassette duplicators and quality control departments of the major music labels for an exact measurement standard for azimuth and head alignment and for checking the quality of finished cassettes. The new reference cassette housing is manufactured in West Germany and is machined to tolerances of 5/1000th of a millimeter.

Mfr.- BASF Corporation
Price - $1200.00

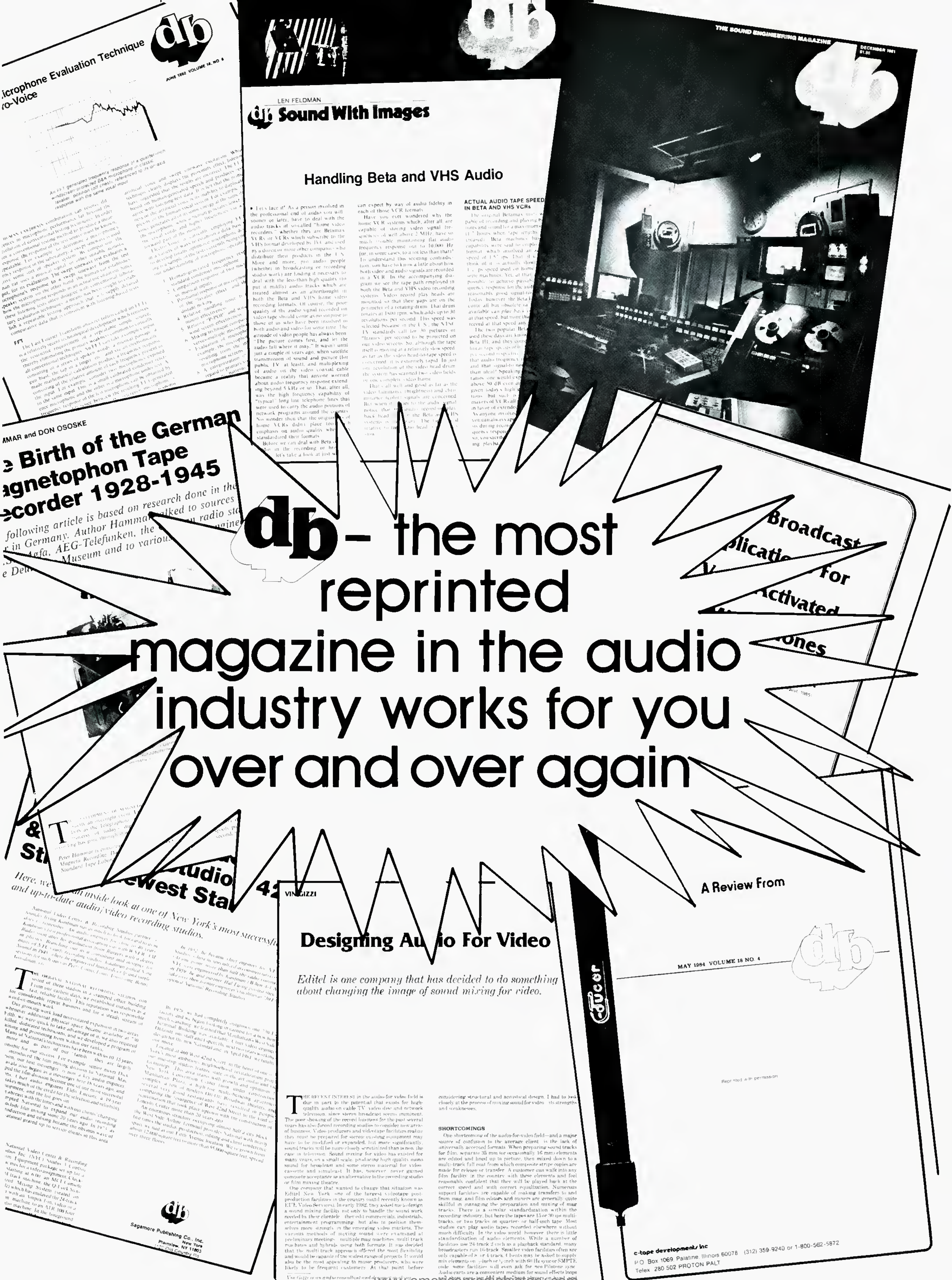
Circle 65 on Reader Service Card

MIC HOLDERS

MIC-EZE Manufacturing of Salt Lake City, Utah introduces MIC-EZE. They are clamp-on multiplex microphone holders that attach to virtually anything on or around a drumset including percussion, amps, keyboard stands as well as many horn instruments. A built-in double spring and O-ring shock mounting system prevents unwanted vibrations from reaching the microphone. The holders have 360 degrees of swivel-like rotation and are held securely in place with a sliding wingnut and bolt assembly. Made of nylon, they are extremely light and durable and come with a limited replacement warranty against breakage.

Mfr.- MIC-EZE Manufacturing
Price- $14.95

Circle 66 on Reader Service Card



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People, Places... & Happenings

• **The Joiner-Rose Group, Inc.** has been awarded design responsibility for the acoustical elements and for the sound reinforcement systems at the **Sky Dome** in Toronto, Ontario, Canada. The Sky Dome will seat up to 70,000, and will be used for live musical concerts as well as major sporting events. This facility will have a fully retractable roof, and the challenge that was given to the Dallas-based Joiner-Rose Group was to design a facility that is acoustically correct with the roof retracted or fully closed and to design a sound system that provides optimum coverage and response in both roof conditions. In addition, **Perry W. Langenstein** has been named **Principle Theatre Consultant** at The Joiner-Rose Group, Inc. Mr. Langenstein brings to the firm 25 years of technological and production experience in the area of theatre design.

• **Studiomaster**, the U.K. manufacturer of amplifiers and mixing consoles, announces the opening of corporate offices for the United States in Anaheim, California. Sales Director **Tony Allen** will oversee all aspects of the U.S. office including distribution, sales, service and promotion.

• **Pacific Cassette Laboratories**, of Torrance, California, the exclusive manufacturer and distributor of Sound of Nakamichi Reference Cassettes, has relocated its offices and studios in a newly expanded facility. The new location will enable PCL to offer custom duplication to a wider market.

• **Otari Corporation**, manufacturer of professional audio recorders, tape duplicators and tape cassette loaders, plans for a new U.S. headquarters by early 1988, located in Foster City, California. These new headquarters will double current office space; new additions will include a sound room, customer training facilities and a special test room for the laser based thermal magnetic video duplicator (TMD).

• **McMartin Industries**, a thirty-year old company, has been acquired by new management and is assuming its previous posture of a source for product and service. After a two-year absence, **Pollution Research Control Corporation** has purchased all of the assets, including the designs developed over two-hundred man years of product evolution. All assets were acquired from the secured lender of the bankrupt McMartin Industries. McMartin Industries is moving to a new facility in Council Bluffs, Iowa.

• In a move designed to accommodate their workload, **Howard Schwartz Recording, Inc.** has recently completed major renovations to its **Studio East**. The cornerstone of the project is a Solid State Logic 6000E with 24 inputs and total recall. Complementing the SSL console is a new Sony 3324 digital multi-track as well as a Sony BVU 3/4-inch video recorder and an Otari MTR 90 MK II multi-track recorder to interface with the existing 1-inch video equipment. Massive reconstruction of the studio itself was done by master studio carpenter **Tom Jahelka**. Acoustic work was done as well as cosmetic changes and some alterations to the studio's air-conditioning system in order to improve the ambient noise floor. All of the electronic design and installation was done by Schwartz's technical engineers.

• The following recent appointments have been made at dbx. **James A. Tipton** has been named **dbx** Vice President for Sales. Mr. Tipton was Northeast General Manager of Toshiba America before joining dbx. **John E. Stiernberg** has been appointed as National Sales Manager for dbx Professional Products. Mr. Stiernberg had been Pro Division National Sales Manager for Bose Corporation since 1981.

• **Ike Benoun** has announced the formation of **Audio L.A.**, a professional audio equipment supplier, specializing in systems design, installations, service and sales, with video and MIDI expertise. As a professional sales company, Audio L.A. will cater to high-end audio, video and MIDI markets, including recording studios, broadcast companies, record companies, producers, musicians, educational, government and military institutions, and a variety of corporate clients. Audio L.A. is the newly appointed sales and service organization in the Western United States for **Tascam's ATR 80 Series 24-track recorder**.

• One high point of the recent AES Convention in New York was a joint announcement by **Teldec Schallplatten GmbH** and **Georg Neumann GmbH** officially launching **DMM CD**, their revolutionary new process for manufacturing compact discs. DMM CDs are already in production and available in record stores. Both Neumann and Teldec are represented in North America by **Gotham Audio Corporation**. Sales and licensing discussions with end users are under Gotham's auspices. DMM CD embodies a new manufacturing concept which is simpler and cheaper than the laser method now in use.

• The appointment of **Michael Wuellner** as **Audio-Technica** sales manager, professional products, was announced by **Mark Taylor**, national sales manager for the division. Wuellner will work from the Audio-Technica headquarters in Stow, Ohio, coordinating the sales efforts of the company with those of its national sales representatives. Before joining Audio-Technica, Wuellner was national sales coordinator of professional products for Nakamichi.

• **Wireworks Corporation**, Hillside, New Jersey, supplier of audio and audio/video cabling assemblies to the professional audio and broadcast industries has announced its recent designation as "official cable supplier for the opening and closing ceremonies of the **Pan American Games.**" **Laurence Estrin**, engineering expert and technical director of the gala ceremonies, selected Wireworks because of his past experience with Wireworks products and personnel. Last year, while organizing the technical aspects of the July 4th Liberty Weekend festivities (see db July/August 1986). Mr. Estrin chose Wireworks to supply the audio cabling assemblies. The Tenth Pan American Games were held in Indianapolis, Indiana on August 7-23, 1987.

• **SPARS**, The Society of Professional Audio Recording Services, is now offering a comprehensive series of 90-minute cassettes and a planning notebook from the recent **Studio Business Conference** held at the **UCLA School of Business**. The 1987 conference covered a number of topics essential to financial success in the current marketplace.

Shirley Kaye has been named the new Executive Director of SPARS. Ms. Kaye has been an active member of SPARS since 1982 and has served as Regional Vice-President and Treasurer.

• Record company **Telarc International Corp.** believes it has set a record by producing the longest-playing compact disc. The CD contains 74 minutes and 23 seconds of choral music performed by the Atlanta Symphony Orchestra and Chorus, conducted by Robert Shaw. According to Robert Woods, president of the Beachwood-Ohio based classical label, the factories that make their compact discs have told Telarc that this is the absolute maximum amount of playing time possible with current technology. Telarc also announces a policy that all its new compact discs will have at least 50 minutes of music, usually more. Except for special projects, Telarc issues its recordings only on compact disc.

• U.S. operations of **Amek/TAC** announces that **Larry Lamoray** has been appointed General Manager to oversee sales and support of both Amek and TAC products in the United States. Mr. Lamoray was previously with AEG.

• The Magnetic Tape Division of **Agfa-Gevaert, Inc.**, of Ridgefield Park, New Jersey, announced the recent promotion of **Teri Sosa** to the position of Marketing Communications Manager. Sosa will be responsible for creating and implementing all marketing communications for the division, including sales promotions, advertising, public relations, trade shows and technical/sales support materials. Sosa has been associated with Agfa-Gevaert for five years as Assistant Advertising/Promotions Manager.

• **Bose Corporation**, the international audio products manufacturer of Framingham, Massachusetts, announces that Bose has been named official supplier of professional sound system equipment by the XV Olympic Winter Games Organizing Committee in Canada. Bose will provide sound systems for all twelve venues at the Calgary Games, which begin February 13, 1988.

• Rock superstar David Bowie's North American "Glass Spider Tour," features musical instruments and stage systems wired entirely with **Monster Cable**, it was announced by **Gary Reber**, special projects manager for Monster Cable. According to Joe Wirsing, Bowie's equipment manager, Monster Cable was chosen for the wiring because of its high current capabilities, punchy bottom end, sonic definition and clarity.

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Len Feldman—db magazine
September/October—1986

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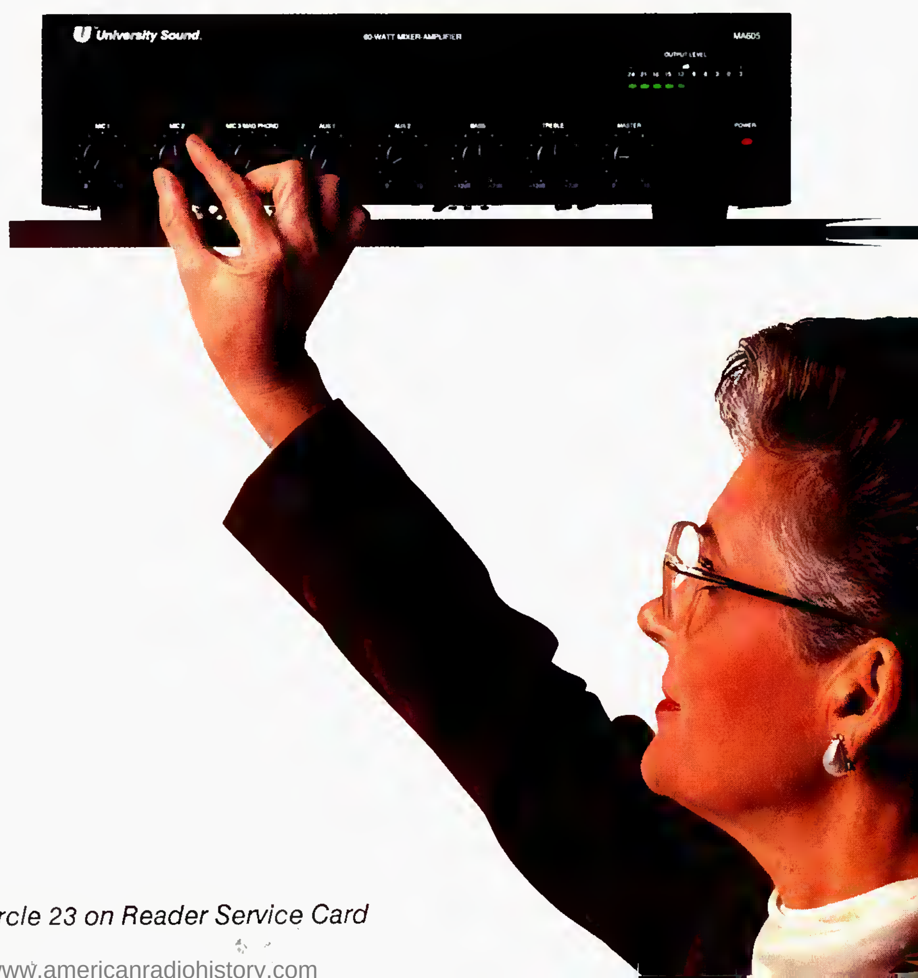

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